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Contents

Chapter 1

Introduction

Mobile networking is quickly reaching a tipping point. While content has been a second-class customer for cellular networks until recently, the wide spread of smart phones, and the access these provide to existing (e.g. social networking) and novel applications, are generating unprecedented amounts of mobile content. Indeed, user demand for content is increasing and creating a shift in focus towards content and content centric systems [1], in both wired and wireless Internet. According to statistics published by ComScore [2], content dissemination through social networking ranks as the fastest-growing mobile content category. It was also reported that mobile content traffic exerted by mobile devices fetching content from the Internet is already a drainage of mobile operators' network resources [3, 4, 5] . Similar to the wired Internet, mobile users are now coping with the congestion at network gateway. The capacity of current cellular infrastructures (e.g. GPRS and 3G) has already been pushed to the limit by even a small number of eager data plan users [3]. To support the increasing number of devices generating content at high rates, ISPs will inevitably be pushed towards either lowering bandwidth quotas [3], adopting non flat rate plans, or deploying (expensive) next generation equipment (e.g. LTE). This has lead many researchers (and industry) to explore alternative or hybrid architectural solutions [6, 7].

To this end, direct mobile-to-mobile communication can be leveraged to harvest the large amounts of unused bandwidth between wireless devices in proximity. Mobile devices with multiple wireless interfaces (e.g. Bluetooth and WiFi) allow two users in range to exchange content at much higher speeds, lower power consumption per bit, and essentially no (direct) monetary cost [8]. This raises an opportunity for a content dissemination overlay over the large numbers of mobile devices *in the wild*, meeting with each other in passing. Nevertheless, users mobility and the much shorter range of high speed interfaces makes *contacts* between devices inherently *intermittent* and *time-limited*. Indeed, as the topology is very unstable, content providers and content consumers might be completely unaware of each other and never connected at the same time to the same part of the network. Therefore, content should be replicated and moved towards users in a *store-carry-and-forward* manner. The latter approach fall within the concept of Disruption Tolerant Networks (DTNs) that tolerates network partitions, long disconnection and topology instability in general and that considers users mobility as being the most effective way to deliver content to interested users. Since the DTN concept was introduced, a significant share of research has focused on the design of point-to-point content routing protocols and applications like the ones dedicated for large-scale disaster recovery, for ecological and ocean monitoring [9, 10], for vehicular networks [11], and projects such as TIER [12], Digital Study Hall [13] and One Laptop Per Child [14] to benefit developing nations. And recently, researchers switched the focus to a second category of DTN architectures, the ones based on point-to-multipoint communication model and that aim to provide complementary content dissemination solutions to the Internet based ones.

Irrespective of the DTN application category, the uncertainty about network conditions in a disruption tolerant environment makes content routing and dissemination a challenging problems and rises many questions with regards to the management of the mobile devices resources (storage, energy, etc). Moreover, mobile devices are usually small and light equipments with limited resources (storage, battery power, limited radio range). Consequently if an application context requires the cooperation from rational users, we should expect them to adopt selfish behaviors when deciding to replicate the content towards maximizing their revenues and conserving their resources (for example, their battery life or storage capacity). This context introduces a new class of problems for content routing and dissemination in DTN(s) which we detail in the following section.

1.1 Challenges faced by content routing and dissemination in DTN(s)

Mobility Due to frequent topology changes, network partitioning and disruption occur very often in mobile networks than in wired networks. Network partitioning severely reduces content availability when the user that holds the desired content is not in the same partition where the client users are. Replicating content in future separate partitions before the occurrence of network partitioning can improve content availability. Content redundancy can also increase the chance for users to find the closest content while moving. Therefore the replication mechanism should consider all these dynamic natures of mobile network in order to replicate content items beforehand. However the combination of long-term storage and the, often expensive content replication imposes a high storage overhead on wireless devices. Therefore, the replication mechanism should also take into consideration the devices storage limitation and provide the suitable management mechanisms towards delivering the contents under optimal conditions. For the same reasons, when mobility results in short contacts between users, available bandwidth could be insufficient to communicate all intended contents. Consequently, efficient scheduling policies should be provided to decide which content should be chosen and forwarded first when bandwidth is limited, regardless of the specific routing algorithm used.

Energy constraints and load balancing Mobile devices operate on batteries which are assumed to have limited capacity despite the advance in battery technology. A single device may serve many clients, which causes its power to be exhausted very quickly. To improve content availability, the replication mechanism should replicate the content items to share the content providing tasks to other devices and prevent some devices from energy exhaustion. Moreover, it should also replicate content in such way that the power consumption of devices is reduced and is balanced among the devices that are in the network. In this case an approach that embraces the peer-to-peer (P2P) paradigm (i.e. no role is per-assigned to a device, every device can be either a client or a server alternatively) would help solving the problem. But we should think about an unstructured version of P2P design to cope with the high dynamic nature of mobile networks.

Content availability A disruption tolerant network may involve a large population with thousands of devices, for example, in a crowded scenario like at a stadium or in a museum. In such dense and large network, to lookup content, a query sent by a client device may need to traverse a long path to reach a replica, therefore increasing the query cost and latency. Moreover, the existence of a large number of querying devices may cause more channel interference among clients, which thus decreases considerably the available bandwidth and increases channel access delay. High users mobility may also affect the availability of content. Thus, the replication scheme should also be designed in such way that its performance will not be greatly affected by the large number of devices and high mobility.

Selfish users Mobile devices are controlled by rational users who are aware of the energy constraint and the cost to share and replicate content. Given this fact, one can predict that users will behave selfishly to minimize their own cost and do not care about the system side effects unless they are provided incentives to replicate the content. The total cost computed at the Nash equilibrium in this case can exceed the optimal cost by a large gap. The system thus should discourage potential selfish behaviors by designing a mechanism that motivate users to store the content if this allows to improve the performance and reduce total cost.

1.2 Contributions

In this thesis, we start by discussing the problem of content dissemination in wireless environments. We describe brifely a solution that we developed towards managing in an efficient way content sharing in a mobile AdHoc network environment (MANET) and we give an overview of the limitations that the latter solution could face in a wireless disruption prone environment. We then study the state of the art in terms of proposed solution for both content routing and dissemination solutions in DTN(s). Our main observations were first that *(i)* despite a large amount of effort invested in the design of efficient routing and content dissemination protocols for DTN, there has not been a similar focus on storage management and scheduling policies and second *(ii)* that considering *one-to-many* large scale content dissemination problem, we identified the need for a content centric solution that should be able to manage in an efficient way the content dissemination process and to implicitly force users to collaborate while preventing selfish ones from impairing the content sharing sessions.

Starting from the latter preliminary study, we wanted to solve the highlighted problems in their foundations. In a first direction *(i)*, we focused on the problem of point-to-point content routing through a DTN. Such a problem is frequently encountered in environment and habitat monitoring based on sensor networks where sensor nodes try to deliver in an efficient way the collected observations and measures to a gateway node. We developed a theoretical framework based on Epidemic message dissemination [15, 16, 17], and proposed a *greedy* optimal joint content scheduling and storage management policy, GBSD (Global knowledge Based Scheduling and Drop) that can either maximize the average delivery rate or minimize the average delivery delay in the context of a congested disruption prone network. GBSD derives a per-message utility by taking into account all information that are relevant for message delivery, and manages messages accordingly. Yet, to derive these utilities, it requires *global* network information, making its implementation difficult in practice, especially given the intermittently connected nature of the targeted networks. In order to amend this, we proposed a second policy, HBSD (History Based Scheduling and Drop), a distributed (*local*) algorithm based on statistical learning. HBSD uses network history to estimate the current state of required (*global*) network parameters and uses these estimates, rather than actual values (as in GBSD), to calculate message utilities for each performance target metric. Furthermore, we looked deeper into our distributed statistics collection solution and identified the available trade-offs between the collection overhead and the resulting performance. Aggressively collecting statistics and exchanging them with every encountered device allows estimates to converge faster (and thus achieves good performance), but it can potentially result in high energy and bandwidth consumption, and also interfere with data transmissions. Our results suggest that close to optimal performance can still be achieved even when the signaling overhead is forced (through sampling) to take only a small percentage of the contact bandwidth. Finally, in this direction, we examined how our algorithm behaves under different congestion regimes. Interestingly, we found that *(i)* at low to moderately congested regimes, the optimal policy is simply equivalent to dropping the message with the oldest age (similarly to the findings of [18]), while *(ii)* at highly congested regimes, the optimal policy is not linear on message age; some young messages have to be dropped, as a means of indirect admission control, to allow older messages to create enough replicas and have a chance to be delivered. Hence, our framework can also explain what popular heuristic policies are doing, in this context, relative to the optimal one.

In a second direction *(ii)*, we cast the solution we proposed for point-to-point content routing to a *channel* based, point-to-multipoint one. Indeed, we believe that towards efficiently handlling content dissemination in disruption tolerant environment at a large scale, a channel based architecture is more suitable. The latter should enable people to express their interests, head out in the real world and wait to get notified whenever a content that matches their interests is carried by any of the encountered devices. To achieve this, a candidate architecture should not only take care of the network and device resources, but also carefully consider: *(i)* the propagation of interests of participating users, *(ii)* the matching of these interests to individual node mobility patterns, and *(iii)* the willingness of involved users to collaborate. This latter point can be a major deal-breaker in any envisioned architecture (as is the case for example in traditional MANETs [19]). As an answer to the latter challenges, we propose MobiTrade, an interest driven content dissemination architecture for opportunistic networks. MobiTrade has no notion of content source or content destination at the device level, but mainly deals with labeled content and content *channels* (a channel is a generic set of labels describing a type of content [20]). Practically, to download content, a user only needs to load the MobiTrade software into his device, express his interests in joining a set of channels (locally on his mobile), publish any content he wants to share, and then turn the application into the background while heading out into the real world. As he walks around, the user device meets other devices and exchanges autonomously with them content matching their specified interests. Unlike many related works [21] [20] [22] [23], MobiTrade supposes that some users might have a selfish behavior. To cope with this, it proposes a *trading* scheme to enforce collaboration among mobile users, to block selfish behaviors and to convey the right content towards the right consumer in a *store-carry-and-forward* mode, without having the source and interested devices explicitly communicating. There are two main components in this trading algorithm. First, Tit-For-Tat is directly employed during individual device meetings. If Alice meets Bob and needs X units of content from him, she must give him X units back (one by one). This is a strict policy that isolates selfish nodes trying to receive content without giving anything back. It also motivates devices to carry content for *foreign* channels (not interested in personally), that they can use as *trading currency*. Second, MobiTrade gives the ability to associate a utility per channel and to identify the most lucrative ones for each device. From a trading point of view, this utility can be seen as the expected reward of storing and carrying a piece of content, by *selling* it to devices encountered in the future. In other words, Mobitrade turns any participating device into a *merchant* with the sole goal of *collecting in its buffer an inventory of data that maximizes its future trading capacity and its resulting revenue (content of interest received).*

The performance evaluation of our two main contributions HBSD and MobiTrade is done respectively through extensive NS2 and NS3 simulations supplied by real mobility traces. In Chapter 4, we implement HBSD, a real external router for the DTN2 [?] reference architecture. the proposed router runs on top of the DTN2 forwarding block, collects and analyses the network history towards approximating network level parameters and providing the right content drop or scheduling decisions to apply in case of a device storage congestion or a contact disruption. We also describe in Chapter 6, the implementation of our channel based content dissemination architecture, MobiTrade, for smart-phones equipped with the Android operating system.

1.3 Thesis organization

The remainder of this thesis is as following. In the next chapter we introduce our problem background and discuss a list of related works. Chapter 3 describes the greedy optimal solution that we propose for point-to-point content routing within a disruption tolerant network. Chapter 4 details the implementation issues of our History Based Scheduling and Drop (HBSD) content dissemination schema as an external router for the DTN2 reference architecture. In Chapter 5, we present MobiTrade, our point-to-multipoint interest driven content dissemination architecture for disruption tolerant networks. Then, we provide in Chapter 6, a detailed implementation analysis of MobiTrade for smart-phones equipped with the Android platform. In Chapter 7, we conclude the results of our study and outline the direction for our future work.

Chapter 2

Background

Contents

Content sharing is currently an universal concern among computer users and has recently become an important requirement for mobile devices. Indeed, thanks to the efficient wireless connectivity offered by mobile devices, users are frequently brought to locate and share content of interest (photos, videos, etc) with other members of the same spontaneous community. With current technologies, they are mainly using point-to-point basic connections, which can be considered as an efficient solution when the number of users interested in the sharing session is very small and that there is no possible connexion disruption. However, even with a guaranteed wireless connexion (Ad-Hoc network) and in the case of a large community (for example, mobile devices users assisting to a conference and willing to share some papers), one is facing the following problem: Increasing the number parallel point-to-point communications may decrease the global Ad-Hoc network capacity, while increasing dramatically the download time. The multi-hop point-to-point communication over long paths is also a serious issue. Therefore, there is a strong need to organize the communication overlay among devices in a way to distribute fairly the burden of content sharing among the set of participants while aiming to decrease the global download time. P2P file sharing solutions are good candidates for such infra-structureless networks (MANET) as they are based on multi-sourcing which balances resource consumption among users and reduces the dependency on any central entity. But unfortunately, P2P content sharing applications developed for the Internet cannot directly be plugged and used into mobile devices. Indeed, on one hand, these solutions are not adapted to the constraints of multi-hop wireless networks. For example, it is known that in a resource constrained environment, the choice of the users to whom to connect cannot be done independently of information on the underlying dynamic topology. Moreover, centralized users management approaches like the centralized tracker used in BitTorrent do not perform well in such environment as the tracker can be either far away or even invisible by some users because of disconnections. Furthermore, computer users rely on Internet search engines and dedicated desktop applications to look for the content they are willing to share. This approach becomes obsolete in the case of a spontaneous MANET based community and thus, a dedicated distributed content discovery approach must be provided.

Then, if we consider a more general/challenging mobile environment where the topology is unstable and users' contact can be disrupted frequently (for example, mobile devices users moving in the street), users cannot rely any more on the content dissemination systems proposed for conventional MANETs [24] [25]. Indeed, the later ones are built based on the assumption that the network path are almost stable and that content providers and content consumers are connected to the same part of the network at the same time. Therefore, they are not suitable for disruption tolerant environment. From this perspective and to allow some services to operate even under these challenging conditions, researchers have proposed a new networking paradigm, often referred to as Disruption Tolerant Networking, based on the store-carry- and-forward routing principle. Devices there, rather than dropping a session (and respective packets) when no forwarding opportunity is available, store and carry content until new communication opportunities arise.

This Chapter describes the background behind our work. We start by presenting BitHoc, an open-source standalone content sharing solution that we proposed and developed for MANETs and we detail the reasons that prevent users from adopting BitHoc as a content dissemination solution in the context of a disruption tolerant environment. Then we give an overview of already exisiting solutions for content routing in a disruption tolerant environment (both point-to-point content routing and poit-to-multipoint routing protocols) and we detail the limitations of the different proposed approaches which we were able to overcome throught the architectures we are proposing in this thesis.

2.1 BitHoc: A Content Sharing Solution for MANET

In this section, we described our solution for content sharing in wireless Ad-Hoc networks. It contains three components: a distributed membership management service, a content search engine and an optimized content sharing service. The design of these services takes into consideration the constraints of mobile environments and the user needs. With the help of a real test-bed composed of PDAs and Smartphones, we were able to validate our solution in a real scenario and to compare it to other classical approaches. The experiments show that our application outperforms the classical approach and highlight the utility of its features. In particular, we were able to reduce by a factor of 2 to 3 the download time and to increase dramatically the sharing ratio. We designed our package in such a way to be standalone. So, the user has just to install the software to start publishing and discovering contents and sharing them later with those having the same interest. The wireless nodes not interested by the same content collaborate by forwarding packets at the routing level. Through BitHoc, we provide solutions to the following problems:

- In the classical version of BitTorrent [26], peers periodically contact a central rendezvous point called Tracker to obtain fresh information about the peers interested in a specific content and to update their information on the progress of the download. This membership information is dynamic since peers can join and leave the content sharing overlay (called torrent) at any time during the session. Because of the inappropriateness and the large overhead of client/server architectures in wireless Ad-Hoc networks, it is important to introduce a distributed Trackerless solution to manage the membership of the sharing session. The BitHoc tracker component of our architecture is designed for this purpose and is inspired from the membership management protocol we presented in details in [24].

- The classical version of BitTorrent [26] supposes that the cost of sending data packets to peers is in somehow independent of their locations. In an Ad-Hoc network, performance metrics like achievable throughput, delay, and energy consumption strongly depend on the number of hops to the peer node. So, it is clearly suboptimal and even unrealistic to deal with peers without considering the underlying topology. Furthermore, when applying the classical BitTorrent incentives in a wireless multi-hop network, nodes fail to reciprocate data fairly among them. The content dissemination scheme is close to a wave transferring data from the initial seed to the farthest peers. Through new peer selection and content piece scheduling strategies, our solution is topology-aware and ensures fair sharing. These strategies are described in details in [24]

- To join a sharing session, a user should find and download the Torrent file related to that session. In the Internet, peers usually find their torrent files by the help of search engines which mainly look for the files in different central servers. This method does not apply in a mobile Ad-Hoc environment as MANETs. The BitHoc search engine overcomes this challenge by maintaining a distributed torrent file database thanks to the overlay constructed by the BitHoc Tracker.

2.1.1 Architecture of BitHoc

Figure 2.1 depicts the principal components of this architecture and the interactions between them. We illustrate these interactions through three typical usage scenarios:

Content publishing and discovery A user willing to share some content with the members of his community needs to indicate to the BitHoc client the location of

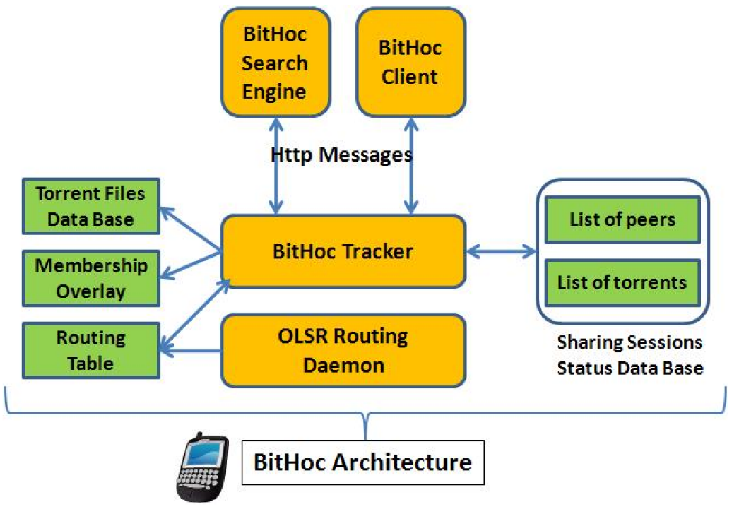

Figure 2.1: Architecture of BitHoc

the content in the mobile device file system. First, the client creates a meta-info file (Torrent file) that identifies in a unique manner a sharing session for this specific content. After that, the user publishes (locally) the new torrent file and a short text description of the related content using the BitHoc Search Engine service, which will update the local Torrent file database maintained in the underlying BitHoc Tacker via HTTP messages. A remote user, willing to share the same content, has to use the BitHoc search engine to find and download the Torrent file. He specifies for that the name of the content or some keywords related to its description. The request is sent via HTTP messages to its local tracker which looks for the closest match in its local database. If there are no matches, it forwards the HTTP request to the other trackers in the discovery overlay. Then, it presents the received results through an ergonomic user interface (see Figure 2.2). Based on the details of received answers (fitness to the search, number of peers involved in the sharing session, number of seeders, and number of lechers, etc), the user can choose the torrent file to download, then start sharing the content using the BitHoc Client.

Membership management When a peer wants to join or leave the sharing session, the BitHoc client informs the BitHoc Tracker about this event using a specific HTTP message. This local agent disseminates this modification to the other BitHoc Tracker agents in other nodes in order to update their knowledge about the global membership information. The communications between Tracker agents are established in an event-driven fashion and use HTTP messages. Each tracker holds a HTTP server accepting HTTP requests from other agents and from the local BitTorrent client. The BitHoc Tracker component receives from the routing daemon up-to-date routing entries. In our testbed, the dynamics of the Ad-Hoc network are

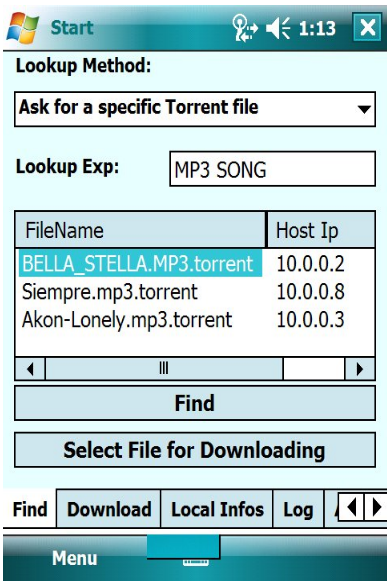

Figure 2.2: Search Engine screen shot

captured by the OLSR routing protocol [27]. Each time the number of hops toward a given peer changes, the routing daemon fires an event, which will be caught by the BitHoc Tracker and forwarded internally to the BitHoc client. This way we are sure the peer selection strategy always uses the updated number of hops to other peers. The parameters of the communications among tracker agents like HTTP listening ports and IP addresses can easily be configured by users via an ergonomic GUI. In addition to these functionalities, the BitHoc Tracker allows the user to monitor in real-time the status of the overlay (Contents it shares, members of the session, current topology of the Ad-Hoc network). He can even decide to keep traces about all the events in a file. For this, he just needs to activate the tracing option provided by the application.

Content sharing Before starting a new sharing session, the user can choose between two versions of BitTorrent algorithms: The classical version [26] and our version adapted to mobile Ad-Hoc networks described in [24]. The BitHoc client offers a Wizard allowing the user to configure the parameters of BitTorrent (communication ports, choking slot duration, minimum and maximum number of peers, etc). Once the torrent file is obtained, the BitHoc client can start the sharing session where it can either play the role of a leecher or a seed. It contacts periodically the local BitHoc tracker to get the current list of members of the same content sharing session (torrent). Using this list and the routing table, it manages the connections with the interested peers. Briefly a client implementing our algorithms exchanges pieces with close peers and only seeds distribute pieces across the network. Note that we allow the user to pause or resume the download while conserving the session context. He can also monitor in real time the status of the session (downloaded bytes, uploaded bytes, numbers of leechers, number of seeders, elapsed time, etc). Furthermore, the BitHoc client keeps in a log file statistics on the content sharing session and provides different levels of event traces. It also manages the storage of the downloaded contents and their classification. Figure 2.3 shows a screen-shot of the BitHoc client.

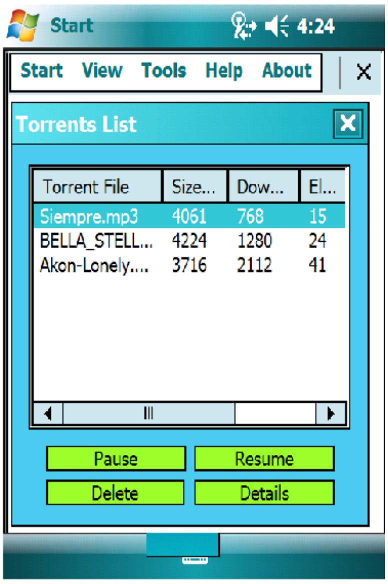

Figure 2.3: BitHoc Client screen shot

2.1.2 Experimentation and results

Test-bed description Our wireless Ad-Hoc network experimental environment consists of 14 mobile devices including 7 PDAs (HP iPAQ 214) and 7 smartphones (HP iPAQ 614c). Each handheld is equipped with an IEEE802.11b wireless card. The characteristics of the two types of devices are detailed in Table 2.1.2. The Ad-Hoc connectivity is maintained thanks to OLSR daemons run by the different devices. In our experiments, we constructed several network topologies containing a maximum of 6 hops. The objective of the realized swarm was to download a 4 MB MP-3 content. All PDAs were supposed to participate to the sharing of the file. The original seed of the content was chosen randomly among the set of the 14 PDAs.

Experimentation results The metrics tracked during our experiments are the download time and the average sharing ratio of nodes. We define R_h as the sharing ratio of peers located at h hops from the original seed. It measures the level of reciprocity between downloads and uploads. In the ideal case, the ratio should be close to 1. The two versions of BitTorrent (The legacy one and ours) have been

Table 2.1: Characteristics of mobile handhelds

	PDA	**Smartphone**
Name	HP iPAQ 214	HP iPAQ 614c
Processor speed	624 MHz	520 MHz
RAM	128 MB	128 MB
Operating system	Windows Mobile 6	Windows Mobile 6

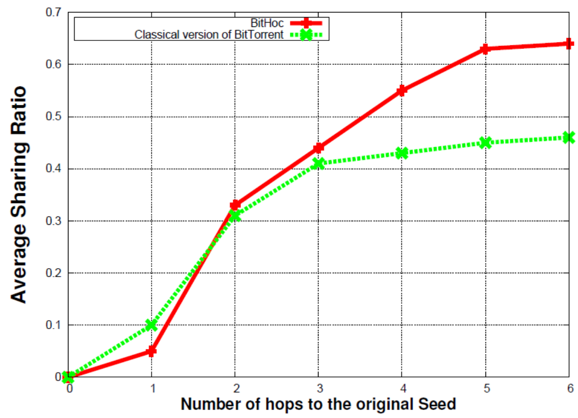

Figure 2.4: Sharing ratio

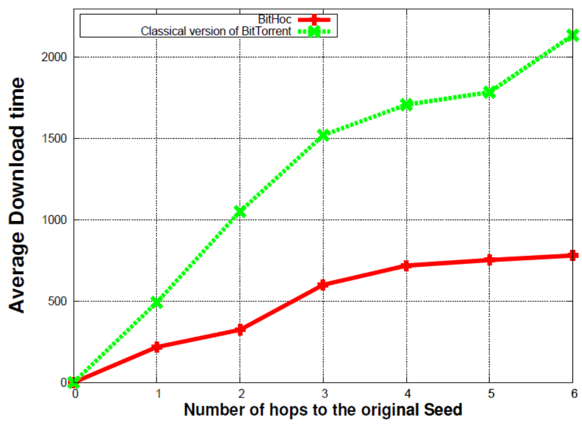

Figure 2.5: Download time

tested and the results are presented in Figures 2.4 and 2.5. Figure 2.4 shows a dramatic increase of sharing opportunities when our adapted version is deployed. The routing overhead generated by the classical version makes any gain obtained by important diversification of pieces negligible. Our method finds the good equilibrium between sharing and diversification. Figure 2.5 shows that BitHoc outperforms the classical version of BitTorrent in terms of download time. It is in accordance with our research results presented in [28]. More information about our experiments and our GPL licensed open-source code can be found on the BitHoc web site [28].

2.1.3 BitHoc limitations with respect to a disruption prone environment

We should note that BitHoc is built based on the assumptions that the network path are almost stable and that content providers and content consumers are connected to the same part of the network at the same time. Indeed, as described in Figure 2.1, BitHoc relies on the routing table built and maintained by the OLSR MANET routing protocol towards maintaining the sharing sessions membership overlay and in order to deliver messages to peers at more than one hop. Therefore, BitHoc is not suitable solution for content dissemination in a disruption tolerant environment.

2.2 Disruption Tolerant Networks

Delay and disruption tolerant networks (DTNs) are a new class of wireless networks that seek to address the networking issues in mobile or challenging environments that lack continuous network connectivity. DTNs have emerged recently and are continuing to gain extensive efforts from the networking research community [29, 30, 31]. In the literature, these networks are found under different terminologies such as sparse mobile ad hoc networks, extreme wireless networks, or under another commonly used term intermittently connected networks. Basically, DTNs appear in areas where the network spans over large distances with low node density and/or with high node mobility. DTNs might appear also due to short radio range, power saving mechanism at the nodes, or nodes failure. Examples of such networking scenarios include, but are not limited to :

- Vehicular networks, e.g. [11, 32]. In [32], the authors propose the Drive-thru Internet architecture where the objective is to provide network and Internet connectivity to mobile users in vehicles. The network is constituted by hot spots that are placed along the roads providing thus intermittent connectivity to the users that can connect within proximity. In [11], Burgess et al. introduce UMass DieselNet which is a network made of 30 buses equipped with 802.11b wireless interfaces and GPS devices. The objective of the network is to provide real DTN testbed for experimental and research studies. The buses move on regular trajectories inside the UMass Amherst campuses and surrounding areas. When two buses pass nearby, they transfer data to each other. Additionally, buses can connect to open wireless access points along the roads.

- Mobile sensor networks for environmental monitoring, e.g. [33, 34]. Zebranet [33] is a wireless networking architecture designed to support wildlife tracking for biology research. In ZebraNet, the network is constituted by sensor collars that are attached to zebras, which log movement patterns of the zebras, and by researchers base stations that are mounted on cars which move around sporadically. When two zebras meet, the corresponding sensors exchange collected data for a potential data delivery back to researchers base-stations. Another similar biological acquisition system has been proposed in [34], where the network is made of a set of sensors attached to whales and a set of fixed info-stations that act as collecting nodes.

- Communication between rural zones in toward development countries, e.g. [35]. Examples include DakNet [35] which is a wireless ad hoc network that has the capacity to provide asynchronous Internet access to remote rural residents using motorcycles and buses to carry users email and web search messages.

- Deep space networks such as the Inter-planetary network (IPN) [36]. The interplanetary network is a network of regional Internet networks. A region is an area where the characteristics of communication are the same. An example of regions includes the terrestrial Internet as a region or a ground-to-orbit region. IPN aims to achieve end- to-end communication through multiple regions in a disconnected, variable-delay environments.

- Challenged networks such as disaster healing networks after natural disaster, travel information and advertisements dissemination systems in large cities using local transport systems, military ad hoc networks where disconnection occurs because of the war or for security reasons where some links need to be shut down from time to time.

Generally speaking, DTNs are wireless networks that do not conform to Internet or to traditional multihop and ad hoc wireless networks underlying structures and assumptions. In particular, they are characterized mainly by the following specific features [30, 37]:

- Intermittent connectivity where an end-to-end path between a given source-destination pair does not exist most of the time. Path disconnections are frequent and arise from two main factors, namely motion and/or limited power at the nodes. Disconnection due to motion can arise when one or both nodes at the end of a communication link move, or due to some intervening object or signal that obstruct the communication. These disconnections can be predicted, for instance when the nodes move away according to a predetermined schedule or, an opportunistic for instance according to random walk of the nodes. Disconnections that are due to power outage result commonly from some power saving mechanisms at the wireless devices, e.g. case of sensor networks. The latter disconnections are often predictable.

- Nodes have low power capabilities and limited resources. In many DTNs, nodes are generally battery powered and/or deployed in areas lacking power infrastructure. In some other situations, nodes have limited memory and/or processing capabilities.

- Large delays which are basically due to long queuing times resulting from frequent disconnections, or from low data rate at the devices.

2.3 Content Routing in Disruption Tolerant Networks

Due to frequent disconnections in DTNs, instantaneous end-to-end routes do not exist, and hence most of the traditional Internet and/or mobile ad hoc content routing protocols fail [38]. However, end-to-end routes may exist over time if the nodes can take advantage of their mobility by exchanging and carrying other node messages upon meetings, and by delivering them afterward to their corresponding destinations. The latter concept has given rise to a novel routing paradigm in these networks called the store-carry-and-forward approach, in which intermediate nodes serve as relays for each other. Thus, the term "mobility-assisted routing approach" that is used in conjunction to describe these schemes.

Unfortunately, these techniques result in high latency, since packets need to be carried for long time periods before being delivered. When the delivery latency is not critical, as the case of delay-tolerant networks, the store-carry-and-forward paradigm can prove to be adequate. For instance, this is the case when the delivery of the messages is very important, possibly more important than the delay. Basically, with the store-carry-and-forward approach, the delivery delays of packets depend on the rate at which contact opportunities are created in the network, as well as the availability of network resources, such as storage space and energy. The various studies that considered routing techniques in DTNs have examined the trade-offs between optimizing the delivery ratio and delivery delay from one side, and reducing node resource consumptions in terms of storage and battery usage from the other side. However, the intricacy of each one depends on the particularity of network environment at hand, the mobility model of the nodes, the performance objectives to attain, and other criteria.

This section will survey and classify various research works that have considered content routing schemes for DTNs. Actually, there are different approaches to categorize these schemes [39, 40]. Hereafter, we propose a classification that is based on the content distribution method. Specifically, depending on whether these schema operate on a point-to-point basis or point-to-multipoint one.

2.3.1 Point to Point Content Routing in Disruption Tolerant Networks

We classify related existing DTN point-to-point routing protocols as those that replicate packets and those that forward only a single copy. Epidemic routing protocols replicate packets at transfer opportunities hoping to find a path to a destination. However, naive flooding wastes resources and can severely degrade performance.

Proposed protocols attempt to limit replication or otherwise clear useless packets in various ways: *(i)* using historic meeting information [41, 42, 11]; *(ii)* re-moving useless packets using acknowledgments of delivered data [11]; *(iii)* using probabilistic mobility information to infer delivery [43]; *(iv)* replicating packets with a small probability [44]; *(v)* using network coding [45] and coding with redundancy [46]; and *(vi)* bounding the number of replicas of a packet [43, 47, 48].

In contrast, forwarding routing protocols maintain at most one copy of a packet in the network [38, 49, 50]. Jain et al. [38] propose a forwarding algorithm to minimize the average delay of packet delivery using oracles with varying degrees of future knowledge. Deployment experience [51] suggests that, even for a scheduled bus service, implementing the simplest oracle is difficult; connection opportunities are affected by many factors in practice including weather, radio interference, and system failure. Jones et al. [49] propose a link-state protocol based on epidemic propagation to disseminate global knowledge, but use a single path to forward a packet. Shah et al. [52] and Spyropoulos et al. [50] present an analytical framework for the forwarding-only case assuming a grid-based mobility model. They subsequently extend the model and propose a replication-based protocol, Spray and Wait [47]. The consensus appears to be [47] that replicating packets can improve performance (or security [53]) over just forwarding, but risk degrading performance when resources are limited.

Our position is that most existing point-to-point routing schemes does not take into consideration the impact of buffer management and scheduling policies on the performance of the underlaying system. The later issues has been largely disregarded, in comparison, by the DTN community. And thus, most routing protocols only have an incidental effect on desired performance metrics, including commonly evaluated metrics like average delay or delivery probability. For example, Spray and Wait [47] like many other routing protocols [43, 48] that route packets using the number of replicas as the heuristic, to enhance a given routing metric, does not take explicitly into account bandwidth or storage constraints which makes the effect of their design decision on the performance of a given resource constrained network scenario unclear. Nevertheless, some works already investigated the impact of plugging simple drop policies to already exiting routing protocols, like in [54], Zhang et al. present an analysis of buffer constrained *Epidemic* routing, and evaluate some simple drop policies like drop-front and drop-tail. The authors conclude that drop-front, and a variant of it giving priority to source messages, outperform drop-tail in the DTN context. A somewhat more extensive set of combinations of *heuristic* buffer management policies and routing protocols for DTNs is evaluated in [18], confirming the performance of drop-front. In [55], Dohyung et al. present a drop policy which discards a message with the largest expected number of copies first to minimize the impact of message drop. However, all these policies are also heuristic, i.e. not explicitly designed for optimality in the DTN context. Also, these works do not address scheduling.

Yet, the combination of long-term storage and the, often expensive, message replication performed by many DTN routing protocols impose a high bandwidth and storage overhead on wireless nodes. Moreover, the data units disseminated in this context, called bundles, are self contained, application-level data units, which can often be large. As a result, it is expected that nodes' buffers, in this context, will often operate at full capacity. Similarly, the available bandwidth during a contact could be insufficient to communicate all intended messages. Consequently, we believe that regardless of the specific routing algorithm used, it is important to have: *(i)* efficient drop policies to decide which content(s) should be discarded when a node's buffer is full, and *(ii)* efficient scheduling policies to decide which content(s) should be chosen to exchange with another encountered node when bandwidth is limited.

Table 2.2 shows a taxonomy of many existing DTN routing protocols based on assumptions about bandwidth available during transfer opportunities and the storage carried by nodes; both are either finite or unlimited. For each work, we state in parentheses the mobility model used. $R1$ and $R2$ are important to examine for valuable insights that theoretical tractability yields but are impractical for real DTNs with limited resources. Many studies [56, 41, 42] analyze the case where storage at nodes is limited, but bandwidth is unlimited ($R3$). This scenario may happen when the radios used and the duration of contacts allow transmission of more data than can be stored by the node. However, we find this scenario to be uncommon typically storage is inexpensive and energy efficient. Trends suggest that high bitrate radios will remain more expensive and energy-intensive than storage [57]. Finally, for mobile DTNs, and especially vehicular DTNs, transfer opportunities are short-lived [11].

Table 2.2: A classification of some related work into DTN routing scenarios

Cat.	Storage	Bandwidth	Routing	Work (and mobility)
R1	Unlimited	Unlimited	Replication	Epidemic [48], Spray and Wait [47]: Constraint in the form of channel contention (Grid-based synthetic)
R2	Unlimited	Unlimited	Forwarding	Modified Djikstra's algorithm Jain et al. [38] (simple graph), MobySpace [58] (Powerlaw)
R3	Finite	Unlimited	Replication	Davis et al. [41] (Simple partitioning synthetic), SWIM [43] (Exponential), MV [42](Community-based synthetic), Prophet [59] (Community-based synthetic)
R4	Finite	Finite	Forwarding	Jones et al. [49] (AP traces), Jain et al. [38] (Synthetic DTN topol- ogy)
R5	Finite	Finite	Replication	Our proposal (Vehicular DTN traces, testbed deployment), RAPID [60] (Vehicular DTN traces, testbed deployment), MaxProp [11] (Vehicular DTN traces)

We were able to find mainly two protocols that belong to the category $R5$. The first *(ii)*, Max-Prop [11] that assume limited storage and bandwidth. However, it is unclear how to optimize a specific routing metric using MaxProp, so we categorize it as an incidental routing protocol. And the second *(ii)*, RAPID [51] that is the first protocol to explicitly assume both bandwidth and (to a lesser extent) buffer constraints exist, and to handle the DTN routing problem as an optimal resource allocation problem, given some assumption regarding node mobility. As such, it is the most related to our proposal, and we will compare directly against it. Despite the elegance of the approach, and performance benefits demonstrated compared to well-known routing protocols, RAPID suffers mainly from the following drawbacks: *(i)* its policy is based on suboptimal message utilities (more on this in Section 3.1); *(ii)* in order to derive these utilities, RAPID requires the flooding of information about all the replicas of a given message in the queues of all nodes in the network; yet, the information propagated across the network might arrive stale to nodes (a problem that the authors also note) due to change in the number of replicas, change in the number of messages and nodes, or if the message is delivered but acknowledgements have not yet propagated in the network; and *(iii)* RAPID does not address the issue of signaling overhead. Indeed, in [60], the authors showed that whenever the congested level of the network starts increasing, their meta-data channel consumes more bandwidth. This is rather undesirable, as meta-data exchange can start interfering with data transmissions amplifying the effects of congestion. In another work [61], Yong et al. present a buffer management schema similar to RAPID. However they do not address the scheduling issue nor the trade-off between the control channel overhead and system performance. Through our proposal, we successfully address all these three issues.

2.3.2 Point to Multi-Point Content Dissemination in Disruption Tolerant Networks

As hilighted in the latter section, a significant share of research on opportunistic networks has focused on *unicast* point-to-point content routing (see e.g. [62] or [63]). Instead, in this section, we consider the problem of content dissemination. This is a key research problem, particularly in opportunistic networks. In this environment, according to the user-generated content wave, users are expected to generate large amounts of content by exploiting capability-rich mobile devices (such as PDAs, smartphones, etc.), and to share them with people around them. The problem of efficiently disseminating contents in opportunistic networks is thus very relevant, and not widely explored in the literature yet.

Content dissemination in opportunistic networks is a difficult problem. As the topology is very unstable, and users appear in and disappear from the network dynamically, content providers and content consumers might be completely unaware of each other, and never connected at the same time to the same part of the network. Therefore, contents should be moved and replicated in the network in order to carry them to interested users despite disconnections and partitions. On the other hand, content dissemination systems should take care of both network and device resource constraints. For example, a trivial solution would be to flood the whole network with any generated content, but this would clearly saturate both network resources (in terms of available bandwidth) and device resources (e.g., in terms of energy, storage, etc.). Content dissemination systems should also take care of the willingness of people to collaborate. Indeed, experience teaches us that selfish behavior is often the norm, unless incentives are provided, and can be a major impediment to any such peer-to-peer system in the wild [19]. Thus, we believe that content dissemination systems should consider users to be inherently selfish, instead of inherently collaborative and provide the necessary mechanisms to enforce collaboration and prevent the bad impact of selfish behaviors on the overall system performance.

Content dissemination systems have been proposed for the Internet, and also for conventional MANETs [24]. In general, these systems assume that network paths are rather stable, and often generate a significant amount of traffic to maintain knowledge of other devices' caches. Therefore, they are not suitable for opportunistic networks. Table 2.3 shows a taxonomy of most of existing DTN content dissemination systems. We classify the later systems into three categories, namely *D1* content centric dissemination systems guided by users *Interests*, *D2* systems driven by users interests + social links and finally *D3*, dissemination systems guided by users interests and their locations. We also detail in Table 2.3 whether the presented content dissemination systems provide or not needed mechanisms to handle devices' buffers management in case of congestion, contents scheduling during short lived contact opportunities and users selfishness.

Table 2.3: A classification of content dissemination systems for DTN(s)

Cat.	**Driven by**	**Work (buffer management, scheduling, users selfishness, mobility)**
D2	Users Interests	Our proposal, MobiTrade(handled, handled, inherently selfish, Vehicular DTN traces & testbed deployment), DTN Podcasting by May et al. [20] [64](not handled, not handled, inherently collaborative, testbed deployment), TACO-DTN by Solazzo et al. [65](handled, handled, inherently collaborative, random waypoint), BlueTorrent [25]()
D1	Users Interests + Social Links	ContentPlace by Boldrini et al. [66](handled, handled, inherently collaborative, synthetic based on the HCMM model [67]), SocialCast by Helgason et al. [68, 22](not handled, not handled, inherently collaborative, real human mobility traces + testbed deployment)
D3	Users Interests + Location	Locus by Thompson et al. [69](handled, handled, inherently collaborative, synthetic via MobiSim [70]), PeopleNet by Motani et al. [23](handled, not handled,inherently collaborative, random walk)

TACO-DTN [65] by Sollazzo et al. is a time-aware approach to delay tolerant content based dissemination. It is implemented as a publish/subscribe system and was mainly designed to distribute temporal events. Temporal profiles are associated to each subscription and allow the construction of temporal profiles of infostations.

Events also have temporal validity. TACO-DTN uses temporal profiles in order to achieve two main tasks: buffer management, in order to decide which events to store when buffer space is limited, and event routing, to select the right infostation or carrier on which to publish content.

ContentPlace [21] by Boldrini et al. attempts to improve Podcasting using explicit knowledge of social networking links of participants. The idea behind ContentPlace is to exploit social information on the environment the nodes operate, in order to enhance content dissemination. In the framework of opportunistic networks, this approach has already been successfully applied to message forwarding (e.g., [71]). The idea is to move messages closer and closer to their destinations following a path based on the social interactions between nodes. In the case of forwarding protocols, however, messages have a specific destination node, while in ContentPlace, following the user generated content approach, content generators might be unaware of the nodes interested in their data, and so might be the content consumers about the nodes that generate the content they are interested in. ContentPlace provides also mechanisms to handle devices buffer congestion and content scheduling. Indeed, it assumes that users belong to social communities, and autonomically learns the time spent by them in each community, which types of contents users of each community are interested in, and how spread in the communities the contents are. This information is used to evaluate the utility of each encountered content which is later used to evaluate the contents the remote peer is carrying and to select the ones that should be fetched in order to maximise the total utility of the contents in the local buffer. Compared to ContentPlace, our proposal does not require such user reported social information and does not make any hard assumptions regarding node mobility.

Peoplenet [23] is hybrid system propagating and matching queries over, first, infrastructure, and, second, using DTN device-to-device communication in the wireless "last hop" (e.g. inside a cell) to forward further. It uses the infrastructure to propagate queries of a given type to users in specific geographic locations, called *bazaars*. Within each bazaar, the query is further propagated between neighboring nodes via peer-to-peer connectivity until it finds a matching query.

BlueTorrent [25] is an opportunistic file sharing application for Bluetooth enabled devices that mimics BitTorent. Authors propose an index (shared contents database) dissemination and file swarming protocols for dynamic, sparse networks. The concept of distributing large files using small resumable atomic chunks is similar to our proposal. However, BlueTorrent relies on Bluetooth whereas our proposal leverages any link-layer technologies. Furthermore, we propose to structure the data in the network into channels and rely on an entirely receiver-driven content dissemination protocol.

SocialCast [68, 22] is an interest driven content distribution framework that exploits predictions based on metrics of social interaction (e.g., patterns of movements among communities) coupled to interest-content matching mechanism to identify the best content carriers. In SocialCast, Kalman filter forecasting techniques [3] are used to predict the future evolution of the movement based on previous observations on some attributes characterizing social behavior. These predictions are used to derive an utility U_i per device and interest i, the latter utility is used to identify whether the corresponding device is the best carrier for the contents matching the interest i or not with respect to all the neighbors devices. Compared to SocialCast, our proposal uses a considerably more sophisticated utility that considers both content demand/popularity and the collaboration level of any user it encounters.

To our best knowledge, the only other work looking at pure content centric dissemination architecture for opportunistic networks is the research thread first initiated by the PodNet project [**?**, 20]. This work proposes a DTN Podcasting architecture, built around the concept of *content channels*, that we also use in this paper. In the first version of PodNet [20], users only store and share channels they are interested in. In a later version [**?**], simple strategies to cache other channels as well are considered, in order to improve the overall performance.

Finally, the most recent work in this thread, by Hu et al [72], attempts a rigorous formulation of the problem of optimally matching channels (to store) to a population of devices. A distributed algorithm is then proposed based on the framework of Markov Chain Monte Carlo optimization. While we find this framework particularly interesting, it also comes at the expense of high complexity, long convergence delays (known in MCMC), and a need for carefully tuned simulated annealing [73].

A major difference of MobiTrade, is that we consider users to be inherently selfish, instead of inherently collaborative as in all the aforementioned studies. Experience teaches us that selfish behavior is often the norm, unless incentives are provided, and can be a major impediment to any such peer-to-peer system in the wild [19]. The only proposal we are aware of, dealing with selfish users in the context of DTNs is [74], where a Tit-For-Tat mechanism ("bartering") is also used between nodes to exchange content. While Tit-For-Tat (TFT) ensures selfish users are blocked, it does not answer itself *how collaborative nodes should optimally (re-)act in the presence of TFT*. This is answered in MobiTrade by a personalized inventory management mechanism, key to almost all the system's functions and good performance.

As a final note, MobiTrade is not a reputation system, as e.g. [75]. In reputation systems, nodes collect and share their opinions about peers with others. In our case, each node forms a personal opinion of peers used to only optimize her actions. Our system is more similar to a *market* of independent traders. As a result, a bad customer for node X might be a good customer for node Y.

Chapter 3

A Greedy Optimal Point-to-Point Content Routing Schema for DTN

Contents

3.1 Optimal Joint Scheduling and Drop Policy

In this section, we first describe our problem setting and the assumptions for our theoretical framework. We then use this framework to identify the optimal policy, GBSD (Global Knowledge based Scheduling and Drop). This policy uses global knowledge about the state of each message in the network (number of replicas). Hence, it is difficult to implement it in a real world scenario, and will only serve as reference. In the next section, we will propose a distributed algorithm that can successfully approximate the performance of the optimal policy.

3.1.1 Assumptions and Problem Description

We assume there are L total nodes in the network. Each of these nodes has a buffer, in which it can store up to B messages in transit, either messages belonging to other nodes or messages generated by itself. Each message has a Time-To-Live (TTL) value, after which the message is no more useful to the application and should be dropped by its source and all intermediate nodes. The message can also be dropped when a notification of delivery is received, or if an "anti-packet" mechanism is implemented [54].

Routing: Each message has a single destination (unicast) and is assumed to be routed using a replication-based scheme [76]. During a contact, the routing scheme used will create a list of messages to be replicated among the ones currently in the buffer. Thus, different routing schemes might choose different messages. For example, epidemic routing will replicate all messages not already present in the encountered node's buffer [48]. For the purposes of this paper, we will use epidemic routing as a case study, for the following reasons. First, its simplicity allows us to concentrate on the problem of resource allocation, which is the focus of this paper. Second, it consumes the most resources per message compared to any other scheme. As a result, it can be easily driven to medium or high congestion regimes, where the efficient resource allocation problem is most critical. Third, given the nature of random forwarding schemes, unless a buffer is found full or contact capacity is not enough to transfer all messages, epidemic forwarding is optimal in terms of delay and delivery probability. Consequently, epidemic routing along with appropriate scheduling and message drop policies, can be viewed as a new routing scheme that optimally adapts to available resources [60]. Finally, we note that our framework could be used to treat other types of traffic (e.g. multicast), as well.

Mobility Model: Another important element in our analytical framework is the impact of mobility. In the DTN context, message transmissions occur only when nodes encounter each other. Thus, *the time elapsed between node meetings is the basic delay component*. The meeting time distribution is a basic property of the mobility model assumed [17, 16][1]. To formulate the optimal policy problem, we will first assume a class of mobility models that has the following properties:

A.1 Meeting times are exponentially distributed or have at least an *exponential tail*;

A.2 Nodes move *independently* of each other;

[1]By *meeting time* we refer to the time until two nodes starting from the stationary distribution come within range ("first meeting-time"). If some of the nodes in the network are static, then one needs to use *hitting times* between mobile and static nodes. Our theory can be easily modified to account for static nodes by considering, for example, two classes of nodes with different meeting rates (see e.g. [77]).

A.3 Mobility is homogeneous, that is, all node pairs have the same meeting rate λ.

Regarding, the first assumption, it has been shown that many simple synthetic mobility models like Random Walk, Random Waypoint and Random Direction [17, 16] have such a property. Furthermore, it is a known result in the theory of random walks on graphs that hitting times on subsets of vertices usually have an exponential tail [78]. Finally, it has recently been argued that meeting and inter-meeting times observed in many traces also exhibit an exponential tail [79]. As we will see in Section 3.1.2, in our framework, *we sample the remaining meeting time only when a drop or scheduling decision needs to be taken.* In a sparse network (as in our case), it can be shown that, at this time, the two nodes in question have already *mixed* with high probability. Thus, the quantity sampled can be approximated by the meeting time from stationarity, or the tail of the inter-meeting time distribution, which, as explained, is often exponential [80]. In other words, it is not required to make the stronger assumption of Poisson distributed inter-meeting times, as often done in related literature.

Regarding the second assumption, although it might not always hold in some scenarios, it turns out to be a useful approximation. In fact, one could use a mean-field analysis argument to show that independence is not required, in the limit of large number of nodes, for the analytical formulas derived to hold (see e.g. [81]).

Finally, in Section 3.1.4, we discuss how to remove assumption [A.3] and generalize our framework to heterogenous mobility models.

Buffer Management and Scheduling: Let us consider a time instant when a new contact occurs between nodes i and j. The following resource allocation problem arises when nodes are confronted with limited resources (i.e. contact bandwidth and buffer space)[2].

(*Scheduling Problem*) If i has X messages in its *local* buffer that it should forward to j (chosen by the routing algorithm), but does not know if the contact will last long enough to forward all messages, which ones should it send first, so as to maximize the *global* delivery probability for *all* messages currently in the network?

(*Buffer Management Problem*) If one (or more) of these messages arrive at j's buffer and find it full, what is the best message j should drop among the ones already in its buffer (*locally*) and the newly arrived one, in order to maximize, let's say, the average delivery rate among all messages in the network (*globally*)?

To address these two questions, we propose the following policy. Given a routing metric to optimize, *our policy, GBSD, derives a per-message utility that captures the* marginal value *of a given message copy, with respect to the chosen optimization metric.* Based on this utility, two main functions are performed:

1. *Scheduling*: at each contact, a node should replicate messages in decreasing order of their utilities.

[2]We note that, by "limited resources", we do not imply that our focus is only small, resource-limited nodes (e.g. wireless sensors), but rather that the offered forwarding or storage load exceeds the available capacity. In other words, we are interested in congestion regimes.

GBSD Optimization Scheme (Input F, Input K, Output U_i)

- **Inputs:**
  - **F**: The expectation of the global metric to optimize.
  - **K**: The total number of different messages within the network.
- **Output:** U_i: The utility of message i.
- **Objective:** At each contact, maximize the gain $\Delta(F)$ on the global metric **F**

$$F = \sum_{i=0}^{K(t)} f_i$$ Where f_i is the per-message metric value (ex: message delivery probability or delivery delay)

- In the case of congestion (limited transfer opportunity) a DTN node should take the drop (replication) decision leading to the best gain for **F**. To find this decision, we differentiate **F** with respect to n_i, (number of copies of the message i), and then discretize and replace dn_i by $\Delta(n_i)$ to obtain:

$$\Delta(F) = \sum_{i=0}^{K(t)} \frac{\partial f_i}{\partial n_i} \Delta(n_i) = \sum_{i=0}^{K(t)} U_i \Delta(n_i) \quad \text{and} \quad U_i = \frac{\partial f_i}{\partial n_i}$$

$$\begin{cases} \Delta(n_i) = -1 & \text{If we drop an already existing message i from the buffer} \\ \Delta(n_i) = 0 & \text{If no action for message i is taken} \\ \Delta(n_i) = +1 & \text{If we replicate message i during a contact (scheduling), or if we store a newly received message i (buffer management)} \end{cases}$$

- Based on this, a DTN node should (i) replicate messages in the order of decreasing U_i (ii) drop the message with the lowest U_i when buffer is full

Figure 3.1: GBSD Global optimization policy

2. *Drop*: when a new message arrives at a node with a full buffer, this node should drop the message with the smallest utility among the one just received and the buffered messages.

We will derive next such a per-message utility for two popular metrics: maximizing the average delivery probability (rate), and minimizing the average delivery delay. Table 5.1 contains some useful notation that we will use throughout the paper. Finally, the GBSD optimization policy is summarized in Figure 3.1.

3.1.2 Maximizing the average delivery rate

We first look into a scenario where each message has a *finite* TTL value. The source of the message keeps a copy of it during the whole TTL duration, while intermediate nodes are not obliged to do so. To maximize the average delivery probability among all messages in the network, the optimal policy must use the per message utility derived in the following theorem, in order to perform scheduling and buffer management.

Theorem 3.1.1 *Let us assume that there are K messages in the network, with elapsed time T_i for the message i. For each message $i \in [1, K]$, let $n_i(T_i)$ be the number of nodes who have a copy of the message at this time instant, and $m_i(T_i)$ those that have "seen" the message (excluding the source) since its creation*[3] *($n_i(T_i) \leqslant m_i(T_i) + 1$). To maximize the average delivery rate of all messages, a DTN node*

[3] We say that a node A has "seen" a message i, when A had received a copy of message i in the past, regardless of whether it still has the copy or has already removed it from its buffer.

Table 3.1: Notation

Variable	Description
L	Number of nodes in the network
K(t)	Number of distinct messages in the network at time t
TTL_i	Initial Time To Live for message i
R_i	Remaining Time To Live for message i
$T_i = TTL_i$ - R_i	Elapsed Time for message i. It measures the time since this message was generated by its source
$n_i(T_i)$	Number of copies of message i in the network after elapsed time T_i
$m_i(T_i)$	Number of nodes (excluding source) that have *seen* message i since its creation until elapsed time T_i
λ	Meeting *rate* between two nodes; $\lambda = \frac{1}{E[H]}$ where $E[H]$ is the average meeting time

should apply the GBSD policy using the following utility per message i:

$$U_i(DR) = (1 - \frac{m_i(T_i)}{L-1})\lambda R_i \exp(-\lambda n_i(T_i) R_i). \tag{3.1}$$

The probability that a copy of a message i will not be delivered by a node is given by the probability that the next meeting time with the destination is greater than R_i, the remaining lifetime of a message ($R_i = TTL - T_i$). This is equal to $\exp(-\lambda R_i)$ under our assumptions.

Knowing that message i has $n_i(T_i)$ copies in the network, and assuming that the message has not yet been delivered, we can derive the probability that the message itself will not be delivered (i.e. none of the n_i copies gets delivered):

$$P\{\text{message } i \text{ not delivered} \mid \text{not delivered yet}\} = \prod_{k=1}^{n_i(T_i)} \exp(-\lambda R_i) = \exp(-\lambda n_i(T_i) R_i). \tag{3.2}$$

We need to also take into consideration what has happened in the network since the message generation, in the absence of an explicit delivery notification (this part is not considered in RAPID [60], making the utility function derived there suboptimal). Given that all nodes including the destination have the same chance to see the message, the probability that a message i has been already delivered is equal to:

$$P\{\text{message } i \text{ already delivered}\} = m_i(T_i)/(L-1). \tag{3.3}$$

Combining Eq.(3.2) and Eq.(3.3), the probability that a message i will get delivered before its TTL expires is:

$$\begin{aligned} P_i &= P\{\text{message } i \text{ not delivered yet}\} * (1 - \exp(-\lambda n_i(T_i) R_i)) \\ &\quad + P\{\text{message } i \text{ already delivered}\} \\ &= (1 - \frac{m_i(T_i)}{L-1}) * (1 - \exp(-\lambda n_i(T_i) R_i)) + \frac{m_i(T_i)}{L-1}. \end{aligned}$$

So, if we take at instant t a snapshot of the network, the global delivery rate for the whole network will be:

$$DR = \sum_{i=1}^{K(t)} \left[(1 - \frac{m_i(T_i)}{L-1}) * (1 - \exp(-\lambda n_i(T_i) R_i)) + \frac{m_i(T_i)}{L-1} \right].$$

In case of a full buffer or limited transfer opportunity, a DTN node should take respectively a drop or replication decision that leads to the best gain in the global delivery rate DR. To define this optimal decision, we differentiate DR with respect to $n_i(T_i)$,

$$\begin{aligned} \Delta(DR) &= \sum_{i=1}^{K(t)} \frac{\partial P_i}{\partial n_i(T_i)} * \triangle n_i(T_i) \\ &= \sum_{i=1}^{K(t)} \left[(1 - \frac{m_i(T_i)}{L-1}) \lambda R_i \exp(-\lambda n_i(T_i) R_i) * \triangle n_i(T_i) \right] \end{aligned}$$

Our aim is to maximize $\Delta(DR)$. In the case of message drop, for example, we know that: $\Delta n_i(T_i) = -1$ if we drop an already existing message i from the buffer, $\Delta n_i(T_i) = 0$ if we don't drop an already existing message i from the buffer, and $\Delta n_i(T_i) = +1$ if we keep and store the newly-received message i. Based on this, GBSD ranks messages using the per message utility in Eq.(3.1), then schedules and drops them accordingly. This utility can be viewed as the *marginal utility value* for a copy of a message i with respect to the total delivery rate. The value of this utility is a function of the global state of the message i (n_i and m_i) in the network.

As is evident from the above description, the GBSD policy is a greedy, locally optimal policy. However, greedy policies in general, are not guaranteed to converge to globally optimal outcomes. We will investigate the optimality properties of GBSD further in Section 3.1.5.

3.1.3 Minimizing the average delivery delay

We next turn our attention to minimizing the average delivery delay. We now assume that all messages generated have infinite TTL or at least a TTL value large enough to ensure a delivery probability close to 1. The following Theorem derives the optimal per-message utility, for the same setting and assumptions as Theorem 3.1.1.

Theorem 3.1.2 *To minimize the average delivery delay of all messages, a DTN node should apply the GBSD policy using the following utility for each message i:*

$$U_i(DD) = \frac{1}{n_i(T_i)^2\lambda}(1 - \frac{m_i(T_i)}{L-1}). \tag{3.4}$$

Let us denote the delivery delay for message i with random variable X_i. This delay is set to 0 (or any other constant value) if the message has been already delivered. Then, the total expected delivery delay (DD) for all messages for which copies still exist in the network is given by,

$$DD = \sum_{i=1}^{K(t)} \left[\frac{m_i(T_i)}{L-1} * 0 + (1 - \frac{m_i(T_i)}{L-1}) * E[X_i|X_i > T_i]\right]. \tag{3.5}$$

We know that the time until the first copy of the message i reaches the destination follows an exponential distribution with mean $1/(n_i(T_i)\lambda)$. It follows that,

$$E[X_i|X_i > T_i] = T_i + \frac{1}{n_i(T_i)\lambda}. \tag{3.6}$$

Substituting Eq.(3.6) in Eq.(3.5), we get,

$$DD = \sum_{i=1}^{K(t)}(1 - \frac{m_i(T_i)}{L-1})(T_i + \frac{1}{n_i(T_i)\lambda}).$$

Now, we differentiate D with respect to $n_i(T_i)$ to find the policy that maximizes the improvement in D,

$$\Delta(DD) = \sum_{i=1}^{K(t)} \frac{1}{n_i(T_i)^2\lambda}(\frac{m_i(T_i)}{L-1} - 1) * \Delta n_i(T_i).$$

The best drop or forwarding decision will be the one that maximizes $|\Delta(DD)|$ (or $-\Delta(DD)$). This leads to the per message utility in Eq.(3.4).

Note that, the per-message utility with respect to delivery delay is different than the one for the delivery rate. This implies (naturally) that both metrics cannot be optimized concurrently.

3.1.4 The Case of Non-Homogeneous Mobility

Throughout our analysis, we have so far assumed homogeneous node mobility. Recent measurement studies have revealed that, often, different node pairs might have different meeting rates. We extend here our analytical framework, in order to derive per-message utilities that maximize the global performance metric, in face of such heterogeneous mobility scenarios. We illustrate the extension with the delivery rate[4]. Specifically, we assume that meetings between a given node pair are

[4]The treatment of delivery delay utilities does not involve Laplace transforms, but poses no extra difficulties. We thus omit it here, due to space limitations

exponentially distributed with meeting rate $\tilde{\lambda}$, where $\tilde{\lambda}$ is a *random variable* such that:

$$\tilde{\lambda} \in [0, \infty), \text{distributed as } f(\tilde{\lambda}).$$

$f(\tilde{\lambda})$ is a probability distribution that models the heterogeneous meeting rates between nodes, and can be any function integrable in $[0, \infty)$, capturing thus a very large range of conceivable mobility models.

The analysis of Theorem 3.1.2 is thus modified as follows. Let's assume that message i has n_i copies in the network, and that the n_i carriers have (unknown) meeting rates $\tilde{\lambda_1}, \tilde{\lambda_2}, \ldots, \tilde{\lambda_{n_i}}$, respectively. Eq.(3.2) becomes:

$$P\{\text{message } i \text{ not delivered} \mid \text{not delivered yet}\} =$$

$$E_{\tilde{\lambda_1},\tilde{\lambda_2},\ldots,\tilde{\lambda_{n_i}}}[\prod_{j=1}^{n_i} \exp(-\tilde{\lambda_j} R_i)] = \tag{3.7}$$

$$\prod_{j=1}^{n_i} \int_0^{\infty} \exp(-\tilde{\lambda_j} R_i) f(\tilde{\lambda_j}) d\lambda_j = (F_{\mathcal{L}}(R_i))^{n_i}, \tag{3.8}$$

where $F_{\mathcal{L}}(R_i)$ is the Laplace transform of distribution $f(x)$ evaluated at R_i. Continuing as in the proof of Theorem 3.1.2, we get the unconditional probability of delivery P_i:

$$P_i = (1 - \frac{m_i}{L-1}) * (F_{\mathcal{L}}(R_i))^{n_i} + \frac{m_i}{L-1}.$$

Differentiating P_i with respect to n_i, we derive the following generic marginal utility per message:

$$(1 - \frac{m_i}{L-1}) * \ln(F_{\mathcal{L}}(R_i)) * (F_{\mathcal{L}}(R_i))^{n_i}. \tag{3.9}$$

We now consider some example distributions for node meeting rates, and derive the respective marginal utility.

Dirac delta funtion: Let $f(\tilde{\lambda}) = \delta(\tilde{\lambda} - \lambda)$, where $\delta(x)$ is an impulse function (Dirac's delta function). This corresponds to the case of homogeneous mobility, considered earlier, with average meeting rates for all nodes equal to λ. The laplace distribution of $f(\tilde{\lambda})$ is then equal to $F_{\mathcal{L}}(R_i) = \exp(-\lambda R_i)$. Replacing this in Eq.(3.9), the generic marginal utility, gives us Eq.(3.1), the utility for homogeneous mobility, as expected.

Exponential distribution: Let $f(\tilde{\lambda}) = \lambda_0 \exp(-\tilde{\lambda}\lambda_0)$, for $\tilde{\lambda} \geq 0$. This corresponds to a mobility model, where individual rates between pairs differ, but the variance of these rates is not high and their average is equal to λ_0. The laplace transform of $f(\tilde{\lambda})$ is

$$F_{\mathcal{L}}(R_i) = \frac{1}{(R_i + \lambda_0)^2}.$$

Replacing this in Eq.(3.9) gives us the marginal utility per message that should be used:

$$(1 - \frac{m_i}{L-1}) * \ln(\frac{1}{(R_i + \lambda_0)^2}) * \frac{1}{(R_i + \lambda_0)^{2n_i}}. \tag{3.10}$$

Unknown distribution in large networks: If the actual probability distribution of meeting rates is not known, the following approximation could be made in order to derive marginal utilities per message and use them for buffer management. Let us assume that the meeting rates come from an unknown distribution with first and second moments $\bar{\lambda}$ and σ^2, respectively. Let us further assume that there is a large number of nodes, such that n_i, the number of copies of message i at steady state, is large. Using the central limit theorem, we have:

$$Prob(\sum_{j=1}^{n_i} \tilde{\lambda}_j \leq \lambda) \underset{n_i \to \infty}{\sim} \mathcal{N}(n_i\bar{\lambda}, \sigma\sqrt{n_i}), \tag{3.11}$$

that is, the sum of meeting rates with the destination of the n_i relays for message i is (approximately) normally distributed. Replacing this in Eq.(3.8), we get the (unconditional) delivery probability P_i

$$P_i = (1 - \frac{m_i}{L-1}) * F_{\mathcal{L}}(R_i) + \frac{m_i}{L-1},$$

where $F_{\mathcal{L}}(R_i)$ is the Laplace transform of the above normal distribution[5]. After some algebraic manipulations we can get the new marginal utility for message i:

$$(1 - \frac{m_i}{L-1}) * \frac{(\bar{\lambda}^2\sqrt{8}(n_i)^{-\frac{1}{2}} + \sqrt{2}\sigma^2(n_i)^{-\frac{5}{2}})\exp(n_i\frac{\bar{\lambda}^2}{\sigma^2} + \frac{R_i^2}{4})}{8\sigma^4} * \operatorname{erfc}(\frac{R_i}{2}). \tag{3.12}$$

In a large enough network, even if the actual distribution of meeting rates is not known, a node could still derive good utility approximations, by measuring and maintaining an estimate for the first and second moments of observed or reported meeting rates (e.g. with techniques similar to the ones discussed in the next Section). Furthermore, the homogeneous assumption could be considered as a useful approximation for large networks where the common rate is taken as $\bar{\lambda}$. Additional complexity in the mobility model (e.g. correlated meeting rates) could still be handled in our framework, yet at the expense of ease of interpretation (and thus usefulness) of the respective utilities. We will therefore consider the simple case of homogeneous mobility for the remainder of our discussion, in order to better elucidate some additional key issues related to buffer management in DTNs, and resort to a simulation-based validation under realistic mobility patterns.

3.1.5 Optimality of Gradient Ascent Policy

We finally turn our attention back to the distributed (*local*) buffer management policies of Sections 3.1.2 and 3.1.3, in order to further investigate their optimality. Let us observe our network at a random time instant, and assume there are K total undelivered messages, with remaining Times To Live $R_1, R_2, \ldots, R_K$, respectively. The centralized version of our buffer management problem then consists of assigning the available buffer space across the network (L nodes each able to store B message

[5]Note that the Laplace transform is not raised anymore to the n_i^{th} power, as the distribution already corresponds to the sum of all rates.

copies) among the copies of these messages, $n_1, n_2, \ldots, n_K$, so as to maximize the expected delivery probability for all these messages (where the expectation is taken over mobility decisions of all nodes). This corresponds to the following optimization problem:

$$\max_{n_1,n_2,\ldots,n_K} \sum_{i=1}^{K} (1 - \exp(-\lambda n_i R_i)) \tag{3.13}$$

$$\sum_{i=1}^{K} n_i - LB \leq 0 \tag{3.14}$$

$$n_i - L \leq 0, \forall i \tag{3.15}$$

$$n_i \geq 1, \forall i \tag{3.16}$$

This is a constrained optimization problem, with K variables and $2K + 1$ inequality constraints. The optimization function in Eq.(3.13) is a concave function in n_i. Constraint in Eq.(3.14) says that the total number of copies (for all messages) should not exceed the available buffer space in all L nodes, and is linear. Finally, the $2K$ constraints of Eq.(3.15) are also linear, and simply say that there is no point for any node to store two copies of the same message. Consequently, if we assume that n_i are real random variables (rather than integers), this is a *convex optimization* problem, which can be solved efficiently [82] (but not easily analytically).

Having found an optimal vector $\mathbf{n}$, a centralized optimal algorithm can easily assign the copies to different nodes (e.g. picking nodes sequentially and filling their buffers up with any non-duplicate copy, starting from the messages with highest assigned n_i — due to uniform mobility the choice of specific nodes does not matter). It is important to note that, given this assignment, *no further message replication or drop is needed. This is the optimal resource allocation averaged over all possible future node movements.* The optimal algorithm must perform the same process at every subsequent time step in order to account for new messages, messages delivered, and the smaller remaining times of undelivered messages.

Our local policies offer a *distributed implementation of a gradient ascent algorithm for this problem.* Gradient ascent algorithms look at the current state, i.e. vector $\mathbf{n}(k)$ at step k, and choose a neighboring vector $\mathbf{n}(k+1)$ that improves the optimization function in Eq.(3.13), and provably converge to the optimal solution [82]. In our case, a step corresponds to a *contact* between two nodes, and the neighboring states and permitted transitions depend on the messages in the buffers of the two nodes in contact. In other words, our gradient ascent algorithm is supposed to make enough steps to converge to the optimal copy vector $\mathbf{n}^*$, before the state of the network (i.e. number and ID of messages) changes enough for the optimal assignment to change significantly. This depends on the rate of update steps ($\approx \lambda L^2$) and the message TTL. If $TTL * \lambda * L^2 \gg 1$, then we expect the distributed, local policy to be able to closely follow the optimal solution at any time t. In Section 3.3.4, we use simulation to prove that this is indeed the case for the scenarios considered.

3.2 Using Network History to Approximate Global Knowledge in Practice

It is clear from the above description that the optimal policy (GBSD) requires global information about the network and the ”spread” of messages, in order to optimize a specific routing metric. In particular, for each message present in a node's buffer, we need to know the values of $m_i(T_i)$ and $n_i(T_i)$. In related work [60], it has been suggested that this global view could be obtained through a secondary, “instantaneous” channel (e.g. cellular network), if available, or by flooding (“in-band”) all necessary meta-data. Regarding the former option, cellular network connections are known to be low bandwidth (measurements suggest only few kbps even for 2.5-3G technologies [83]) and high cost in terms of power and actual monetary cost per bit. In networks of more than a few nodes, the amount of signalling data might make this option prohibitive. Concerning flooding, our experiments show that the impact of the flooding delay on the performance of the algorithm is not negligible. In practice, intermittent network connectivity and the long time it takes to flood buffer status information across DTN nodes, make this approach inefficient.

A different, more robust approach is to find estimators for the unknown quantities involved in the calculation of message utilities, namely m and n. We do this by designing and implementing a learning process that permits a DTN node to gather knowledge about the global network state at different times in the past, by making in-band exchanges with other nodes. Each node maintains a list of encountered nodes and the state of each message carried by them as a function of time (i.e. its buffer state history). Specifically, it logs whether a given message was present at a given time T in a node's buffer (counting towards n) or whether it was encountered earlier but is not anymore stored, e.g. it was dropped (counting towards m). In Section 3.4, we describe our statistics maintenance and collection method, in more detail, along with various optimizations to considerably reduce the signalling overhead.

Since global information gathered thus about a specific message might take a long time to propagate (as mentioned earlier) and hence might be obsolete when we calculate the utility of the message, we follow a different route. Rather than looking for the current value of $m_i(T)$ and $n_i(T)$ *for a specific message* i at an elapsed time T, we look at what happens, *on average, for all messages after an elapsed time* T. In other words, the $m_i(T)$ and $n_i(T)$ values for message i at elapsed time T are estimated using measurements of m and n for the same elapsed time T but *measured for (and averaged over) all other older messages*. These estimations are then used in the evaluation of the per-message utility.

Let's denote by $\hat{n}(T)$ and $\hat{m}(T)$ the estimators for $n_i(T)$ and $m_i(T)$ of message i. For the purpose of the analysis, we suppose that the variables $m_i(T)$ and $n_i(T)$ at elapsed time T are instances of the random variables $N(T)$ and $M(T)$. We develop our estimators $\hat{n}(T)$ and $\hat{m}(T)$ so that when plugged into the GBSD's delivery rate and delay per-message utilities calculated in Section 3.1, we get two new per-message utilities that can be used by a DTN node without any need for global information about messages. This results in a new scheduling and drop policy, called HBSD (History Based Scheduling and Drop), a deployable variant of GBSD that uses the same algorithm, yet with per-message utility values calculated using estimates of m and n.

3.2.1 Estimators for the Delivery Rate Utility

When global information is unavailable, one can calculate the average delivery rate of a message over all possible values of $M(T)$ and $N(T)$, and then try to maximize it. In the framework of the GBSD policy, this is equivalent to choosing the estimators $\hat{n}(T)$ and $\hat{m}(T)$ so that the calculation of the average delivery rate is unbiased:

$$E[(1-\frac{M(T)}{L-1})*(1-\exp(-\lambda N(T)R_i))+\frac{M(T)}{L-1}] =$$
$$(1-\frac{\hat{m}(T)}{L-1})*(1-\exp(-\lambda\hat{n}(T)R_i))+\frac{\hat{m}(T)}{L-1}$$

Plugging any values for $\hat{n}(T)$ and $\hat{m}(T)$ that verify this equality into the expression for the per-message utility of Eq.(3.1), one can make sure that the obtained policy maximizes the average delivery rate. This is exactly our purpose. Suppose now that the best estimator for $\hat{m}(T)$ is its average, i.e., $\hat{m}(T) = \bar{m}(T) = E[M(T)]$. This approximation is driven by the observation we made that the histogram of the random variable $M(T)$ can be approximated by a Gaussian distribution with good accuracy. To confirm this, we have applied the Lillie test [84], a robust version of the well known Kolmogorov-Smirnov goodness-of-fit test, to $M(T)$ for different elapsed times (T = 25%,50% and 75% of the TTL). This test led to acceptance for a 5% significance level. Consequently, the average of $M(T)$ is at the same time the unbiased estimator and the most frequent value among the vector $M(T)$. Then, solving for $\hat{n}(T)$ gives:

$$\hat{n}(T) = -\frac{1}{\lambda R_i}\ln\left(\frac{E[(1-\frac{M(T)}{L-1})\exp(-\lambda N(T)R_i)]}{(1-\frac{\bar{m}(T)}{L-1})}\right) \tag{3.17}$$

Substituting this expression into Eq.(3.1) we obtain the following new per message utility for our approximating HBSD policy:

$$\lambda R_i E[(1-\frac{M(T)}{L-1})\exp(-\lambda R_i N(T))] \tag{3.18}$$

The expectation in this expression is calculated by summing over all known values of $N(T)$ and $M(T)$ for past messages at elapsed time T. Unlike Eq.(3.1), this new per-message utility is a function of past history of messages and can be calculated locally. It maximizes the average message delivery rate calculated over a large number of messages. When the number of messages is large enough for the law of large numbers to work, our history based policy should give the same result as that of using the real global network information.

Finally, we note that L, the number of nodes in the network, could also be calculated from the statistics maintained by each node in the network. In this work, we assume it to be fixed and known, but one could estimate it similar to n and m, or using different estimation algorithms like the ones proposed in [85].

3.2.2 Estimators for the Delivery Delay Utility

Similar to the case of delivery rate, we calculate the estimators $\hat{n}(T)$ and $\hat{m}(T)$ in such a way that the average delay is not affected by the estimation. This gives the following per-message utility specific to HBSD,

$$\frac{E[\frac{L-1-M(T)}{N(T)}]^2}{\lambda(L-1)(L-1-\bar{m}(T))} \tag{3.19}$$

This new per-message utility is only a function of the locally available history of old messages and is thus independent of the actual global network state. For large number of messages, it should lead to the same average delay as when the exact values for m and n are used.

3.3 Performance Evaluation

3.3.1 Experimental Setup

To evaluate our policies, we have implemented a DTN framework into the Network Simulator NS-2 [86]. This implementation includes (i) the Epidemic routing protocol with *FIFO* for scheduling messages queued during a contact and *drop-tail* for message drop, (ii) the RAPID routing protocol based on flooding (i.e. no side-channel) as described, to our best understanding, in [60], (iii) a new version of Epidemic routing enhanced with our optimal joint scheduling and drop policy (GBSD), (iv) another version using our statistical learning based distributed algorithm (HBSD), and (v) the VACCINE anti-packet mechanism described in [54][6].

In our simulations, each node uses the 802.11b protocol to communicate, with rate 11Mbits/sec. The transmission range is 100 meters, to obtain network scenarios that are neither fully connected (e.g. MANET) nor extremely sparse.

Our simulations are based on five mobility scenarios: two synthetic mobility models and three real-world mobility traces.

Synthetic Mobility Models: We've considered both the Random Waypoint mobility model and the HCMM model [67]. The later is inspired from Watts' Caveman

[6]We have also performed simulations without any anti-packet mechanism, from which similar conclusions can be drawn.

model that was shown to reproduce statistics of human mobility, such as inter-contact times and contact duration.

Real Mobility Traces: The first *(i)* real trace is the one collected as part of the ZebraNet wildlife tracking experiment in Kenya and described in [87]. The second mobility trace tracks San Francisco's Yellow Cab taxis. Many cab companies outfit their cabs with *GPS* to aid in rapidly dispatching cabs to their costumers. The Cabspotting system [88] talks to the Yellow Cab server and stores the data in a database. We have used an API provided by the Cabspotting system in order to extract mobility traces[7]. And finally, the third *(iii)* trace consists on the KAIST real mobility trace collected from a university campus (KAIST) in South Korea [89]. We consider a sample of the KAIST campus trace taken from 50 students, where the GPS receivers log their position at every 30 seconds.

To each source node, we have associated a CBR (Constant Bit Rate) application, which chooses randomly from $[0, TTL]$ the time to start generating messages of $5KB$ for a randomly chosen destination. We have also considered other message sizes (see e.g. [90]), but found no significant differences in the qualitative and quantitative conclusions drawn regarding the relative performance of different schemes[8]. Unless otherwise stated, each node maintains a buffer with a capacity of 20 messages to be able to push the network towards a congested state without exceeding the processing and memory capabilities of our simulation cluster. We compare the performance of the various routing protocols using the following two metrics: the average delivery rate and average delivery delay of messages in the case of infinite TTL[9]. Finally, the results presented here are averages from 20 simulation runs, which we found enough to ensure convergence.

3.3.2 Performance evaluation for delivery rate

First, we compare the delivery rate of all policies for the three scenarios shown in Table 3.2.

Figure 3.2 shows the delivery rate based on the Random Waypoint model. From this plot, it can be seen that: the GBSD policy plugged into Epidemic routing gives the best performance for all numbers of sources. When congestion-level decreases, so does the difference between GBSD and other protocols, as expected. Moreover, the HBSD policy also outperforms existing protocols (RAPID and Epidemic based on FIFO/drop-tail) and performs very close to the optimal GBSD. Specifically, for 70 sources, HBSD offers an almost 60% improvement in delivery rate compared to

[7] Note that this trace describes taxi's positions according to the *GPS* cylindrical coordinates (*Longitude*, *Latitude*. In order to uses these traces as input for the NS-2 simulator, we have implemented a tool [86] based on the Mercator cylindrical map projection which permit us to convert traces to plane coordinates.

[8] In future work, we intend to evaluate the effect of variable message size and its implications for our optimization framework. In general, utility-based scheduling problems with variable sized messages can often be mapped to Knapsack problems (see e.g. [91]).

[9] By infinite TTL, we mean any value large enough to ensure almost all messages get delivered to their destination before the TTL expires.

Table 3.2: Simulation parameters

Mobility pattern:	RWP	ZebraNet	Taxis	KAIST	HCMM
Sim. Duration(h):	7	14	42	24	24
Sim. Area (km^2):	3*3	3*3	-	-	5*5
Nbr. of Nodes:	70	70	70	50	70
Avg. Speed (m/s):	2	-	-	-	-
$TTL(h)$:	1	2	6	4	4
CBR Interval(s):	360	720	2160	1440	1440

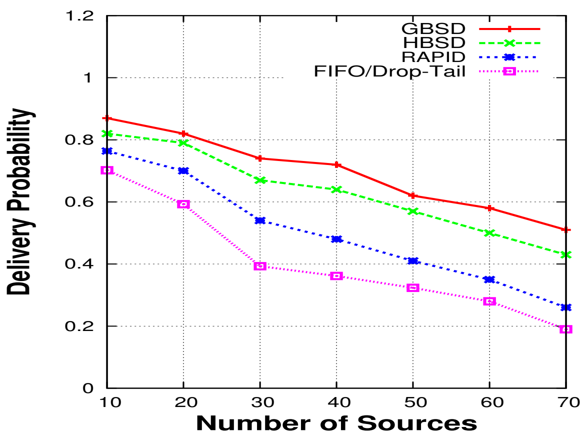

Figure 3.2: Delivery Probability for Epidemic Routing with different scheduling and drop policies (both buffer and bandwidth constraints).

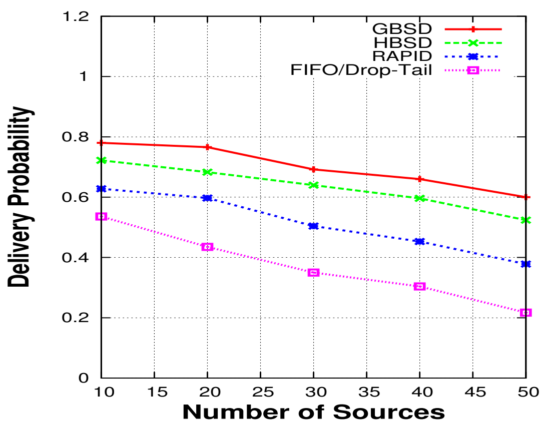

Figure 3.3: Delivery Probability (KAIST mobility trace).

RAPID and is only 14% worse than GBSD. Similar conclusions can be also drawn for the case of the real Taxi trace, ZebraNet trace, KAIST trace or the HCMM model and 70 sources. Results for these cases are respectively summarized in Table 3.3, Table 3.4, Figure 3.3 and Table 3.5.

Table 3.3: Taxi Trace & Limited buffer and bandwidth

Policy:	GBSD	HBSD	RAPID	FIFO\DT
D. Probability:	0.72	0.66	0.44	0.34
D. Delay(s):	14244	15683	20915	36412

Table 3.4: ZebraNet Trace & Limited buffer and bandwidth

Policy:	GBSD	HBSD	RAPID	FIFO\DT
D. Probability:	0.68	0.59	0.41	0.29
D. Delay(s):	4306	4612	6705	8819

Table 3.5: HCMM Trace (70 CBR sources)

Policy:	GBSD	HBSD	RAPID	FIFO\DT
D. Probability:	0.62	0.55	0.38	0.23
D. Delay(s):	3920	4500	6650	8350

3.3.3 Performance evaluation for delivery delay

To study delays, we increase messages' TTL (and simulation duration), to ensure almost every message gets delivered, as follows. Random Waypoint: (duration 10.5h, TTL = 1.5h). ZebraNet: (simulation duration = 28h, TTL = 4h). Taxi trace: (simulation duration = 84h, TTL = 12h). Traffic rates are as in Section 3.3.2.

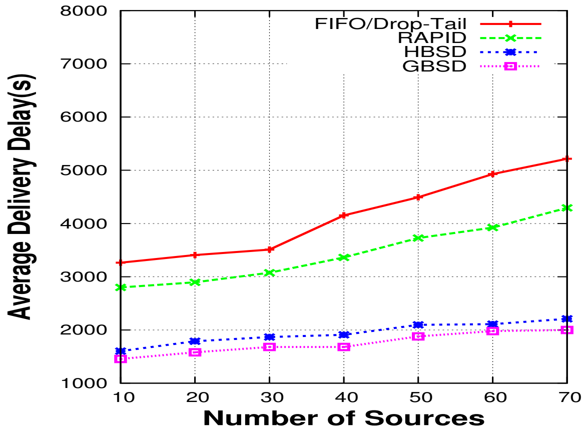

Figure 3.4: Delivery Delay for Epidemic Routing with different scheduling and drop policies (both buffer and bandwidth constraints).

For the random waypoint mobility scenario, Figure 3.4 depicts the average delivery delay for the case of both limited buffer and bandwidth. As in the case of delivery rate, GBSD gives the best performance for all considered scenarios. Moreover, the HBSD policy outperforms the two routing protocols (Epidemic based on FIFO/drop-tail, and RAPID) and performs close to GBSD. Specifically, for 70 sources and both limited buffer and bandwidth, HBSD average delivery delay is 48% better than

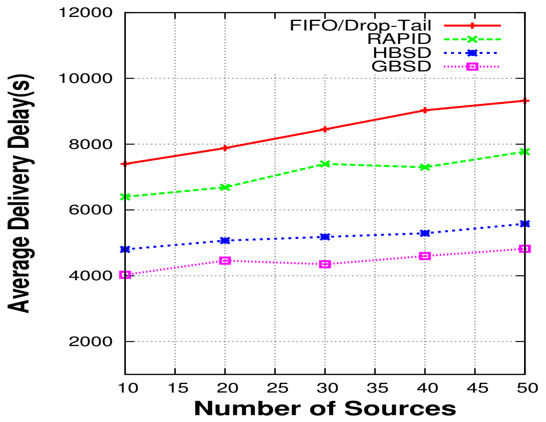

Figure 3.5: Delivery Delay (KAIST mobility trace).

RAPID and only 9% worse than GBSD.

Table 3.3, Table 3.4, Figure 3.5 and Table 3.5 show that similar conclusions can be drawn for the delay under respectively the real Taxi(s), ZebraNet trace, KAIST trace and the HCMM model.

3.3.4 Optimality

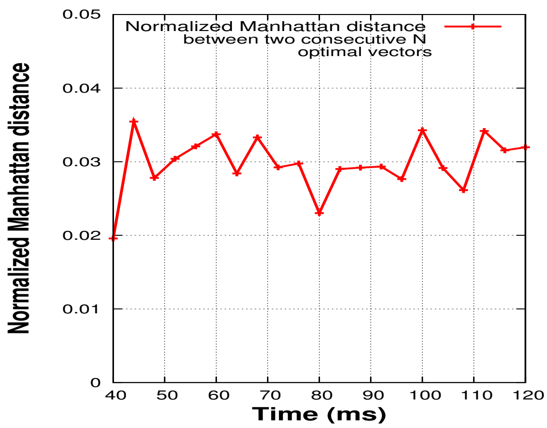

Figure 3.6: Normalized Manhattan distance between two consecutive N optimal vectors.

Here, we use simulations results (based on the RW scenario) that our proposed policy (GBSD) can "keep up" with the optimal algorithm described in Section 3.1.5. Fig. 3.6 plots the normalized Manhattan distance $d(X,Y) = \frac{\sum_{i=1}^{K}|x_i - y_j|}{K}$ between two consecutive optimal copy vectors, resulting from solving the optimal centralized version offline. These optimal vectors are calculated every $4ms$, corresponding to the average time between any two consecutive contacts among the network. As is evident in the figure, this distance is very small, implying that our distributed gradient-

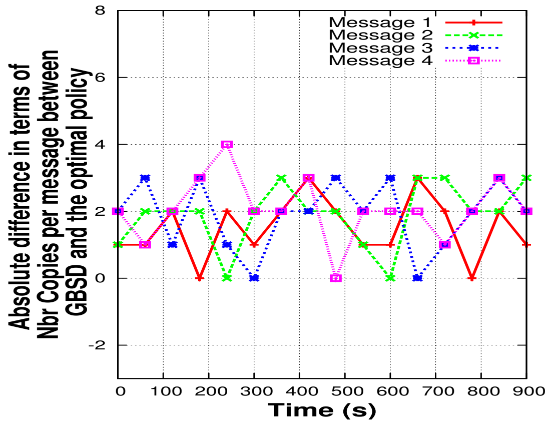

Figure 3.7: Difference in terms of Nbr of copies.

ascent implementation of this policy (GBSD/HBSD) has enough time to converge to the optimal vector, before this changes significantly. In order to further validate the optimality of our policy, we compare in Fig. 3.7 the absolute difference between the number of copies assigned to a message by our GBSD policy and the number of copies allocated to the same message by the optimal algorithm 3.1.5. We have picked some messages randomly and plot this difference along a time window in their lifetime. These results show that the GBSD policy stays is able to follow the optimal one with an average error of $1-2$ copies allocated at most. We believe this result consolidates the optimality properties of our proposed distributed implementation of the optimal policy.

3.4 Maintaining Network History

The results of the previous section clearly show that our distributed policy (HBSD) that uses estimators of global message state (rather than actual state) successfully approximates the performance of the optimal policy (GBSD). This is as an important step towards a practical implementation of efficient buffer management and scheduling algorithms on wireless devices. Nevertheless, in order to derive good estimators in a distributed manner, nodes need to exchange (a possibly large amount of) metadata during every node meeting. Potentially, each node needs to know the history of all messages having passed through a node's buffer, for every node in the network. In a small network, the amount of such "control" data might not be much, considering that large amounts of data transfers can be achieved between 802.11 transceivers during short contacts (data transfers of a few 10s of MBytes have been reported for experiments between vehicles moving at high speeds [92]). Nevertheless, in larger networks, this method can quickly become unsalable and interfere with (or starve) data transmissions, if statistics maintenance and collection is naively done.

In this section, we describe the type of statistics each node maintains towards calculating the HBSD utility for each message, and propose a number of mechanisms and optimizations to significantly reduce (and control) the amount of metadata exchanged during contacts. Finally, we explore the impact of reducing the amount of collected statistics on the performance of our buffer management and scheduling policy. Our results suggest that, with a carefully designed statistics collection and maintenance scheme, order(s) of magnitude less metadata can be exchanged (compared to maintaining a complete view about the network), without significantly affecting performance.

3.4.1 Maintaining Buffer State History

In order to keep track of the statistics about past messages necessary to assign appropriate utility values to messages considered for transmission or dropping, we propose that each node maintains the data structure depicted in Figure 3.8. Each node maintains a list of messages whose history in the network it keeps track of (we will see in the next section how a node chooses which messages to include in this list). For each message, it maintains its ID (a unique string resulting from the combination of some of its attributes), its TTL and the list of nodes that have *seen* it before (i.e. had stored the messages at some time in the past and should be accounted towards calculating m or n). Then, for each of the nodes in the list, it maintains a data structure with the following data: (i) the node's ID, (ii) a boolean array $Copies\_Bin\_Array$, and (iii) the version $Stat\_Version$ associated to this array.

The $Copies\_Bin\_Array$ array (Fig. 3.9) enables nodes to maintain dynamically what each message experienced during its life time. For a given entry pair (message a and node b) in this list, the $Copies\_Bin\_Array[k]$ indicates if the node a had already stored or not a copy of message b in its buffer during $Bin\ k$. In other words, time is quantized into "bins" of size Bin_Size, and bin k correspond to the period of time between $k * Bin\_Size$ and $(k+1) * Bin\_Size$. As a result, the size of the $Copies\_Bin\_Array$ is equal to TTL/Bin_Size.

How should one choose Bin_Size? Clearly, the larger it is, the fewer the amount of data a node needs to maintain and to exchange during each meeting; however, the smaller is also the granularity of values the utility function can take and thus the higher the probability of an incorrect (scheduling or buffer management) decision. As already described in Section 3.1, message transmissions can occur only when nodes encounter each other. This is also the time granularity at which buffer state changes occur. Hence, we believe that a good trade-off is to monitor the evolution of each message's state at a bin granularity in the order of meeting times[10]. This *already results in a big reduction of the size of statistics to maintain locally* (as opposed to tracking messages at seconds or milliseconds granularity), while still enabling us to infer the correct messages statistics.

Finally, the $Stat\_Version$ indicates the Bin at which the last update occurred.

[10]According to the *Nyquist-Shannon* [93] sampling theorem, a good approximation of the size of a *Bin* would be equal to inter-meeting-time/2. A running average of the observed times between consecutive meetings could be maintained easily, in order to dynamically adjust the bin size [76].

Let's assume that a message a is first stored at a node b during bin 3. It then creates a new entry in its list for pair (a,b), inserts 0s in bins $0-2$ of the new $Copies\_Bin\_Array$ and 1s in the rest of the bins, and sets the $Stat\_Version$ to 3. If later, at in bin 5 node b decides to drop this message, then the list entry is maintained, but it sets all bins from 5 to TTL/Bin_Size to 0, and updates the $Stat\_Version$ to 5. Finally, when the TTL for message a elapses (regardless of whether a is still present in b's buffer or not), b sets the $Stat\_Version$ to TTL/Bin_Size, which also indicates that all information about the history of *this* message in *this* buffer is now available. The combination of how the $Copies\_Bin\_Array$ is maintained and the $Stat\_Version$ updated, ensures that only the minimum amount of necessary metadata *for this pair of (message, node)* is exchanged during a contact.

We note also that, in principle, a $Message\_Seen\_Bin\_Array$ could be maintained, indicating if a node a had *seen* (rather than *stored* a message b at time t, in order to estimate $m(T)$. However, it is easy to see that the $Message\_Seen\_Bin\_Array$ can be deduced directly from the $Copies\_Bin\_Array$, and thus no extra storage is required. Summarizing, based on this lists maintained by all nodes, any node can retrieve the vectors $N(T)$ and $M(T)$ and can calculate the HBSD per-message utilities described in Section 3.2 without a need for an *oracle*.

3.4.2 Collecting Network Statistics

We have seen so far what types of statistics each node maintains about each past (message ID, node ID) tuple it knows about. Each node is supposed to keep up-to-date the statistics related to the messages it stores locally (i.e. entries in the list of Fig. 3.8 corresponding to its own node ID). However, it can only update its knowledge (and the respective entry) about the state of a message a at a node b when it either meets b directly, or it meets a node that has more recent information about the (a, b) tuple (i.e. a higher $Stat\_Version$). The goal of the statistics collection method is that, through such message exchanges, nodes converge to a unified view about the state of a given message at *any* buffer in the network, during its lifetime.

Sampling Messages to Keep Track of: We now look in more detail into what kind of metadata nodes should exchange. The first interesting question is the following: *should a node maintain global statistics for* every *message it has heard of or only a subset?* We argue that monitoring a dynamic subset of these messages is sufficient to quickly[11] converge to the correct expectations we need for our utility estimators. This dynamic subset is illustrated in Figure 3.10 as being the Messages Under Monitoring, which are stored in the *MUM* buffer; it is dynamic because its size is kept fixed while messages inside it change. When a node decides to store a

[11] While speed of convergence is not that important, due to our history-based approach, it becomes significant in non-stationary scenarios with traffic load fluctuations and node churn, as we shall see.

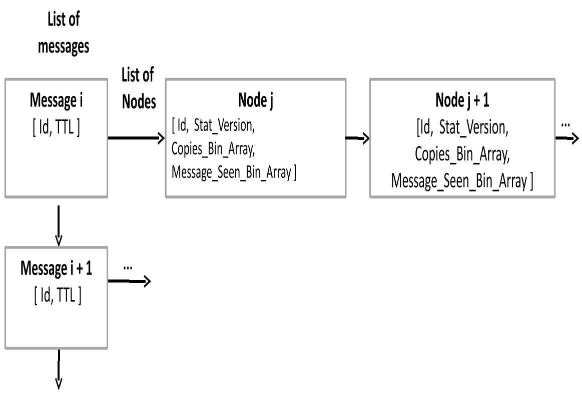

Figure 3.8: Network History Data Structure

message for the first time, if there is space in its MUM buffer, it also inserts it there and will track its global state. In other words, each node randomly chooses a few messages it will *sample*, for which it will attempt to collect global state, and does not keep track of all messages currently alive in the network. The actual *sampling rate* depends on the size of the MUM buffer and the offered traffic load, and results in significant further reduction in the amount of metadata exchanged. At the same time, a smaller MUM buffer might result to slower convergence (or even lack of). In Section 3.4.3 we study the impact of MUM buffer size on the performance of our algorithm.

Handling Converged Messages: Once the node collects an entire history of a given message, it removes it from the MUM buffer and pushes it to the buffer of Messages with a Complete History (MCH). A node considers that it has the complete history of a given message only when it gets the last version (i.e. $Stat\_Version = TTL/Bin\_Size$) of the statistics entries related to all the nodes the message goes through during its TTL[12] Finally, note that, once a node decides to move a message to the MCH buffer, it only needs to maintain a short summary (i.e. number of nodes with a copy $n(T)$ and number of nodes having seen the message, $m(T)$, at time T) rather than the per node state as in Fig. 3.8.

Statistics Exchanged: Once a contact opportunity is present, both peers have to ask only for newer versions of the statistics entries (message ID, node ID) related to the set of messages buffered in their MUM buffer. This ensures that, even for the sampled set of messages, only new information is exchanged and no bandwidth is wasted. This optimization does not introduce any extra latency in the convergence of our approximation scheme.

[12]Note that there is a chance that a node might "miss" some information about a message it pushes in its MCH. This occurs, for example, if it receives the last version for a subset of nodes which had the message, before it receives *any* version from another node that also had the message. This probability depends on the statistics of the meeting time (first and second moment) and the TTL value. Nevertheless, for many scenarios of interest, this probability is small and it may only lead to slightly underestimating the m and n values.

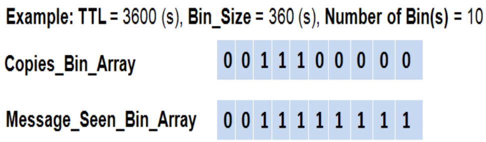

Figure 3.9: Example of Bin arrays

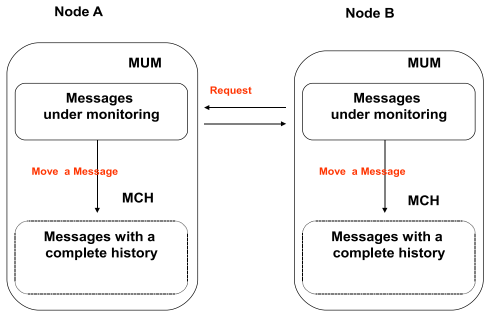

Figure 3.10: Statistics Exchange and Maintenance.

3.4.3 Performance Tradeoffs of Statistics Collection

We have presented a number of optimizations to (considerably) reduce the amount of metadata stored and the amount of signalling overhead. Here, we explore the tradeoff between the signalling overhead, its impact on performance, and the dynamicity of a given scenario. Our goal is to identify operation points where the amount of signalling overhead is such that it interferes minimally with data transmission, while at the same time it suffices to ensure timely convergence of the required utility metrics per message. We will consider throughout the random waypoint simulation scenario described in Section 3.3.2. We have observed similar behaviour for the trace-based scenarios.

Amount of Signalling Overhead per Contact: We start by studying the effect of varying the size of the *MUM* buffer (number of messages under monitoring) on the average size of exchanged statistics per-meeting. Figure 3.11 compares the average size of statistics exchanged during a meeting between two nodes for three different sizes of the *MUM* buffer (20, 50 and 80), as well as for the basic epidemic statistics exchange method (i.e. unlimited MUM). We vary the number of sources in order to cover different congestions regimes.

Our first observation is that increasing the traffic load (and thus the amount of congestion) results in decreasing the average amount of statistics exchanged per-meeting (except for the MUM size of 20 messages). This might be slightly counterintuitive, since a higher traffic load implies more messages to keep track of. However, note that a higher congestion level also implies that much fewer copies per message will co-exist at any time (and new versions are less frequently created). As a result, much less metadata per message is maintained and exchanged, resulting in a downward trend. In the case of a MUM size of 20, it seems that these two effects balance each other out. In any case, the key property here is that, in contrast with the flooding-based method of [60], *our distributed collection method scales well, not increasing the amount of signalling overhead during high congestion.*

A second observation is that, using our statistics collection method, a node can reduce the amount of signalling overhead per meeting up to an order of magnitude (e.g. for MUM = 20), compared to the unlimited MUM case, even in this relatively small scenario of 70 nodes. (Note also that, the plot shown for the epidemic case, already implements the binning and versioning optimizations of Section 3.4.1).)

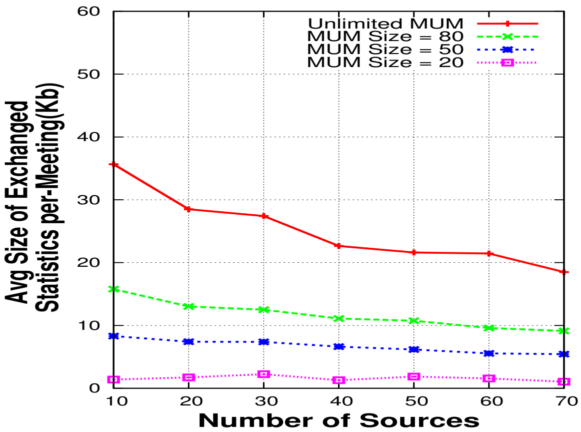

Figure 3.11: Signalling overhead (per contact) resulting from HBSD statistics collection.

Finally, we plot in Figure 3.12 the average size of exchanged (non-signalling) data per-meeting. We can observe that increasing the size of the *MUM* buffer results in a slight decrease of the data exchanged. This is due to the priority we give to statistics exchange during a contact. We note also that this effect becomes less pronounced when congestion increases (in line with Fig. 3.11). Finally, in the scenario considered, we can observe that, for MUM sizes less than 50, signalling does not interfere with data transmissions (remember that packet size is 5KB). This suggests that, in this scenario, a MUM size of 50 messages represents a good choice with respect to the resulting signalling overhead. In practice, a node could find this value online, by dynamically adjusting its MUM size and comparing the resulting signalling overhead with average data transfer. It is beyond the scope of this paper to propose such an algorithm. Instead, we are interested in exposing the various tradeoffs and choices involved in efficient distributed estimation of statistics. Towards this goal, we explore next the effect of the MUM sizes considered on the performance of our HBSD algorithm.

Convergence of Utilities and Performance of the HBSD Policy : In this last part, we fix the number of sources to 50 and we look at the impact of the size of the *MUM* buffer on (i) the time it takes the HBSD delivery rate utility to converge, and (ii) its accuracy. We use the *mean relative square error* to measure the accuracy of the HBSD delivery rate utility, defined as follows:

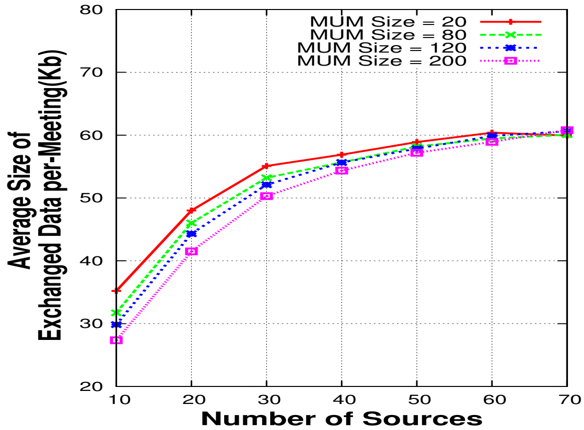

Figure 3.12: Average size of exchanged (non-signalling) data per contact.

$$\frac{1}{\#Bins} * \sum_{Bins} \frac{(A-B)^2}{B^2},$$

where, for each *bin*, A is the estimated utility value of Eq. (3.18) (calculated using the approximate values of m and n, collected with the method described previously) and B is the utility value calculated using the real values of m and n.

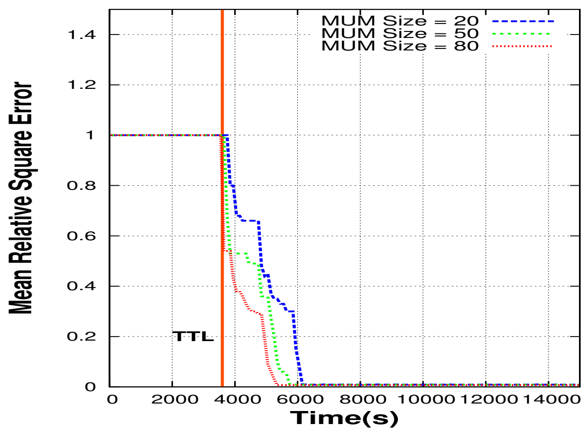

Figure 3.13: Mean relative square errors for HBSD delivery rate utility.

Figure 3.13 plots the *mean relative square errors* for the HBSD delivery rate utility, as a function of time. We can observe that, increasing the size of the *MUM* buffer results in faster reduction of the *mean relative square error* function. With a *MUM* buffer of 80 messages, the delivery rate utility estimate converges 800 seconds faster than using an *MUM* buffer of 20 messages. Indeed, the more messages a node tracks in parallel, the faster it can collect a working history of past messages that it can use to calculate utilities for new messages considered for drop or transmission.

We observe also that all plots converge to the same very small error value [13]. Note also that it is not the absolute value of the utility function (during different time bins) that we care about, but rather the *shape* of this function, whether it is increasing or decreasing, and the relative utility values. (We will look into the shape of this function at different congestion regimes in the next section.)

In fact, we are more interested in the end performance of our HBSD, as a function of how "aggressively" nodes collect message history. In Figures 3.14 and 3.15, we plot the delivery rate and delay of HBSD, respectively, for different MUM sizes. These results correspond to the scenario described in Section 3.3.2, where we have a fixed number of CBR sources. As is evident from these figures, regardless of the size of the *MUM* buffer sizes, nodes eventually gather enough past message history to ensure an accurate estimation of per message utilities, and a close-to-optimal performance. In such scenarios, where traffic intensity is relatively stable, even a rather small MUM size (i.e. very low sampling rate) suffices to achieve good performance. This is not necessarily the case when traffic load experiences significant fluctuations (e.g. due to new popular content appearing in the network).

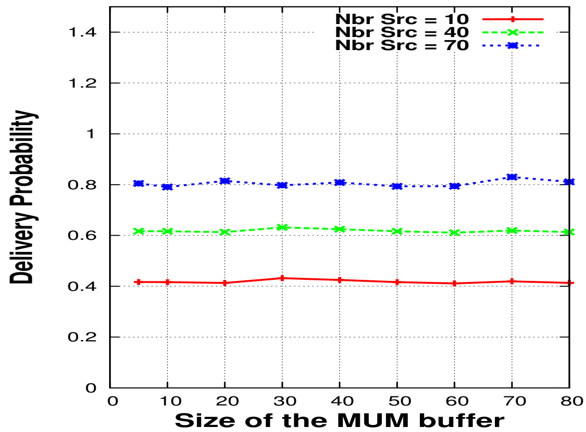

Figure 3.14: Delivery Probability for HBSD with statistics collection (static traffic load).

When the offered traffic load changes frequently (or node churn is high, e.g. experiencing "flash crowds"), convergence speed becomes important. The bigger the *MUM* buffer the faster our HBSD policy react to changing congestion levels. We illustrate this with the following experiment. We maintain the same simulation scenario, but we vary the number of CBR sources among each two consecutive TTL(s), from 10 to 70 sources (i.e. the first and second TTL window we have 10 sources, the third and fourth window 70 sources, etc. — this is close to a *worst case* scenario, as there is a sevenfold increase in traffic intensity within a time window barely higher than a TTL, which is the minimum required interval

[13] We speculate that this remaining error might be due to slightly underestimating m and n, as explained earlier.

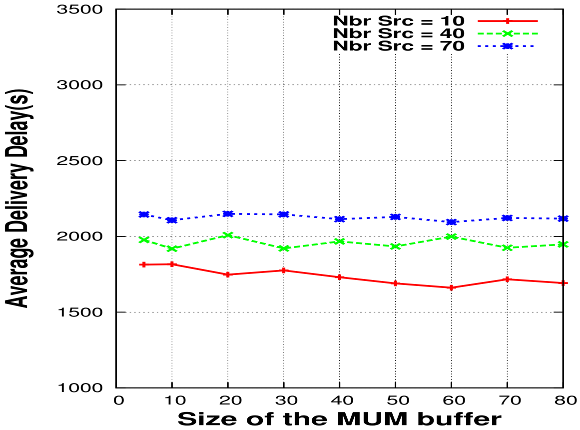

Figure 3.15: Deliver Delay for HBSD with statistics collection (static traffic load).

to collect any statistics). Furthermore, to ensure nodes use non-obsolete statistics towards calculating utilities, we force nodes to apply a *sliding window* of one *TTL* to the messages with complete history stored in the *MCH* buffer, and to delete messages out of this *sliding window*[14]

Figures 3.16 and 3.17 again plot the HBSD policy delivery rate and delay, respectively, as a function of MUM buffer size. Unlike the constant load case, it is easy to see there that, increasing the size of the *MUM* buffer, results in considerable performance improvement. Nevertheless, even in this rather dynamic scenario, nodes manage to keep up and produce good utility estimates, with only a modest increase on the amount of signalling overhead required.

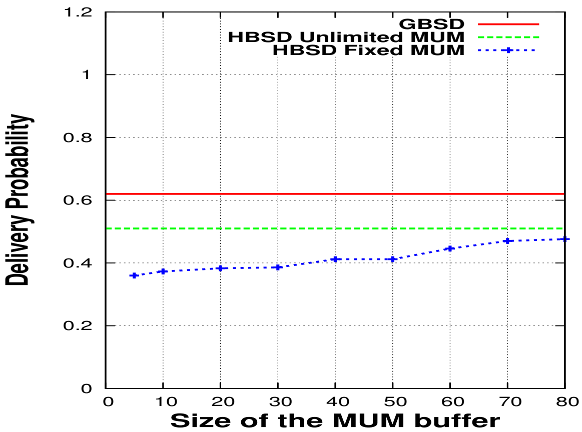

Figure 3.16: Deliver Probability for HBSD with statistics collection (dynamic traffic load).

[14] A running average could be used for smoother performance. We only care here to demonstrate the effect of dynamic traffic loads.

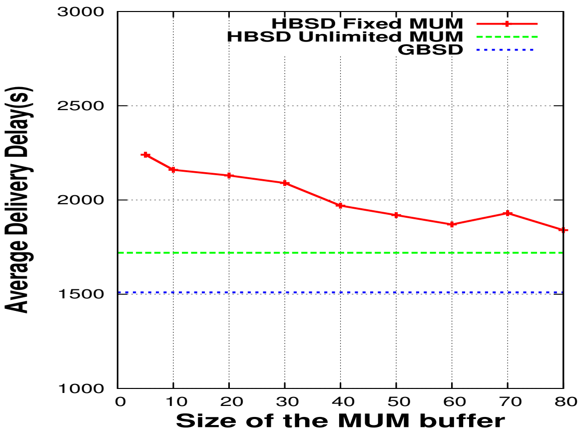

Figure 3.17: Deliver Delay for HBSD with statistics collection (dynamic traffic load).

3.5 Distribution of HBSD Utilities

We have described how to efficiently collect the necessary statistics in practice, and derive good estimates for the HBSD utility distribution during the lifetime of a message. In this last section, we turn our attention to the utility distributions themselves. First, we are interested whether the resulting distributions for HBSD delivery rate and delivery delay utilities react differently to different congestion levels, that is, if the priority given to messages of different ages shifts based on the offered load. Furthermore, we are interested whether the resulting utility shape (and respective optimal policy) could be approximated by simple(r) policies, in some congestion regimes.

We consider again the simulation scenario used in Section 3.3.2 and Section 3.4.3. First, we fix the number of sources to 50, corresponding to a *high congestion regime*. In Figure 3.18 and Figure 3.19, we plot the distribution of the HBSD delivery rate and delivery delay utilities described in Sections 3.2.1 and 3.2.2. It is evident there that the optimal utility distribution has a non-trivial shape for both optimization metrics, resulting in a complex optimal scheduling and drop policy. This also helps explain why simple drop and scheduling policies (e.g. Drop Youngest or Oldest Message, DropTail, FIFO or LIFO scheduling, etc.), considered in earlier work [54, 90] lead to incorrect decisions during congestion and perform worse than the GBSD and HBSD policies [90].

Next, we consider a scenario with low congestion. We reduce the number of sources to 15, keep the buffer size of 20 messages, but we also decrease the CBR rate of sources from 10 to 2 messages/TTL. In Figures 3.20 and 3.21, we plot the distribution of the HBSD delivery rate and delivery delay utilities, respectively, for this low congestion scenario. Surprisingly, our HBSD policy behaves very differently now, with both utility functions decaying monotonically as a function of time (albeit not at constant rate). This suggests that the optimal policy in low congestion

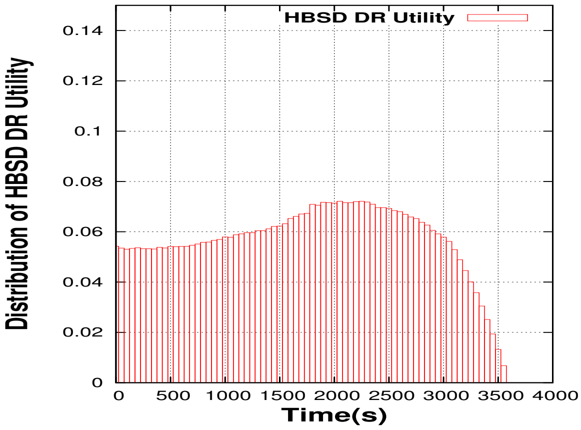

Figure 3.18: Distribution of HBSD DR utility in a congested network.

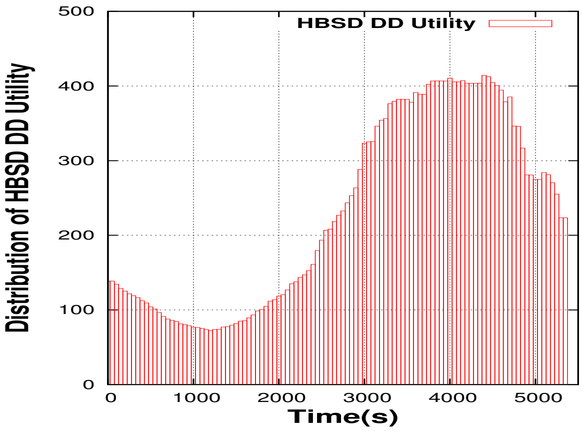

Figure 3.19: Distribution of HBSD DD utility in a congested network.

regimes could be approximated by the simpler "Drop Oldest Message" (or schedule younger messages first) policy, which does not require any signalling and statistics collection between nodes.

To test this, in Tables 3.6 and 3.7, we compare the performance of the HBSD policy against a simple combination of "Drop Oldest Message" (for Buffer Management) and "Transmit Youngest Message First" (for Scheduling during a contact). We observe, that in the low congestion regime (Tables 3.7) the two policies indeed have similar performance (4% and 5% difference in delivery rate and delivery delay, respectively). However, in the case of a congested network (Table 3.6), HBSD clearly outperforms the simple policy combination.

We can look more carefully at Figures 3.18 and 3.19, to understand what is happening in high congestion regimes. The number of copies per message created at steady state depends on the total number of messages co-existing at any time instant, and the aggregate buffer capacity. When too many messages exist in the network (for the provided buffer space per node), uniformly assigning the available messages to the existing buffers (which is what a random drop and scheduling policy would do), would imply that every message can have only a few copies created. Specifically, for congestion higher than some level, the average number of copies per message allowed is so low that most messages cannot reach their destination during their TTL (this depends only on the number of copies and mobility model). *Uniformly assigning resources between nodes is no more optimal.* Instead, to ensure that at least some messages can be delivered on time, the optimal policy gives higher priority to older messages that have managed to survive long enough (and have probably created enough copies), and "kills" some of the new ones being generated. This is evident by the values assigned at different bins (especially in the delivery delay case). In other words, when congestion is excessive *our policy performs an indirect admission control function.*

Contrary to this, when the offered load is low enough to ensure that all messages can on average create enough copies to ensure delivery, the optimal policy simply performs a fair (i.e. equal) distribution of resources (ensured by the utility functions of Figures 3.20 and 3.21).

Table 3.6: HBSD vs. Schedule Younger First\Drop-Oldest in a congested network.

Policies:	HBSD	Schedule Younger First\Drop-Oldest
D. Rate(%):	54	29
D. Delay(s):	1967	3443

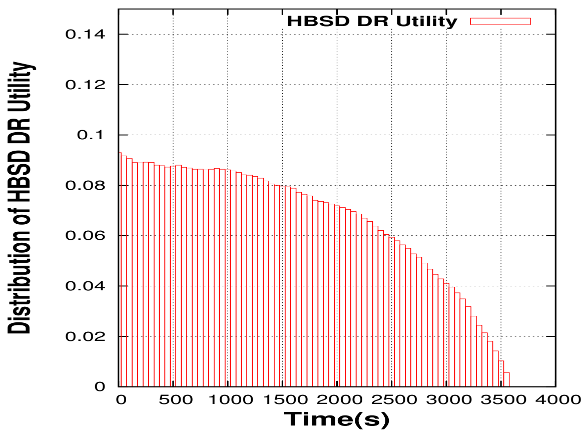

Figure 3.20: Distribution of HBSD DR utility in a low congested network.

The above findings suggest that it would be quite useful to find a generic way to signal the congestion level and identify the threshold based on which nodes can decide to either activate our HBSD scheme or just use a simple Drop/Scheduling policy. Suspending a complex Drop/Scheduling mechanism and its underlying statistics collection and maintenance methods, whenever not needed, can help nodes save an

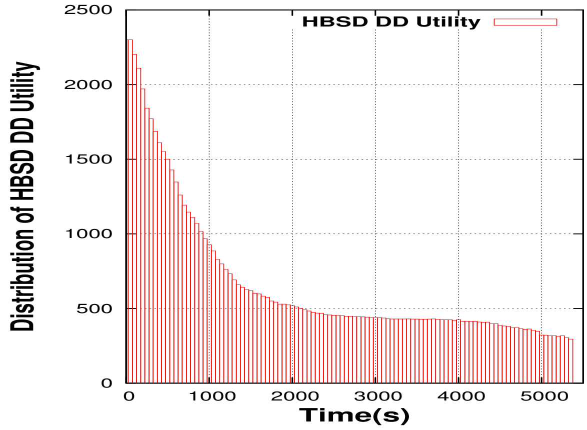

Figure 3.21: Distribution of HBSD DD utility in a low congested network.

Table 3.7: HBSD vs Schedule Younger First\Drop-Oldest in a low congested network.

Policies:	HBSD	Schedule Younger First\Drop-Oldest
D. Rate(%):	87	83
D. Delay(s):	1530	1618

important amount of resources (e.g. energy), while maintaining the same end performance. Finally, we believe that the indirect signalling provided by the behaviour of the utility function during congestion, could provide the basis for an end-to-end flow control mechanism, a problem remaining largely not addressed in the DTN context.

3.6 Summary and Open Issues

Chapter 4

HBSD: Implementation on top of the DTN2 reference architecture

Contents

We have implemented HBSD for the Delay-Tolerant Networking Research Group's DTN reference implementation (DTN2). In our DTN2 implementation of HBSD, users can choose either to maximise the delivery probability or to minimise the delivery delay for all DTN2 bundles. HBSD executes as an external DTN2 router, using DTN2's XML-based External Router Interface. HBSD is implemented in C++. This Chapter provides an overview of the implementation of HBSD for DTN2. We start by describing shortly the DTN2 architecture, we detail the DTN2 external router interface operation and we describe the implementation of the HBSD external router.

4.1 An Overview of the DTN2 Platform

Figure 4.3 is a block diagram enumerating the major components of the DTN Bundle (application specified data unit) forwarding system. As can be seen from the diagram, the bundle router module represents the most central component of the implementation; in general, it requires the most detailed information regarding the state of the system upon which to base routing decisions. Decisions made by the router are passed as a set of instructions (actions) to the forwarder which is responsible for executing the actions. This separation between policy and function allows for easy extension, modification, and replacement of the potentially complex router module.

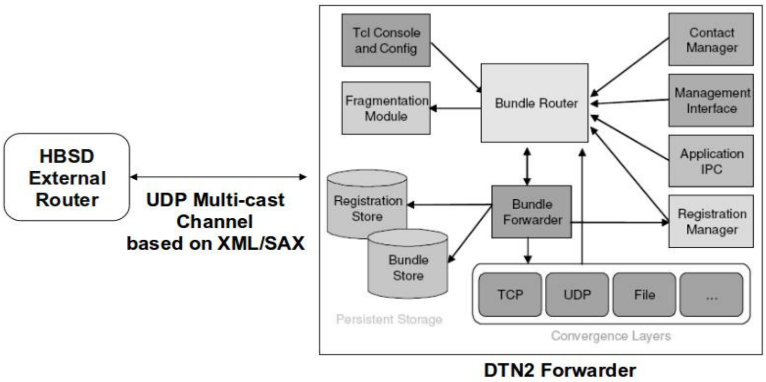

Figure 4.1: DTN2 Architecture

The DTN2 architecture provides also a set of generic external interfaces that enables third-party developers to implement plug-in modules without having to rewrite or understand the internal workings of the DTN2 reference implementation. Plug-in modules are stand-alone processes designed to work in collaboration with DTN. The External Router Interface is the first DTN external interface designed to move bundle forwarding decisions to an external process or processes. Benefits of this interface include: *(i)* classified and proprietary bundle routing algorithms are protected *(ii)* external routers may be written in any programming language (XML based interface). *(iii)* prototyping of new bundle routing algorithms is streamlined and *(iiii)* routing algorithms may be added and removed at runtime.

4.1.1 Message Processing Modules

Bundle Router and Bundle Forwarder The router component implements all the route selection and scheduling policy decision making. It takes as input a large variety of events that could potentially affect routing decisions and issues encoded instructions that are passed to the bundle forwarder, which is in turn charged with the responsibility to execute them. The forwarder executes the router's decisions by interacting with the Convergence Layers, Registrations, and the Persistent Store. The separation of router from forwarder represents an instance of separating policy from mechanism. Also, since there is different varieties of possible routing policies, separating the calculation of instructions from their execution helps to isolate the routing code from changes in the other internal APIs.

Convergence Layers Convergence Layers are the adapters between the DTN bundling protocols and various underlying transports, similar to drivers within an operating system. At the most basic level, they perform basic data plane functions: a particular layer must be able to transmit and receive bundles over a single hop (in the overlay topology). In some cases they also process signaling information required by the bundle router (e.g. such as failed connections and restarts). Convergence Layers are discussed in more detail in the following section.

Persistent Store Persistent storage is used to hold the contents of bundles during the store-and-forward process. This module provides a common abstraction for persistence storage which enables the use of a wide variety of storage methods for holding in-transit bundles. This allows a particular system instance to select (at runtime) to use either a relational database model or a simple file model.

Fragmentation Module The fragmentation module is responsible for fragmenting and reassembling bundle fragments. In DTN, fragmentation is used in routing both proactively when a large message is to be moved over a contact of smaller known finite volume as well as reactively when a data transfer fails unexpectedly. This module is able to signal the bundle router when all the fragments of a subject bundle have been received.

4.1.2 Management Modules

Contact Manager The Contact Manager is responsible for keeping track of which links are currently available, any historical information regarding their connectivity or performance, and any known future schedules of when connectivity may be available. The primary task of the contact manager is to transform the information learned about contacts from environment-specific mechanisms into abstract contact descriptions that can be used by the bundle router.

Management Interface The management interface is used to signal the bundle router about any special policy constraints or preferences that may affect its data routing decisions. It is implemented as a generic interprocess communication capability so that multiple applications or processes may be supported. For example, this hook could be used to signal the router to scan for potential contacts when a WiFi link detects a hotspot.

Console / Config The console/configuration module provides a command line interface and an event loop for testing and debugging of the implementation, as well as a structured method to set initial configuration options.

4.1.3 Application Support Module

Application IPC / Registration Module DTN applications are written to use a thin library that communicates with the router via an inter-process communication channel. Most of this interaction relates to sending and receiving application messages and manipulating message demultiplexing bindings.

4.2 DTN2 External Router Interface Operation

Before diving into the details of HBSD implemantation, this section discusses the overall design and operation of the DTN2 external router interface. When compiled, the DTN2 produces one multi-threaded executable, a DTN daemon. This daemon is a complete DTN node; it accepts bundles from a number of built-in convergence layers, provides persistent bundle storage, delivers bundles to local DTN applications, selects routes using built-in routing logic, and forwards bundles to peer DTN nodes. A forwarder is a DTN daemon as described above, but in our case, one that depends on external routing processes to make bundle routing decisions on its behalf.

The forwarder communicates with external routing processes with an XML-based messaging protocol using a well-known IPv4 multicast address and UDP port on local or remote hosts. Before forwarders and external routers can communicate, they each must join the all-routers multicast group (224.0.0.2) and bind to a well-known UDP port. Forwarders are not aware if zero, one, or more external routers have joined the multicast group. It is the responsibility of the system administrator to ensure router availability.

Nothing in the interface design precludes running forwarders and external routers on different hosts, however this approach is not recommended especially in wireless and/or bandwidth-constrained environments. The interface is fairly chatty, UDP is unreliable, and there is a high risk of packet loss leading to a breakdown in synchronized state. (The DTN2 external router interface is hard coded to use the loopback interface and therefore requires forwarders and external routers to reside on the same host.)

Inter-process messages are XML-based and must be valid against the external router XML schema. Interface messages are broadly divided into four categories. Event messages are issued by forwarders to indicate state changes that may be of interest to external routers. Request messages are sent by external routers to direct forwarders to perform an action (e.g. "forward the bundle with ID 56 out link tcp0"). Query and report messages are used by external routers to synchronize their state with forwarders after bootup or during failure recovery. The proper usage and interpretation of each interface message is covered in the next section.

Note that DTN2 forwarder does not authenticate external processes. The forwarder makes the assumption that (local or remote) external routers are within the same security domain. In addition, system administrators must ensure there exists one authoritative router or policy module per DTN node, or that multiple external routers are configured in a cooperative manner to correctly handle all events.

4.3 HBSD Implementation Overview

The DTN2 / HBSD architecture is very simple at the highest level. The DTN2 daemon sends multicast packets to be received by a local router process. These packets contain XML data and provide notification of events, such as the creation of a link or the receipt of a bundle. The router process, in this case HBSD, sends multicast XML messages to the DTN2 daemon requesting that some action be taken, such as the transmission of a bundle. Note that the exchanged messages are not

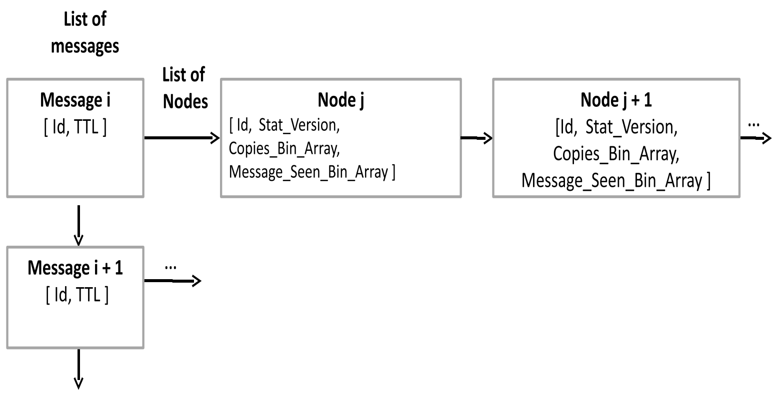

Figure 4.2: Matrix used to maintain network statistics

to be viewed only as being conversational. For example, HBSD may request that the DTN2 daemon transmit a bundle but it does not expect a response. If the DTN2 daemon does transmit the bundle, it will send a multicast message about the transmission, but that message is a separate event and not a reply to the transmit request. The implication of this is that requests to the DTN2 daemon do not have negative acknowledgments. When DTN2 establishes a connection between two nodes, HBSD is notified and the HBSD peers exchange meta data. This information is used to prioritize the transmission of bundles in subsequent meetings between nodes and to decide of the bundles to be deleted in case of storage congestion.

The exchanged meta data consist of a subset of the matrix described in Figure 4.2. The subset is defined based on the received list of (Message_ID, Node_ID) entries and their associated statistics versions.

By default, when the DTN2 daemon successfully transmits a bundle it then deletes the bundle. This is not the desired behavior; otherwise HBSD would not be able to replicate bundles on multiple nodes. That is why it's important to set early_deletion to false in the DTN2 configuration. The proper behavior is for dtnd to retain a copy of the bundle after transmission.

In the case where a bundle is destined for the local node, the DTN2 daemon will automatically delete the bundle after it has been received by the endpoint. The DTN2 daemon will then send HBSD a bundle deletion notification.

the DTN2 daemon will delete bundles when they expire and send notifications to HBSD. If HBSD learns of a bundle delivery acknowledgement via meta data and it is in possession of that bundle, it will ask the DTN2 daemon to delete the bundle. If HBSD forwards a bundle to the destination node, HBSD will request that the bundle be deleted.

Needless to say, HBSD does not actually possess, delete or transmit bundles. This is all performed by the DTN2 daemon, dtnd. Instead, HBSD makes decisions and informs the DTN2 daemon which bundles are to be transmitted or deleted.

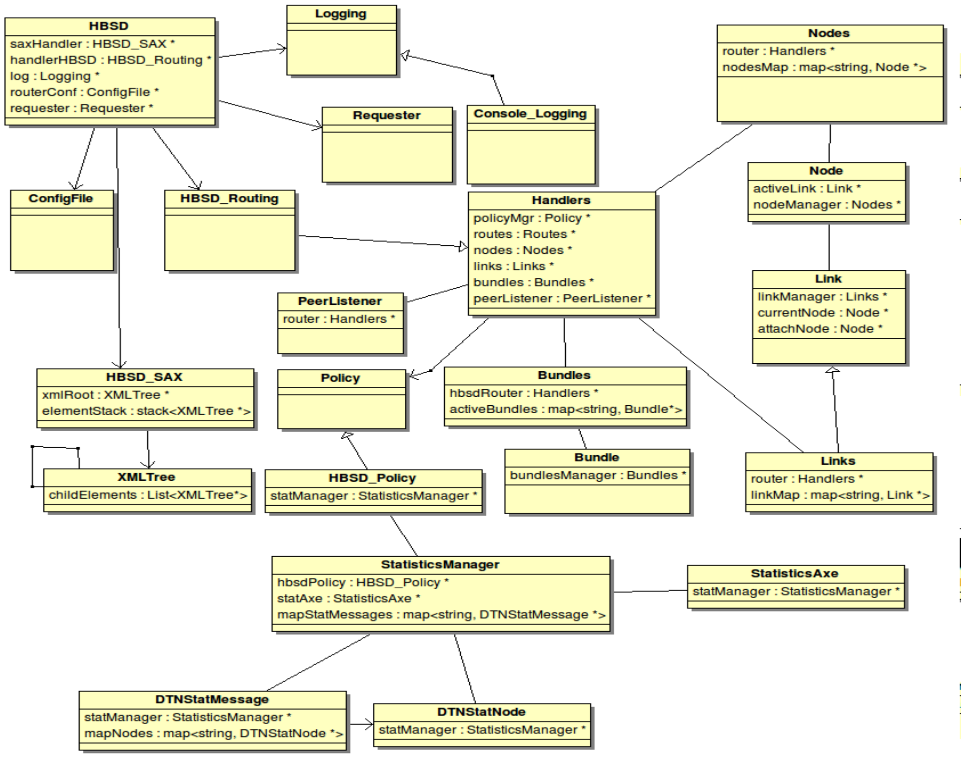

Figure 4.3: HBSD Class Diagram

4.4 Main HBSD external router building blocks

StatisticsManager class: This class implements the algorithms [94] we designed to maintain network history and infer the utilities needed by HBSD either to maximize the messages average delivery probability or to minimize their average delivery delay. StatisticsManager class maintains network statistics and update it each time a new metadata bundle is received or whenever some local storage related events occur (a bundle is dropped due to congestion, a new bundle is added,...). StatisticsManager also calculates and returns the utility of a given bundle (given its life time) with respect to the routing metric to optimize (either the average delivery rate or delivery delay).

PeerListener Class: The PeerListener object exists as a thread dedicated to processing HBSD router-specific bundles received by peer routers. DTN2 specifies that any EID with an endpoint that begins with ext.rtr (e.g., dtn://node.dtn/ext.rtr/HBSD) is to be destined for an external router, and it generates a specific XML event when it receives a bundle containing the ext.rtr endpoint. When the HBSD Routing class receives the event, it dispatches the PeerListener objects thread to process the bundle. Specifically, the PeerListener thread extracts the data from the bundle, deletes the bundle, and then makes a call into the Policy

Manager passing the bundles data. This is the recipient half of the mechanism for exchanging meta data between routers. To send meta data, the policy manager must inject a bundle into DTN2. This is done by the Policy Manager via a method in the Requester class.

Requester Class: This is a utility class responsible for composing and sending all XML messages to the DTN2 daemon. These classes are used throughout HBSD. To expand on injecting a bundle, since it is fundamental to HBSD exchanging meta data with a peer node: When the Requester injects a bundle for the router it assigns a unique request ID to the bundle. It is up to the policy manager to later associate the injected bundle with the request using the id. The Requester only plays a minor role in injecting bundles: it sends the request to DTN2 after the policy manager creates the data. But it should be noted that injected bundles are handled differently from other bundles by HBSD. HBSD creates a Bundle object for an injected bundle, but it does not retain knowledge of the bundle in the Bundles class.

GBOF class: This is another utility class, and it contains static methods for manipulating the GBOF. This includes formatting the GBOF as XML as required by DTN2. It also includes methods for creating a hash key from the values that make up the GBOF. This hash key is used extensively and consistently throughout HBSD and the Policy Manager for referencing a bundle.

HBSD Class: The HBSD class contains the main body of the router. This class reads the command line arguments, loads the configuration file (if defined), starts logging, and sets up the SAX (Simple API for XML) handler. Once initialized, it joins the DTN2 multicast group and continuously loops receiving locally broadcast messages from the DTN2 daemon, dtnd. When XML messages are received from DTN2, the SAX handler is responsible for parsing the message and dispatching the appropriate method.

HBSD SAX class: This class is invoked when HBSD receives an XML message from the DTN2 daemon. It extends the C++ SAX DefaultHandler class. HBSD SAX parses the XML message and calls the corresponding method in the Handlers class.

Handlers class: This is an abstract class that defines a method for each XML event message that may be received by HBSD SAX. The HBSD Routing class is the real implementation of the Handlers class. We use an abstract class that supplies null methods for all XML messages. If the class that extends Handlers, i.e. HBSD Routing, does not support an event then the empty method in Handlers in invoked.

XMLTree class: This is a utility class. When HBSD SAX parses an XML message each element is placed in an XMLTree object. XMLTree objects may be linked to each other to represent the hierarchy of elements in an XML message. XMLTree objects have methods for accessing attributes and child elements.

HBSD Routing class: The HBSD Routing class is the heart of the router, extending the methods defined by Handlers. It is here that the XML messages sent by the DTN2 daemon, as represented by XMLTree objects, are initially acted upon.

Logging class: This is an interface that defines the logging class used by the HBSD router. HBSD provides one implementation of the Logging class: Console Logging. By default, Console Logging is used though you can define which implementation to invoke via the HBSD configuration file.

Console Logging class: This is a simple implementation of the Logging interface that outputs logging messages to stdout. It is the default logging class

ConfFile class: This is a utility class that reads and parses the HBSD configuration file.

Bundles class: This class manages the set of individual Bundle objects.

Node class: A Node object represents a node, e.g. dtn://node.dtn. HBSD creates a Node object whenever it learns of a node, such as when a link is established to a node, or when a received bundle references a node.

Nodes class: This is the class that manages the set of individual Node objects.

Link class: A Link object represents a DTN2 link and an instance is created whenever DTN2 notifies HBSD that it has created a link. A Link object may become associated with a Node object when the link is opened; a DTN2 link that is not open is not associated with a Node. When a link is open it represents communication with another node. HBSD will associate the Link with the corresponding Node object, unless the Node object is already associated with another Link object. A node will never be associated with more than one link, even if there are multiple links open to the same node.

Links class: The Link class manages the set of individual Link objects.

Policy class: The Policy class defines the interface to be implemented by a Policy Manager. The interface source code describes the individual methods. By default, HBSD Policy implements this class, but other implementations can be defined via the HBSD configuration file.

HBSD Policy class: This class is an implementation of the Policy class. It provides the HBSD scheduling and drop algorithm, but it is also generically referred to as the Policy Manager. There are calls into the Policy class sprinkled throughout the router, often mirroring the XML events defined by the /etc/router.xsd schema file. HBSD Policy largely consists of manipulating shadow data structures dealing with bundles and nodes. The primary function of HBSD Policy is to prioritize the delivery and replication of bundles in anticipation of the local node coming into contact with another node. The assumption is that HBSD will be able to replicate only a subset of its bundles on each node that it meets, and that some of the bundles will expire before HBSD comes in contact with the actual destination node.

4.5 Summary and Open Issues

Chapter 5

Interest Driven Content Dissemination Architecture for Disruption Tolerant Networks

Contents

5.1 MobiTrade Architecture

5.1.1 MobiTrade Data Records

In a content dissemination architecture, nodes need first to somehow express their interests for different (types of) content and advertise these interests. To this end, we borrow the concept of *channels*, introduced in [20]. Specifically, the MobiTrade architecture relies on two data records: content and channel records (Fig. 5.1).

Channel Record: A user asks for a set of contents by creating locally a channel record that encapsulates the set of keywords the user thinks they better describe the contents she is looking for. Channels can be added or deleted by the user, at any time. In contrast to the CCN model [1], the channel record in MobiTrade is *one-for-many* (not one-for-one). A desirable content is identified based on a match

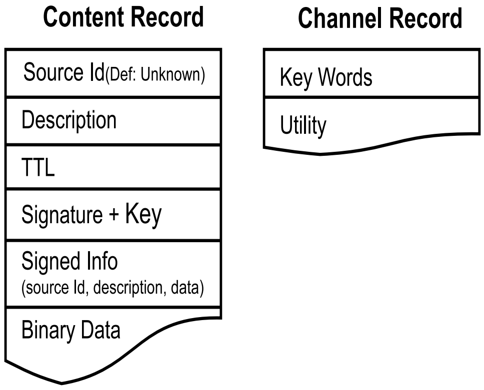

Figure 5.1: MobiTrade data records and channel storage

between the channel keywords and the content description (also characterized by a set of keywords). We think that such a match process is more appropriate for the communication model we are proposing, where the user might not know the exact content he is looking for, or might anyway be interested in all contents matching the description (e.g. Madonna songs, photos of Nice, sports tickets for sale).

A lot more can be said about this channel structure (e.g. hierarchies, merging and splitting of channels, semantic content matching, etc.), but this is beyond the scope of this paper. Instead, we choose to use a simple channel structure here and focus on the algorithmic part of the system. Finally, each channel record contains a *utility* entry. This is a key quantity for MobiTrade, allowing our system to optimize various important functions (scheduling, inventory management, collaboration profiling, etc.). For now, we will assume this as given, but Section 5.2 is devoted to how this utility is derived.

Content Record: In addition to the content description, a content record contains a number of additional fields. First, we associate to each MobiTrade content a TTL (Time to Live). By the end of this time, records can be removed. The cases requiring this TTL field are many, for example someone could be interested in selling something today. To ensure devices respect this TTL field, MobiTrade devices do not reward each other for expired content.

Furthermore, to provide users with a full control over their privacy (i.e., the contents they publish and their center of interests), we choose to keep anonymous any generated channel record (no user or device ID are stored). Nevertheless, for contents, one still has the possibility to associate to them a *canonical* source name that refers in some way to the content publisher. This field does not correspond necessarily to the user device address, but it can hold the publisher postal address or its phone number, which can be used for distinguishing between contents, feedback mechanisms on separate communication mediums or authentication purposes.

Finally, to provide for some MobiTrade security, we choose to use a CCN like *content-based security* model [1]. With this model, protection and trust travel with the content itself, rather than being a property of the connections over which it is transmitted. MobiTrade devices authenticate via the signature field (Fig. 5.1), the binding between the content source ID, its description and some parts of its data. In addition to this signature, each signed MobiTrade content carries with it the public key necessary for its verification by other devices. The signature algorithm can be selected to meet the performance requirements of that particular published data.

5.1.2 MobiTrade Protocol

Communication within the MobiTrade architecture is driven by the consumers of contents. A user asks for contents by "joining" a channel to which these might belong, storing the respective record(s) locally on the device until it becomes out of date. Then, each time a new meeting opportunity arises with another mobile device, both devices initiate the MobiTrade communication protocol that has two main functions:

- (*Fair Content Exchange*) to help nodes identify content of interest in their peer's inventory (buffer) and provide a set of rules to exchange such content in a *fair* manner.
- (*Inventory Management*) to allow nodes to profile each exchange (*learning*) and use this knowledge to improve the outcome of future interactions (*prediction*).

The MobiTrade protocol is summarized in Fig. 5.2. Each device starts by sending its list of channels to the other device. Based on it, each device decides on the set of contents to forward, i.e. content in its buffer matching one of the channels requested by its peer. These contents are scheduled for transmission and passed to the *Tit-For-Tat (TFT)* trading algorithm (Section 5.1.4) that ensures an equitable exchange.

In addition to channels the node is personally interested in, if extra space is available, it might choose to join "foreign" channels. Content for these channels is not stored for personal consumption, but can be used as exchange currency during the TFT phase, in order to acquire additional content of interest. This provides the incentive to nodes to act as *merchants*, collecting and carrying content to be used only for trading. This also allows content to *propagate efficiently across the network and between remote producer-consumer pairs, without any explicit routing mechanism.*

After content starts being exchanged (one-by-one), a node receiving a content might need to perform some inventory management. First, if it already has this piece of content (or the content is expired), it will drop it[1]. If the content is new, and there is available buffer space for the channel this belongs to, the content is stored in the buffer. The exchange finishes once the transfer of the requested contents ends or the two devices get out of the range of each other. At this point, both devices

[1]This is possible, since nodes only send to each other lists of channels and not a detailed list of contents. This is a coding trade-off that tries to avoid large amounts of meta-data being exchanged before any actual content is sent. On the other hand, it might also lead to some wasted bandwidth if lots of duplicate content is transmitted. A possible middle-ground solution to this problem could be the use of Bloom filters.

update the set of channels they are keeping track of as well as their corresponding utilities based on the (profiled) outcome of the session (as explained in Section 5.2).

5.1.3 Proportional Storage and Bandwidth Allocation

In a context with many nodes, various channels, and lots of multimedia content, node buffers will be operated most of the time at capacity and contact duration between nodes might not suffice to exchange all intended content. This highlights the need for efficient resource allocation algorithms. Unlike related work [21] [20], *MobiTrade allocates both buffer space and contact bandwidth in an equitable way among the different channels.*

First, MobiTrade implements a mechanism of *proportional soft quotas* to share available buffer space among channels. Let a node carry N channels with channel utilities $U(i), i \in \{1, N\}$, and have a total buffer capacity of B. Then, the *quota* $B(i)$ of the buffer space channel i is entitled to is

$$B(i) = \frac{U(i)}{\sum_i U(i)} B.$$

The proportion of storage allocated to a channel is proportional to its utility. Quotas are updated whenever one or more of the channel utilities change or channels are added/removed.

Based on these quotas and the amount of storage channel i is *currently* occupying, let $S(i)$, a node receiving a content (of W bytes) for channel i will perform the following actions:

- if $S(i) + W < B(i)$, then store the content.
- if $S(i) + W > B(i)$ and $\sum_i S(i) + W < B$, then store content.
- if $S(i) + W > B(i)$ and $\sum_i S(i) + W > B$, then pick the channel j maximizing $\max_j(S(j) - B(j))$ and drop the oldest content for this channel.

Points (2) and (3) above imply that the quotas $B(i)$ are soft. Channels can exceed their share and take over free space, if any is available. However, as soon as the buffer is full, the policy pushes shares back to their just proportion.

Finally, in the presence of limited contact durations, a device cannot simply forward contents by decreasing order of the utilities of their channels since a channel can match more than one content. For example, a MobiTrade device can face a situation where many contents match a popular channel, and hence it keeps forwarding only those contents at the expense of other channels which are less popular. To remedy this, MobiTrade applies the *Weighted Fair Queuing* policy which prevents starvation of channels and ensures that contents are forwarded proportionally to the utility value of the channel they match.

As a final note, when the scheduling policy decides to forward a set of contents from the same channel, MobiTrade sends the *youngest* contents first. This decision is motivated by our findings in [94] and is complementary to the drop oldest policy

MobiTrade Protocol (A, B, device A meets device B):

1.Initialization: Obtain the list of channels from B.

2.Content Transfer: Schedule the contents that match the received channels and negotiate their forwarding to B.

3.Content Reception: Readjust the channles utilities according to the available exceeding free storage space if any. Then, for each content C the device A receives:
(a) If C is already in A store then ignore it.
(b) If A store is not full then add C.
(c) If A store is full then identify the channels that exceed the most their allowed share of the store (the share is proportional to the utility), and remove as mush as needed to afford C (drop the oldest content among the identified channel).

4.End Transfer: when out of radio range or all requested contents transferred.

5.Channel Management: Add the new received channels and update the utilities of both the received channels and the already stored ones based on the outcome of the session.

Figure 5.2: MobiTrade protocol

applied among the contents of the same channel in case of congestion. More sophisticated policies could be also applied [94], but this is beyond the scope of this paper.

5.1.4 Tit-For-Tat Trading

One major difference of our system compared to other content dissemination solutions for DTNs (e.g. [20, 21, 65]) is that we assume that participating users are selfish by default. Thus they might act as free-riders: during a meeting they receive content they want but don't give something back (even if they have), in order to save transmission power or to save some bandwidth. Or, they might not collect content their peers are requesting. Experience teaches us that even in the absence of power or storage concerns (e.g. wired users of BitTorrent), additional concerns (legal, malice), might motivate users to circumvent collaborative mechanisms [95]. We believe these are *powerful incentives that could sabotage any collaborative content dissemination system, if not properly handled.*

In order to remedy this and isolate selfish users, a *strict* Tit-For-Tat (TFT) trading is enforced during meetings. Content scheduled for exchange is forwarded one-by-one (or in equally sized blocks/pieces, if contents are not of equal size), i.e. node A sends some, then node B sends some, then node A again, etc. Forwarding of content stops when the other device cannot (or chooses not to) reciprocate the amount of bytes it received[2].

In addition to this TFT mechanism ensuring that selfish nodes are not serviced when content for them *is* available, a utility maintenance mechanism (to be de-

[2]In order to solve the bootstrapping issue, when two well-intended devices meet for the first time, MobiTrade enables generous cooperation up to a certain amount as described in Section 5.2.

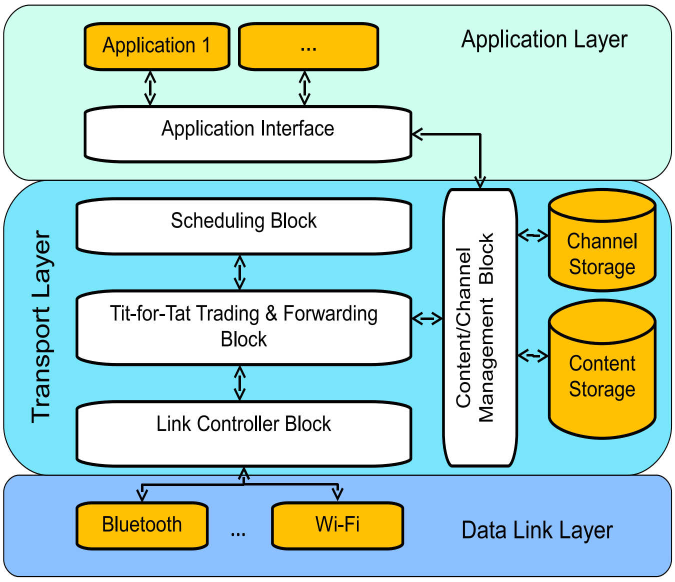

Figure 5.3: MobiTrade building blocks

scribed in Section 5.2) further ensures that well-intended nodes will not waste their resources collecting content for selfish nodes from which they will not receive any reward.

5.1.5 MobiTrade Device Model

Having described the various mechanisms of our system, we give here some more details of the MobiTrade architecture. Fig. 5.3 is a schematic of the core MobiTrade device, which has five building blocks. We are currently implementing this architecture on Android phones in order to conduct small scale experiments.

The *Application Interface* is designed so that developers can easily extend MobiTrade and create new content sharing applications that use the MobiTrade functionalities. Each MobiTrade application has a unique ID that it registers at the application interface. The application then simply communicates with the MobiTrade system using event handlers.

The *Scheduling Block* defines the order to follow while forwarding a set of requested contents within a limited contact opportunity. As explained before, this ordering is done based on the utility values.

The *Tit-For-Tat Trading and Forwarding Block* takes in charge *(i)* the forwarding of the stored channels records whenever a new contact is established, and *(ii)* the negotiation and forwarding of the requested contents.

The *Content/Channel Management Block* manages the channel and content storage space. It provides an API for storing and retrieving channel and content records that hides the storage technology specifics. This block also takes in charge utility updates for the stored channels and buffer management (as described in Section 5.1.3).

The *Link Controller Block* is the lowest layer in the MobiTrade architecture. It provides a common interface for sending and receiving data across the differ-

Table 5.1: Notation

Variable	Description
$U_{CH}(n)$	In bytes. Models the utility of the channel CH at the n_{th} meeting.
$X(CH)$	0 or 1. Expresses whether the met device is also interested in channel CH or not.
$CL(CH)$	In bytes. Expresses the collaboration level of the met device during a given meeting with respect to the channel CH.
$SC(CH)$	In bytes. Expresses the number of bytes sent to the met device during a given meeting with respect to the channel CH.
ω	Between 0 and 1. A weight that decides on the elapsed time window over which we average the utility of channels.
α	In bytes. Expresses MobiTrade device generosity level, used to de-block the situation when two devices meet for the first time and request new channels.
β	In bytes. Controls the speculation that a MobiTrade device makes regarding the expected future reward from a given channel.

ent available wireless interfaces. It is also responsible for periodically scanning the neighborhood for devices, for establishing the contacts whenever meeting opportunities arise and for triggering the Tit-For-Tat trading and forwarding block. The link controller provides an API that hides network technology specifics (Bluetooth and Wi-Fi) from the rest of the upstream blocks.

5.2 Inference of Channel Utility

MobiTrade aims at maximizing the number of collected contents while providing incentives for devices to collaborate. To have their requests satisfied, MobiTrade devices have to propose contents to other devices in counterpart of what they are looking for. As said before, each device is equipped with a storage space that is filled with contents from the different existing channels, which are used later as trading currency. This storage space is filled so as to satisfy the future demand: each channel occupies a proportion of the storage equal to the reward it is expected to bring upon future meetings. Due to Tit-For-Tat trading, the reward from carrying a channel is the amount of data from this channel a device will sell upon future meetings (to get both data for its own usage and data used for later trading). Hence, *to optimize performance, MobiTrade devices should be able to quantify the expected reward from the channels they carry.*

In MobiTrade, the expected reward from each channel is modeled through a utility function used for ranking contents upon a meeting and for dropping them upon saturation of the storage. In this section, we detail how these utilities are calculated. Table 5.1 summarizes some useful notation.

For each channel CH, MobiTrade defines its utility $U_{CH}(n)$ at the n^{th} meeting (counted over all devices) in a way to reflect the *expected* number of bytes the device will sell from this channel (and thus the bytes of interest these will buy back) with any random device it will meet[3]. This is clearly a complex interplay of mobility patterns, available channels and content, and node interests. *We choose to keep our framework as assumption free as possible about mobility and interest patterns*, and take the following approach.

Calculation and update of channel utilities: Assuming stationarity of the network over at least a time window of $2T$, the expected reward from carrying a channel CH can be calculated by averaging all experienced rewards over all meetings in the past time window T. Meetings that do not request channel CH count as zero. To facilitate implementation, an Exponential Weighted Moving Average (low pass) filter is used for averaging.

Let's consider a device A and let's suppose that an $(n+1)^{th}$ meeting opportunity arises. After the establishment of the physical connection, both devices exchange matching contents while applying the Tit-For-Tat trading algorithm (as in Fig. 5.2). Once done, they both record the volume of exchanged data and update each the utilities of the channels they carry. For device A and channel CH, this update is done as follows:

$$U_{CH}(n+1) = \omega.U_{CH}(n) + (1-\omega).X(CH).CL(CH),$$

where $\omega = (t_n - t_{n-1})/T$ is the weight associated to the low pass filter[4]. Concerning the term on the right hand side, it models the amount of data exchanged with the met device B from channel CH. $X(CH)$ is a binary variable that expresses whether B is interested or not in CH. This variable captures the popularity of a given channel over all MobiTrade devices met by A. As for $CL(CH)$, it captures the volume of contents that could be sold to device B in the future (i.e. a prediction of B's future demand for CH).

This calculation leads to several nice properties. First, this utility calculation is per channel and does not account for the physical addresses of encountered devices. The same device met several times or different devices met the same number of times count the same from the viewpoint of MobiTrade, as long as the amount of data sold was the same. This avoids tracking individual devices which improves the scalability of MobiTrade. Second, it does not try to over-optimize for the next meeting only (as could be the case with detailed mobility prediction) but rather optimizes the inventory over a larger time horizon, enough to absorb prediction inaccuracies at individual meetings. Finally, coupled with the Tit-For-Tat algorithm, our utility (through $CL(CH)$) accounts for the collaboration of devices in addition to the popularity of channels. In lack of this feature, a channel widely requested among a group of users, who do not collaborate by bringing back interesting contents for A,

[3]Note that MobiTrade does not differentiate between own and foreign channels (these could also overlap) at the level of calculating utilities. However, foreign content doesn't interfere with content for own consumption. If a peer has content of interest for own consumption, this will always be retrieved, provided it can be "bought", and passed to the application. MobiTrade only decides whether this should be further cached in the content storage for future trading.

[4]By making it function of the elapsed time between the current meeting and the previous one, one can ensure an averaging over a time window T.

could force A to keep collecting contents matching CH, only to discover later that this content buys him nothing. The storage of A could have been better used by carrying contents for less popular channels but more collaborative devices.

Collaboration and Bootstrapping: The $CL(CH)$ term above expresses the collaboration level of the device B with respect to the channel CH. If a channel however is requested for the first time at the $(n+1)^{th}$ meeting, its $U_{CH}(n)$ would be initialized to zero. A new node that asks for a channel CH, would see its request being ignored, as no content for CH was exchanged in this round. Clearly, an appropriate bootstrapping mechanism is needed, in order to avoid this chicken and egg problem for new nodes or channels. This can be implemented as some *slack* or *generosity* in the $CL(CH)$ calculation and the TFT mechanism. At the same time, this generosity should be such that it *cannot be exploited* by selfish nodes. The calculation of $CL(CH)$ below is inspired from TCP slow start, and attempts to best satisfy the above two (conflicting) goals:

$$CL(CH) = \begin{cases} Max(\alpha, 2.SC(CH)) & \text{if } SC(CH) < \beta, \\ SC(CH) + \alpha & \text{otherwise.} \end{cases}$$

The collaboration level $CL(CH)$, that is, the prediction of future demand, is thus a function of the actual (last exchange) demand $SC(CH)$. If $SC(CH)$ is less than some threshold β, we allow $SC(CH)$ to double, to accelerate the collaboration process at its beginning; otherwise, devices will have to meet more often to reach a satisfactory collaboration level. After β[5], the generosity of device A switches into a linear mode when it believes it has successfully approximated the steady-state demand, and only speculates an additional α to $SC(CH)$.

The same factor α is equally used as minimal value for $CL(CH)$ to unblock the situation when device B asks for a new channel ($SC(CH)$ equals zero in this latter case). If a channel does not bring the expected reward for any reason (lack of collaboration, oldness of the contents carried, etc.), this will be reflected by a decrease in $SC(CH)$, which automatically leads to a decrease in the utility value we associate to this channel. Similarly, α also serves to keeps selfish nodes in control. If some selfish device asks for a long list of new channels, MobiTrade will associate initially a small utility value to them. A selfish/malicious user is then obliged to collaborate in order to increase the utilities of his channels and thus the portion of content storage these are given.

From the perspective of a collaborative trader node, a community of non collaborative users is equivalent to a community of users not requesting channels. We believe this improves the robustness of the system and allows it to scale to large networks, without the need for explicit blacklisting and reputation systems (at least for selfish nodes).

[5] We take an optimistic approach and choose it equal to the maximum utility value over all channels of A.

5.3 Performance Evaluation

We move now to performance evaluation of our system. We first describe our experimental setup, and then present simulations results for two main types of scenarios: collaborative scenarios and scenarios including selfish users.

5.3.1 Experimental Setup

Protocols: We have implemented the MobiTrade content dissemination protocol in the NS3 simulator [96]. Throughout our simulations we will be considering two versions of MobiTrade, with (**MobiTrade + TFT**) and without Tit-For-Tat (**MobiTrade - TFT**). Note that this corresponds only to the forwarding module, described in Section 5.1.4. The channel utility maintenance, described in Section 5.2, is kept on in all scenarios. We have also implemented two different versions of the *PodNet* scheme as a baseline for comparison[6], as described, to our best understanding in [20] and [**?**]: (i) non-collaborative Podcasting, where users just carry and share their own channels [20] (**Podcasting**); (ii) collaborative Podcasting with the *Uniform* channel sharing strategy, where, a device records all channels it has seen in the past and solicits contents for these channels randomly [**?**] (**Podcasting + Uniform**). This latter strategy was shown to perform best in [**?**], compared to other heuristics taking into account channel popularity. As a final note, we stress that MobiTrade's first goal is not to compete with optimal collaborative schemes, but rather to efficiently deal with selfish nodes, without compromising the socially optimal (collaborative) performance.

Mobility Models: To evaluate the different protocols, we use two type of mobility scenarios, a state-of-the-art synthetic mobility model (HCMM) [67] and a real mobility trace (KAIST) [89]. HCMM is a mobility model inspired by Watts' Caveman model that was shown to reproduce statistics of human mobility, such as inter-contact times and contact duration. In HCMM, the Caveman model is used to define a graph (overlay) with nodes divided into (well connected) groups and each group is assigned to a physical home location. Also, some users belonging to different groups can have links to each other (bridges). These (intra- and inter-group) links are used in HCMM to drive movements: each user moves towards a given group's home location with a probability proportional to the weight of its links towards the group. In our scenario, we consider 50 users distributed into 5 groups. The plane is divided into a 10*10 grid of cells (5000 m wide), and each cell can serve as a home location for a group.

The KAIST scenario consists of real human mobility traces collected from a university campus (KAIST) in South Korea [89]. We consider a sample of the KAIST campus traces taken from 50 students, where the GPS receivers log their

[6] We choose the PodNet framework, as it is the most directly comparable to our scheme, and consists of simple enough mechanisms that we consider practical for implementation. For example, [72] is considerably more complex and based on an MCMC framework that requires careful simulated annealing and might take a long time to converge. Furthermore, [21] requires explicit knowledge of social network links, not available in our framework.

position at every 30 seconds. We integrated both mobility models in NS3. Both case studies consist of simulations that last 24 hours where devices use the 802.11*b* protocol to communicate with a transmission range around 60 meters.

Traffic Model: Unless otherwise stated, each user joins randomly 2 channels at the beginning of the simulation. For simplicity, we assume that all generated contents have the same size[7]. However, different channels do not need to have the same size (the size of a channel is equal to the sum of its contents' sizes). Some channels might have lots of content, and others less. Finally, we consider that each user generates contents periodically that match one of the channels that were requested by users from other groups[8].

5.3.2 Collaborative scenarios

We first evaluate MobiTrade, assuming all nodes are collaborative, using the following four scenarios (described in Table 5.2) (there are 50 channels in total): $\mathbf{SC_1}$ implements a *homogeneous traffic* pattern, i.e. each channel has the same size and each user joins the same number of channels. In $\mathbf{SC_2}$, users choose a *random number of channels* to join, but channels still have the same size. In $\mathbf{SC_3}$, users ask for the same number of channels but these have *random sizes*. Finally, $\mathbf{SC_4}$ introduces some *churn*, where 10 of the users join the simulation after 8 hours, while existing sessions are ongoing, and leave again 8 hours later. Due to space limitations, for all scenarios, we show plots for the HCMM mobility model, accompanied with respective results for the KAIST trace summarized in Tables.

Table 5.2: Collaborative simulation scenarios

Sim. Scenario:	$\mathbf{SC_1}$	$\mathbf{SC_2}$	$\mathbf{SC_3}$	$\mathbf{SC_4}$
Nbr. of Users:	50	50	50	40 + 10 transient
Requested CH(s) per User	2	random [1, 20]	2	2
Size of CH(s) (# of contents)	20	20	Random [1, 20]	20

Effect of TTL: Before we proceed, we take a quick look first into the impact of content TTL. As explained in Section 5.1, our buffer drop and scheduling policies give priority to younger messages[9]. This is not only in accordance with our daily experience (for many types of content, e.g. news feeds, we prefer to have the most recent version), but has also been shown to be an efficient resource allocation policy in the context of a single channel [94] (intuitively, older content has a higher chance to have been delivered already). Fig. 5.4 depicts the MobiTrade average delivery

[7] Variable sized content could still be split to equal sized pieces or blocks. We defer the study of more complex content structures to future work.

[8] The content generation interval depends on the number of contents for a channel and the duration of the simulation.

[9] Note that this age *only* corresponds to the time the content was inserted in the network by its publisher, and does not (directly) relate to the actual age (e.g. an old vs. a new rock song.)

rate as a function of the content TTL (for scenario $\mathbf{SC_1}$), with and without prioritizing younger messages (per channel). It is evident that the higher the (application chosen) TTL the more old content "hogs" node buffer and contact bandwidth, not allowing new content to reach its audience. When prioritizing younger messages, not only does performance stabilize, but with an infinite TTL the gain from just this mechanism is up to 76%.

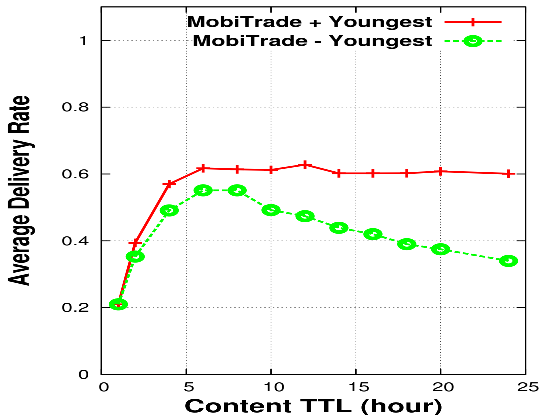

Figure 5.4: Drop and Scheduling policy inside the same channel ($\mathbf{SC_1}$).

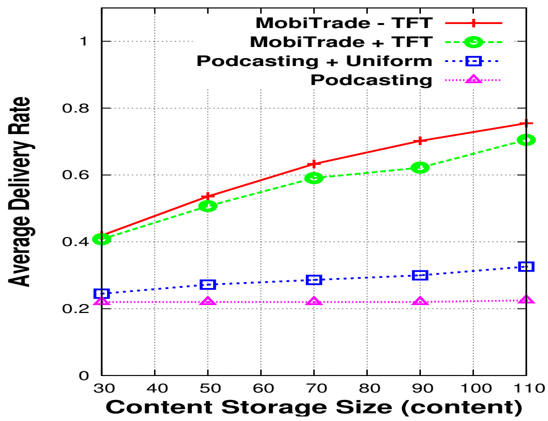

Figure 5.5: Fixed number of channels per user and fixed channel size ($\mathbf{SC_1}$).

Scenario $\mathbf{SC_1}$: Fig. 5.5 compares the performance of MobiTrade with and without TFT and the two versions of Podcasting described in Section 5.3.1. The figure of merit is, again, the average delivery rate, defined as the *amount of content received for channels a node requested* divided by the *total amount of content generated for these channels* (throughout the simulation). This is averaged over all nodes. Delivery rate is plotted as a function of node storage.

There are three main observations to be made in Fig. 5.5. *First*, collecting and sharing foreign channels (**MobiTrade** and **Podcasting + Uniform**) improves performance compared to only storing own channel. This confirms the findings of [?]. *Second*, the uniform sharing policy [?] is clearly not optimal (as suggested also in [72]), and is significantly outperformed by MobiTrade's inventory management (by up to 2×). This is more pronounced as storage is increased. *Third*, using Tit-For-Tat (TFT) in a context where *all* nodes are well-intended results in a small drop of the average delivery rate by about 6%, compared to the case without TFT. Using game theoretic terms, in Section 5.3.4, we show that *rational users will choose to pay this price to secure themselves from selfish users.* Finally, Table 5.3, summarizes the respective results for the KAIST trace (we only show values for a buffer of 110 contents; the plot trend is similar to Fig. 5.5). The results for the KAIST trace (row 1) corroborate the above findings.

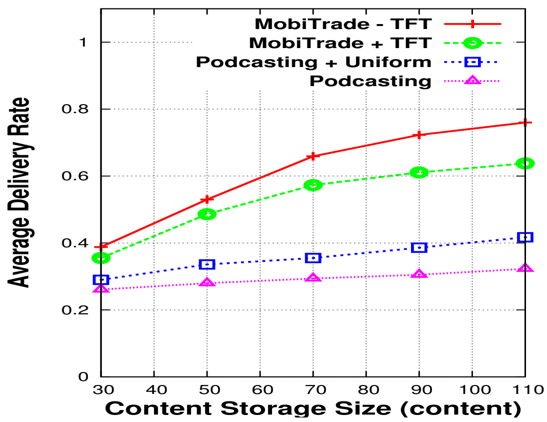

Figure 5.6: Scenario $\mathbf{SC_2}$.

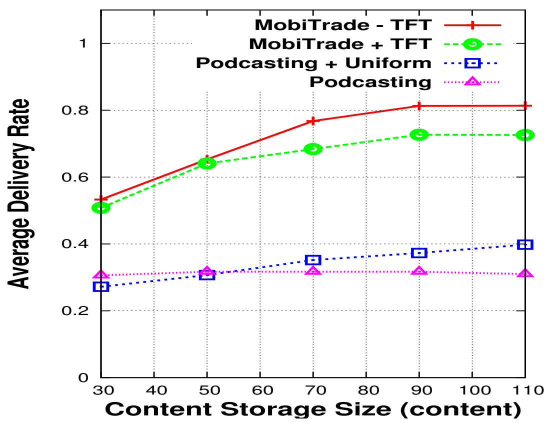

Figure 5.7: Scenario $\mathbf{SC_3}$.

Scenarios $\mathbf{SC_2}$ and $\mathbf{SC_3}$: These two scenarios consider the effect of heterogeneity

Table 5.3: Avg. delivery rate based on the real KAIST trace (collaborative scenario, content storage size = 110 contents).

Policy:	**MobiTrade + TFT**	**MobiTrade - TFT**	**Podcasting**	**Podcasting Uniform**
Scenario $\mathbf{SC_1}$	0.83	0.89	0.6	0.72
Scenario $\mathbf{SC_2}$	0.78	0.86	0.75	0.69
Scenario $\mathbf{SC_3}$	0.79	0.88	0.68	0.74

with respect to channel demand ($\mathbf{SC_2}$) and channel size ($\mathbf{SC_3}$). The goal is to examine whether *asymmetry* of demand or supply of content (common in practice) could give rise to *deadlocks* due to the *inherent symmetry* of the Tit-For-Tat mechanism.

Figures 5.6 and 5.7 show the respective delivery rate for these two scenarios, as a function of storage space. As we can see, traffic asymmetry does not affect the main observations made in scenario $\mathbf{SC_1}$. Interestingly, for ($\mathbf{SC_2}$), Podcasting only own channels seems to outperform uniform sharing of foreign channels. Results for the KAIST trace are again in agreement (rows 2 and 3 of Table 5.3)). We conclude that, even in the presence of asymmetric traffic, MobiTrade performs up to almost 2× better even without selfish nodes. Finally, while it is clear that these two scenarios do not suffice to *exclude* every probability of a deadlock, they constitute positive evidence to the robustness of MobiTrade. We defer an analytical treatment of deadlocks to future work.

Scenarios $\mathbf{SC_4}$: The objective of this last scenario is to study the impact of *node churn* and the ability of MobiTrade to efficiently bootstrap new nodes. Here, 10 new users join the simulation after 8 hours, each one of them asks for 2 already existing channels, then, it leaves the simulation 8 hours later. Fig. 5.8 plots the average delivery rate among the 10 new users and the 40 existing ones as a function of time. It is evident there, that when the new set of users join already existing channels, they are not blocked. Instead, they are able to collaborate and quickly scale up their performance[10].

Delay: Finally, in Fig. 5.9, we look at the average delivery delay of different schemes (scenario $\mathbf{SC_1}$), measured as *time a matching content is received* − *time it was inserted in the channel.* We can clearly see that the ranking of schemes in terms of delay is similar to the one for delivery ratio (Fig. 5.5).

5.3.3 Scenarios with selfish users (SU)

Having established the good performance of MobiTrade in the absence of selfish nodes, we now turn our attention to scenarios where few or more nodes might not reciprocate for content they receive. We deem such scenarios as the norm rather than the exception in the real world. As mentioned earlier, most related proposals do not deal (explicitly) with such users. We consider two such scenarios, as described

[10]The important thing in this plot is the slope of the curves for new and old nodes, which matches, implying that both types get similar service. They differences in absolute value are only because nodes joining late have already missed part of the content already generated for this channel (and possibly dropped due to congestion).

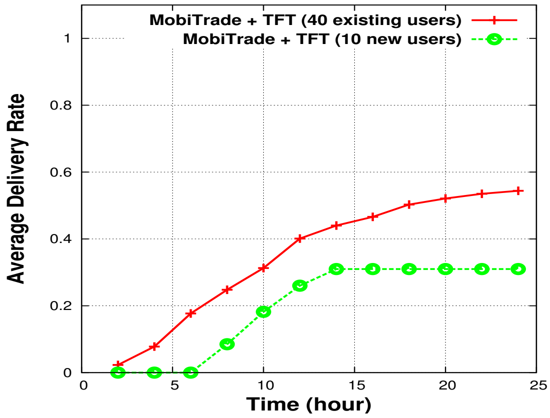

Figure 5.8: Scenario $\mathbf{SC_4}$, 10 new users ask for already existing channels.

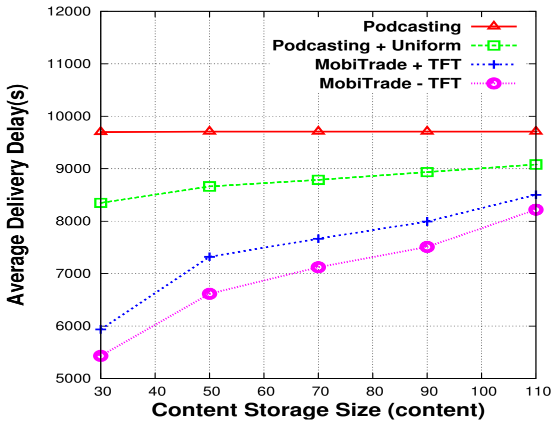

Figure 5.9: Scenario $\mathbf{SC_1}$, studying the average delivery delay.

in Table 5.4: In $\mathbf{SS_1}$, we consider 10 selfish users among 50 that ask for different channels than those requested by the remaining collaborative users. In $\mathbf{SS_2}$, we consider the same number of selfish users which ask randomly for channels already requested by collaborative users. Selfish users and Collaborative users are denoted with "SU" and "CU", respectively.

Table 5.4: Simulation scenarios including selfish users.

Sim. Scenario:	$\mathbf{SS_1}$	$\mathbf{SS_2}$
Nbr. of Users:	40 CU + 10 SU	40 CU + 10 SU
Nbr. of CH(s)	CU: 2/20 - SU: 2/10 (SU and CU channels differ)	CU, SU: 2/20 (among same channels)
Size of CH(s)	CU: 20 - SU: 40	CU, SU: 20

Scenario $\mathbf{SS_1}$: Fig. 5.10 depicts the average delivery rate (for different user strategies, CU and SU) with and without the TFT mechanism enabled. At high congestion (storage of 50 contents), enabling the TFT mechanism increases the average delivery rate among collaborative users by 15% (16% using the KAIST trace, Table 5.5) and decreases it among selfish users by 63%. Indeed, enabling the TFT mechanism blocks selfish users and makes MobiTrade re-dispatch/reuse the saved resources among the channels shared by collaborative users. For a storage of 110, collaborative users are able to reach 73% higher throughput than selfish ones, by using TFT. The latter see a $3 - 4\times$ drop in performance. In the same context, as shown in Table 5.6, the Podcasting scheme cannot control selfish nodes, as expected, and as their numbers increase, the latter end up outperforming collaborative ones.

Scenario $\mathbf{SS_2}$: Here, the 10 selfish nodes ask for channels already requested and carried by collaborative ones. This means that the utility management mechanism cannot affect them, allowing more opportunities to "scrape" content. Fig. 5.11 plots the average delivery rate of (MobiTrade + TFT) among collaborative users in two cases: first *(i)*, when selfish users are active and second *(ii)* when they are inactive. Clearly, when TFT is used, the performance of collaborative users is not harmed (verified also for the KAIST trace, Table 5.5), while the one of selfish users drops severely, by up to $2.1\times$ for a storage of 110 contents[11]. This result consolidates our findings in Section 5.2 regarding the impact of selfish users on the performance of collaborative ones once they both join the same channels. Indeed, selfish users are simply considered by MobiTrade as users which don't ask for the channels. The system resources are kept safe and only dispatched among collaborative users.

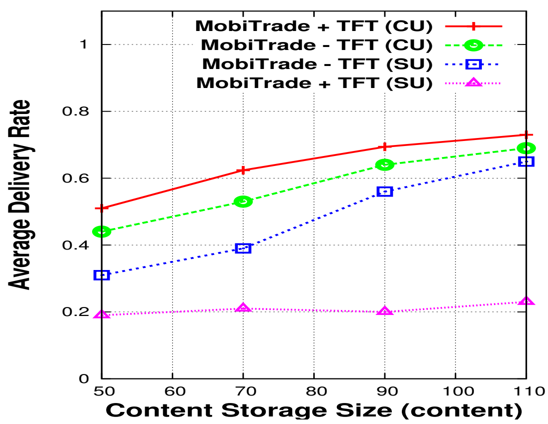

Figure 5.10: Scenario $\mathbf{SS_1}$, impact on selfish and collaborative users.

[11] We observe that in this, as well as the previous scenario, selfish users are not 100% isolated. This is *only due* to the generosity mechanism described in Section 5.2 and the fact that we chose the minimum unit of transmission α to be one content, for simplicity. Increasing the amount of content in the network or reducing the value of α (note that this does not affect collaborative users much, due to the multiplicative increase), further isolates selfish nodes.

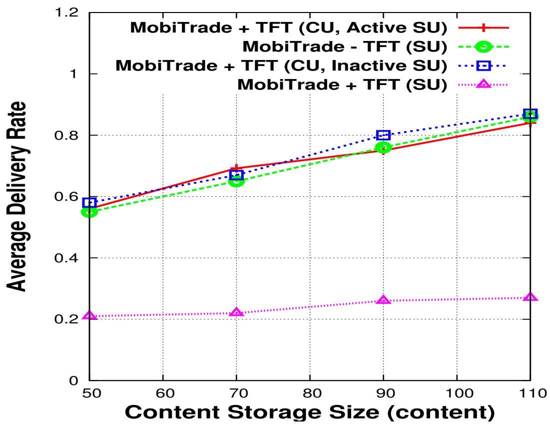

Figure 5.11: Scenario $\mathbf{SS_2}$, impact on collaborative users.

Table 5.5: Avg. delivery rate based on the real KAIST trace (scenario including selfish users, content storage size = 50 contents).

Policy:	**MobiTrade (CU)**	**MobiTrade (CU)**	**MobiTrade + TFT(SU)**	**MobiTrade - TFT(SU)**
$\mathbf{SS_1}$:	0.79 (+TFT)	0.68 (-TFT)	0.21	0.57
$\mathbf{SS_2}$:	0.81 (+TFT, Inactive SU)	0.78 (+TFT, Active SU)	0.24	0.77

Table 5.6: Avg. delivery rate (HCCM mobility model, content storage size = 110 contents, CU and SU ask for different channels).

Nbr. SU(s):	**5**	**10**	**15**	**20**
MobiTrade + TFT(CU):	0.8	0.76	0.71	0.62
MobiTrade + TFT(SU):	0.25	0.22	0.2	0.17
Podcasting + Uniform(CU):	0.46	0.4	0.37	0.34
Podcasting + Uniform(SU):	0.29	0.33	0.36	0.39

5.3.4 Choosing Strategies in MobiTrade

We have so far considered scenarios with x collaborative nodes or y selfish ones. But why would any node choose to be selfish or collaborative? Our results suggest that selfish behavior pays off if other nodes have TFT off, but hurts when TFT is on. And why would a collaborative node choose to employ our TFT mechanism? Our results suggest that *if all nodes are collaborative* they might get more content by turning TFT off. When multiple nodes have with these choices, where would the system converge? We *sketch* below a game theoretic framework that tries to answer these questions (due to space limitations, we don't include here a more rigorous treatment).

Let's consider a simple case of two nodes with contents to exchange. The set of strategies $\mathcal{A}$ for each node is to choose from is: (i) being selfish (SU), (ii) being collaborative while activating TFT (CU + TFT) or (iii) being collaborative but not activating TFT (CU - TFT). Let M be the amount of content these nodes get if they normally use MobiTrade (CU+TFT). Let finally γ be a *discount factor*, capturing the cost to a node (e.g. energy) of reciprocating a piece of content ($0 \leq \gamma \leq 1$). The *total payoff* to each node from getting M contents is

$$payoff = \gamma M.$$

The following matrix describes the "MobiTrade game" and the respective payoffs in normal form. It is not a zero-sum game.

Table 5.7: MobiTrade Game and Payoffs.

	CU + TFT	**CU - TFT**	**SU**
CU + TFT	$[\gamma M, \gamma M]$	$[\gamma M^+, \gamma M]$	$[0, 0]$
CU - TFT	$[\gamma M, \gamma M^+]$	$[\gamma M^+, \gamma M^+]$	$[(\gamma - 1)M^+, M^+]$
SU	$[0, 0]$	$[M^+, (\gamma - 1)M^+]$	$[0, 0]$

We note that M^+ is the somewhat higher payoff a node gets if both nodes are not selfish and disable TFT (see e.g. Fig. 5.5) and $(\gamma - 1)M(\leq 0)$ is the cost to a node of sending M contents without getting something back. We can see that if $\gamma = 0$ (cost per content is equal to gain), then no user has an incentive to participate in the system (i.e. users will be selfish). This however, as well as $\gamma = 1$, are unrealistic cases.

The more interesting cases are for $0 < \gamma < 1$. In this case, it is easy to see that the game has a single *Nash equilibrium* at (CU + TFT, CU + TFT) with payoffs $(\gamma M, \gamma M)$. That is, none of the two nodes can *strictly* improve its payoff by a *unilateral* change of strategy [97]. In other words, for nodes participating in our network choosing to use MobiTrade with TFT is the optimal (selfish or rational) strategy, which is a very desirable outcome for out system. This analysis can be easily extended to N nodes. The main difference there is that (CU+TFT) has a non-zero reward even if all but one other users are not SU (making the equilibrium stronger).

Finally, deactivating TFT, i.e. point (CU-TFT, CU-TFT), is the socially optimal operating point and is also *Pareto optimal.* However, it is not an equilibrium except for the limiting case of $\gamma = 1$ (no cost). For all other cases, the *price of anarchy* per node is equal to $\gamma(M^+ - M)$, which as we saw in the results of this section, is only a small price to pay.

5.4 Summary and Open Issues

Chapter 6

MobiTrade: Implementation on Android SmartPhones

Contents

In this chapter we present the design of MobiTrade, an architecture for opportunistic collaborative content dissemination. We start by giving a high level overview of the system and its design principles in []. Then we present the actual design and implementation in Section []. We describe a set of initial functionalities that we implement as an Android mobile application on top of MobiTrade in Section []. We conclude the chapter with a discussion of the open issues in Section [] and a summary in Section [].

6.1 An Overview of Bluetooth

Bluetooth is a low-power short-range wireless communication technology intended to replace the cables connecting electronic devices. Below we give a short overview on the Bluetooth protocol stack, the device discovery procedure (called inquiry scan in Bluetooth terminology), supported higher layer protocols and application programming interfaces.

Bluetooth operates on a license-free ISM band at 2.4GHz (the same as used by 802.11). The physical layer is based on frequency hopping spread spectrum (FHSS) and transmits data on up to 79 frequency bands (1 MHz each). Each frequency band is divided into time slots and full duplex transmission is provided through the use of a time-division duplex (TDD) scheme.

Bluetooth network has a master-slave structure. A master device can communicate with up to seven devices forming a so called piconet. In a piconet devices communicate on the same physical channel that is defined by a common clock (set by the master) and a frequency hopping pattern. By definition, the device that initiates a connection becomes the master. Once a piconet has been established, master-slave roles may be exchanged. At any given time, data can be transferred between the master and one other device, but never directly between two slaves. A device can only be synchronized to a single channel at a time. Multiple simultaneous operations (e.g. partici- pating in various piconets, being discoverable and connectable) are supported using time-division multiplexing between various channels. However, device can only be the master of a single piconet.

Above the physical layer in the architecture there is a number of logical links for control and data traffic. These are managed by a L2CAP layer that provides a channel based abstraction for applications. One logical (and physical) link can thus carry data for multiple applications. L2CAP provides reliable transmission performing flow control, CRC checks and retransmissions upon request. The main traffic services provided are asynchronous connection-oriented unicast and isochronous constant rate channel (e.g. for audio streaming). However, these channels are rarely used directly by applications, instead several higher layer protocols have been standardized and implemented in various client libraries (see below).

The most commonly adopted Bluetooth specifications include v1.2, v2.0 and v2.1 all being backwards compatible. The specifications differ mainly in supported bit rates and support for some advanced features. The nominal rate for Bluetooth v1.2 is 1Mbit/s. Bluetooth v2.0 increases the bit rate up to 2Mbit/s (Basic Rate) and 3Mbit/s (Enhanced Data Rate or EDR). v2.1 extends the inquiry responses (more on this below) and adds secure pairing among other minor tweaks. The operational ranges of Bluetooth devices vary from approximately 1, 10 to 100 meters (class 3, class 2 and class 1 respectively). Smartphones are generally class 2 devices.

6.1.1 Inquery Scan Procedure

The Bluetooth specification defines two separate physical channels for device discovery (inquiry scan channel) and connection setup (page scan channel). Each Bluetooth devices can be in one of the four states: (1) connectable and discoverable, (2) connectable, (3) discoverable, or (4) neither discoverable nor connectable. A device cannot be discovered nor connected unless it is configured in the correct state.

A discoverable device listens for inquiry requests periodically (called inquiry scan state) on its inquiry scan channel that has a reduced number of hop frequencies and a slower rate of hopping. In order to discover neighboring devices, an inquiring device hops through all possible inquiry scan channel frequencies in a pseudo-random fashion, sending an inquiry request on each frequency and listening for responses. This is done at a faster rate, allowing the inquiring device to cover all inquiry scan frequencies in a reasonably short time period. The Bluetooth specification recommends an inquiry duration of 10.24s. Then, with high probability, all neighboring devices will have entered their inquiry scan state and will hear the inquiry.

An inquiry response consists of an unique 48-bit device address of the discovered device and a 24-bit Class-of-Device code (CoD). The CoD consists of a major and minor device codes. The device codes are standardized and provide information about the device type: major code can tell if the device is a computer or a phone for example while the minor code can specify if the device is a cellular or cordless phone. In addition, each device may have a human readable name that can be queried using a separate control request. The extended inquiry response available in v2.1 can provided the human readable name and additional information about supported services directly in the inquiry response. Older Bluetooth devices must use the separate control request and a service discovery protocol (see below) instead.

Once a device is discovered, a connection setup can take place. A connectable device is listening on its page scan channel for connection requests that are send in a similar fashion as inquiry scans. The connection setup must be completed before any data can be transmitted between the devices.

6.1.2 Higher Layer Protocols

Each Bluetooth device must support the Service Discovery Protocol (SDP). The service discovery mechanism provides the means for client applications to discover the existence of services provided by server applications as well as the attributes of those services. The attributes of a service include the type or class of the service and the protocol information needed to access the service. The SDP protocol itself is run by a SDP server on the device that is responsible of maintaining the local service records and answering service discovery queries for SDP clients on other Bluetooth devices.

The Bluetooth specifications define various specialized protocols on top of the L2CAP layer for different purposes such as audio streaming, telephony and data transmissions. The most commonly used serial data stream protocol is RFCOMM. The RFCOMM protocol provides emulation of serial ports (up to 60 ports can be used simultaneously depending on the implementation). It provides a simple reliable data stream service, similar to TCP. In order to connect to another Bluetooth device over RFCOMM, the client must know the server channel which can be resolved using SDP. It is also possible to use hard-coded channels, but dynamic channel numbers are recommended since the number of available channels is very limited (30).

6.1.3 Application Programming Interfaces

The main interface between user level applications and the Bluetooth device is called Host Controller Interface (HCI) that is standardized in the Bluetooth specification. However, existing Bluetooth protocol stack implementations typically do not allow direct access to the HCI interface but provide their own abstractions of the main Bluetooth operations. The main stacks in use include BlueZ for Linux based devices

2 , Windows Bluetooth stack and WinSock for Windows and Windows CE based devices 3 and Broadcom's Bluetooth stack for Windows based devices 4 .

The client APIs let the applications control the device state (discoverable and/or connectable), the human readable Bluetooth device name and very often the CoD value. The device inquiry can be initiated at anytime through the Bluetooth API and the applications can typically control the duration of the inquiry and/or the number of responses to wait for. The applications can also query for the human readable names of the discovered devices, create local SDP records for the services they provide and query the records of nearby devices. The data services such as RFCOMM are typically accessed using a special type of socket and the familiar socket API.

6.2 Mobile Platforms

The system prototype and real-life experiments presented in this dissertation are designed for off-the-shelf consumer devices running Windows Mobile operating system. However, the choice of an experimental platform is not an obvious one due to availability of various mobile hardware and software platforms. In this section we overview some of the commonly used mobile research platforms and related work on platform considerations and their performance evaluation.

6.2.1 The iMote Platform

The Intel Mote (iMote) has been a popular platform in opportunistic networks research [49, 75, 18, 131]. The iMote is a small ARM based system running TinyOS with 64kB of SRAM, 512kB of flash storage, a multi-colored LED, and a Bluetooth 1.1 radio [93]. As a small single-purpose sensor platform, iMote programming demonstrates well the basic challenges of mobile computing, most notably need to take into account the limited storage and memory, and battery life (and resulting spurious resets) as reported in [99]. In addition, due to their simple configuration iMotes cannot be used for much more than Bluetooth contact logging. This is a limitation for us as one of the major goals of our experiments is to gather real application traffic in an opportunistic setting.

6.2.2 Smartphones in Mobile Systems Research

Mobile systems research is often done using ordinary smartphones. All major smartphone platforms, Android 13 , BlackBerry 14 , iPhone 15 , Symbian 16 and Windows Mobile 17 , support development of third party applications. Each platform provides its own approaches to application development and application level resource management. The solution space varies from application specific strategies to mobile application development frameworks that hide all the details of resource management from applications [118]. Borrowing Windows Mobile terminology, the two extreme approaches are also known as managed code and native code.

Managed applications are typically executed in a separate runtime environment, for example Java Runtime Environment (JRE) in Android and Blackberry or Common Runtime Environment (CRL) on Windows Mobile. The runtimes and related SDKs typically hide much of the complexity of mobile computing from the developers by taking care of task such as memory management, application live cycle, resource sharing and security. But as a result the applications are much more restricted in terms of what can be done. For example, as a result in some environments simple background execution may be impossible. In contrast native applications typically have access to the full operating system functionality exposing also most of the challenges of mobile computing directly to the developer.

A comprehensive survey of the capabilities of currently (as of 2008) available major mobile platforms for mobile systems research is provided in [101]. The authors evaluate five popular smartphone platforms including Android (Linux), BlackBerry, iPhone, Symbian and Windows Mobile. As the evaluation criteria they use the availability (at some API level) of a set of common features required by a variety of mobile systems research prototypes. These features include network scanning (discovery of a contact opportunity using any interface), interface selection (the ability to use any available interface in ad hoc and/or infrastructure mode), location services (e.g. GPS), energy level monitoring, network interfaces power management (turning on/off the interfaces or the screen, controlling the power saving mode), background processing and low-level memory management. Our work shares similar requirements for the mobile platform.

The majority of the studied platforms provide an application programming interface to access most the required features. The main differences are in usability and development tools support: some platforms provide native but hard to use or badly documented interfaces while others incorporate most of the required functionality in the managed runtimes. We do our own survey on Java ME 18 frameworks for Windows Mobile [109] for its portability and ease of use but find that at the time of writing the supported functionality is inadequate and that the memory footprint of the available runtimes is high. We conclude that programmers should rely on native interfaces when implementing advanced mobile systems and applications including opportunistic networking on Windows Mobile (versions 5.x and 6.x).

Many authors have analyzed the performance and implementation trade-offs of various mobile applications on specific mobile platforms [102, 39, 79, 85] 19 . In [102] the authors evaluate the performance of the DTN reference implementation (DRI) on a resource limited computing platform. The authors show that the DRI performance is limited by the slow disk I/O (especially when using Compact Flash storage typical on mobile platforms) and the CPU power. An adaptive location-based micro-blogging service and prototype performance on Nokia N95 is analyzed in [39]. More specifically, the adaption mechanism trades-off the location accuracy for energy-efficiency.

SoundSense [79] is a stand-alone sound profiling application implemented on iPhones that offers a good performance by trading-off memory usage for lower CPU process- ing requirements. Darwin [85] is a collaborative sensing system implemented and tested on Nokia N97 devices and iPhones. With the help of collaboration between co-located mobile devices and some backend processing support, Darwin achieves high energy efficiency in performing complex classification computations on mobile devices.

The results of these studies are encouraging, and also prone to change rapidly as mobile platforms and application development frameworks evolve. Hence, the main take-away is that the available open smartphone platforms and developed implementation techniques offer already the required functionality and system performance for implementing advanced opportunistic mobile social networking systems and applications.

In this dissertation we develop a middleware for opportunistic communications that provides a clean abstraction of the opportunistic networking layer to the applications. The middleware includes a persistent storage component and provides simple resource management functions related to the networking and energy resources usage. We prototype and test our system on Windows Mobile (5.X and 6.X) devices.

6.3 MobiTrade Architecture Overview

6.4 Implementation

We have implemented our system in Java for the Google Android OS platform. Our implementation is based on Bluetooth but we also intend to support the 802.11 in ad-hoc mode in the future. The Android Java libraries (version 2.2) do not currently support the ad-hoc mode of 802.11 although this is supported by both the driver and the hardware interface on the HTC Desire device. Therefore, making our implementation supporting the ad-hoc mode requires the device to be run in privileged user mode (i.e. rooted mode) so that the interface can be reconfigured to run in ad-hoc mode.

6.5 Summary and Open Issues

We have described the design and implementation of our system for the Google Android platform. Our experience from the implementation is that Android is a very powerful platform and quite mature despite its young age. The Java based environment provides a familiar environment with good support for most common OS primitives such as threads and concurrency, database and content storage and inter process communication through the Android service binding mechanism. Some features are however still missing, in particular support for the 802.11 ad-hoc mode (which needs to be implemented in native code). We believe that our design is general and facilitates the implementation of advanced content-centric applications. There are however some issues that are not, or only partially addressed by our design. We do currently not address particularly the issues of privacy, security and power management.

Chapter 7

Conclusions and perspectives

Appendix A

Appendix Example

A.1 Appendix Example section

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This for other bibtex stye file : only one author [?] and many authors [?].

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Towards a better content dissemination applications for Disruption Tolerant Networks

Abstract: **Keywords:** Buffer Management, Scheduling, Content Dissemination Architecture, Disruption Tolerant Networks, DTN2, Android