Framing and Schedule Dissemination for Multi-hop TDMA-based Wireless Networks

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Framing and Schedule Dissemination for Multi-hop TDMA-based Wireless Networks

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Abstract

TDMA-based protocols with a connection setup mechanism can be used to provide QoS guarantees in a network. Wireless Mesh Networks (WMNs) are multi-hop in nature and supporting QoS intensive application on them requires multi-hop TDMA-based protocol. However, most designs of TDMA-based protocols are limited to single-hop based settings.

In this work, we have designed, implemented and evaluated a centrally-controlled TDMA-based MAC protocol for multi-hop wireless networks. To accommodate the changing traffic requirements of the network, the protocol makes use of dynamic scheduling, and therefore, has a mechanism for schedule dissemination. This makes it suitable for real-time voice/video applications.

Our design is not limited to any particular wireless technology. The implementation is carried out on tmote-sky. The achieved throughput is upto 90% of the optimal throughput, which is good enough to support 1-2 GSM-quality voice calls simultaneously over three hops, and even more, over lesser number of hops. We have also conducted feasibility studies for 802.11-based platforms and concluded that an implementation could provide throughput greater than that of the existing CSMA/CA MAC.

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Chapter 1

Introduction

Wireless mesh networks (WMN) have become increasingly popular in the last couple of years due to advantages like low setup cost, easy network maintenance, robustness and reliable service coverage and technology-independence [1]. In WMNs, each node acts both as a host and a router, forwarding packets on behalf of other nodes, provided that the nodes are in range of one another. This property can be exploited to extend wireless connectivity in hard-to-reach locations [2].

1.1 Motivation

Outdoor IEEE 802.11-based [3] mesh networks are increasingly becoming a viable option for rural connectivity [2, 4]. This is because, the use of off-the-shelf 802.11-based hardware provides a low-cost option for the platform to be used. Additionally, rural regions in India are characterized by intermittent power supply, varying population density and low-income levels. Due to these reasons, it is not a profitable option for the service providers/operators to roll out communication technologies like GSM and WiMAX. Taking the above factors into consideration, the solutions being developed for rural connectivity should be characterized by low service-cost as well as low power consumption.

Chebrolu et al present a fresh perspective of providing connectivity to rural regions with the use of commodity 802.11 hardware [5]. The network proposed by the authors is a combination of short-distance (up to few hundred metres) and long-distance (up to few tens of kilometres) links. The short-distance links form the Local Access Networks (LAcN) and the long-distance links form the Long Distance Network (LDN) as show in Figure 1.1. The root in the LDN is the landline node, which has wired connectivity. All the other nodes in the LDN extend connectivity from the landline to a particular point in each village and are, therefore, referred to as the local gateway of the village. Connectivity from the local gateway may be extended by the means of local access links to multiple points within the village. A network with the above mentioned characteristics is termed by the authors as a FRACTEL[1] network.

Traffic ranging from web to real-time audio/video are expected to be supported

[1] wiFi based Rural data ACcess and TELephony

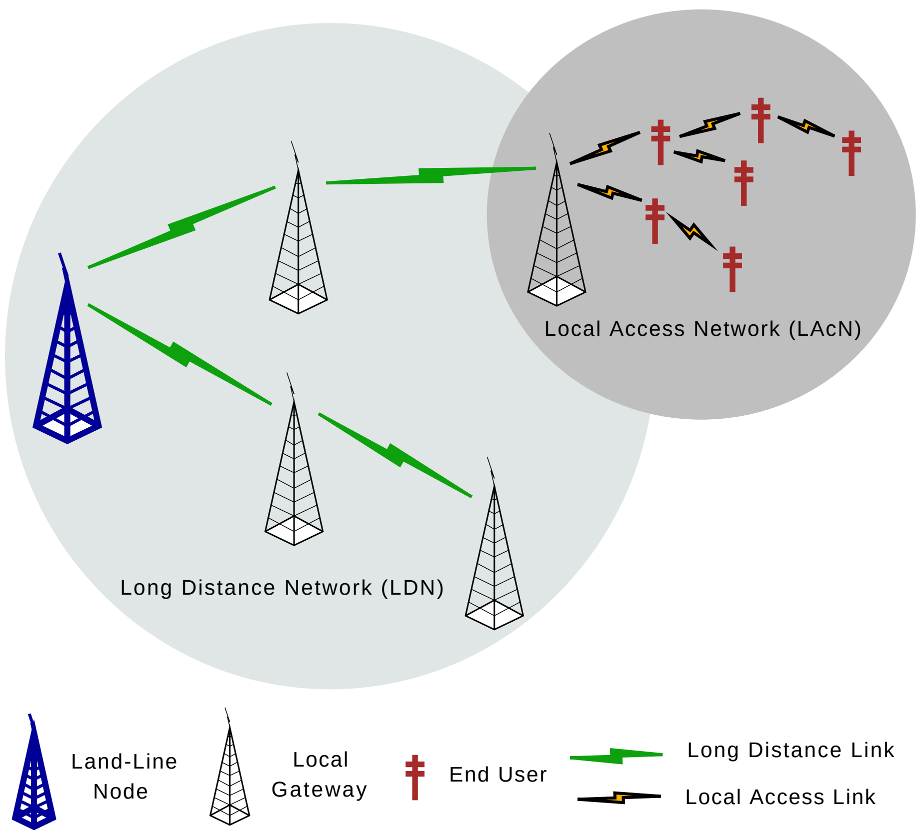


Figure 1.1: Structure of a FRACTEL Network

in FRACTEL, for which QoS guarantees have to be made. Proper spatial planning of links can lead to predictable performance [6, 7], without which it is difficult to provide QoS guarantees. However, even with proper planning, the existing CSMA/CA MAC of 802.11 does not have a provision for ensuring QoS guarantees. The network in FRACTEL has an inherent tree structure and, therefore, a centrally-controlled, multi-hop TDMA-based MAC can be used to provide QoS guarantees. The landline node can act as the central node (controller) in the LDN, whereas, in the LAcN, the local gateway can do the same job. Once the use of TDMA MAC has been assumed, the problem of providing QoS guarantees gets modified to scheduling the links at various time-slots and channels. This problem can be considered independently in the LDN and each of the LAcNs [5].

IEEE 802.15.4-based [8] mesh networks are generally deployed for monitoring physical and environmental conditions. The network is made up of devices called motes, to which need-based sensors can be attached. Such networks can be used in rural setting for carrying low-volume voice traffic too [9].

802.15.4 based multi-hop networks can be setup in rural areas with audio sensors attached to the motes. Push-to-talk and messaging applications can then be built over them [9]. 802.15.4 is poor is terms of offered data rate (256 kbps at 2.4 GHz). Therefore, to support multiple voice calls, QoS guarantees have to be made. This can be done using a TDMA-based MAC.

1.2 Problem Statement

The statement of the thesis problem is as follows:

> *To design, implement and evaluate a centrally-controlled TDMA-based MAC protocol for multi-hop wireless networks, which incorporates a connection setup mechanism, multi-hop framing, and support for dynamic scheduling, in order to provide QoS guarantees for real-time applications.*

The design aims to be generic in nature and applicable to different kinds of wireless technologies. The implementation, on the other hand, has been carried out on

802.15.4. Also included as a part of the problem, is the study of performance of the developed protocol in a network with linear topology.

1.3 Challenges

The problem poses several challenges:

- Since the design is not limited to any particular wireless technology, it should be suitable for platforms with limited capabilities also. Motes operate on small processing, memory and power constraints. Therefore, the protocol should be so designed such that it uses minimal processing and memory and has provision for power-saving.
- In a multi-hop network, the failure of one of the nodes affects all the nodes that belong to the subtree of the failed node. Taking this into consideration, the design should support quick reconstruction of the routing tree as well as end-to-end connection setup mechanism so that if an intermediate node goes down, the connection is not broken.
- Synchronization of clocks to a central node in a large network is a difficult task. Synchronization over multiple hops usually presents uncertainties. These uncertainties should be accommodated in the design of the framing and schedule dissemination mechanisms.
- The design should allow for flexibility in terms of making use of any spatial reuse available in a mesh network.
- We should also allow for dynamic scheduling, since we may have variable bit-rate real-time flows.

1.4 Our Contribution

In this thesis, a solution to the stated problem is provided. The solution set consists of the following components:

- Design of a centrally-controlled, multi-hop TDMA MAC protocol
  - A mechanism for schedule disseminition, which facilitates the dynamic scheduling. (The actual scheduling problem is not within the scope of this thesis.)
  - A mechanism for flow setup and termination with node-specific delay and bandwidth guarantees. The delay requirements are specified while connection is setup. It is up to the scheduler to actually guarantee the delay.
- Implementation of the above designed MAC protocol on tmote platform [10] using TinyOS [11].
- Study of the effects of fine- versus coarse-grained scheduling on delay and throughput of the protocol implementation on tmote in a network with linear topology.
  - The throughput achieved is more in case of coarse-grained scheduling.
  - There is a trade-off between throughput and delay.
- Study of feasibility of implementation of the above designed mechanism on 802.11b hardware.
  - By means of an actual implementation, we have shown that high-precision time-synchronization in multi-hop networks is possible using 802.11 hardware.
  - Theoretically, we have shown that the achievable throughput is good enough to carry out a practical implementation.

1.5 Organization of the Report

The rest of the thesis is organized as follows:

Chapter 2 provides a background on prior related work. Chapter 3 provides a general overview of the multi-hop framing and schedule dissemination protocol. We define various frame types and also examine the requirements of the MAC protocol.

Chapter 4 discusses the TinyOS implementation of the protocol on the Tmote sky 802.15.4-based platform. In Chapter 5, we evaluate the TinyOS implementation of the protocol. In Chapter 6, we conduct a study to see whether it is feasible to implement our protocol on existing WiFi hardware. In Chapter 7, we conclude the thesis and discuss the scope for future work.

Chapter 2

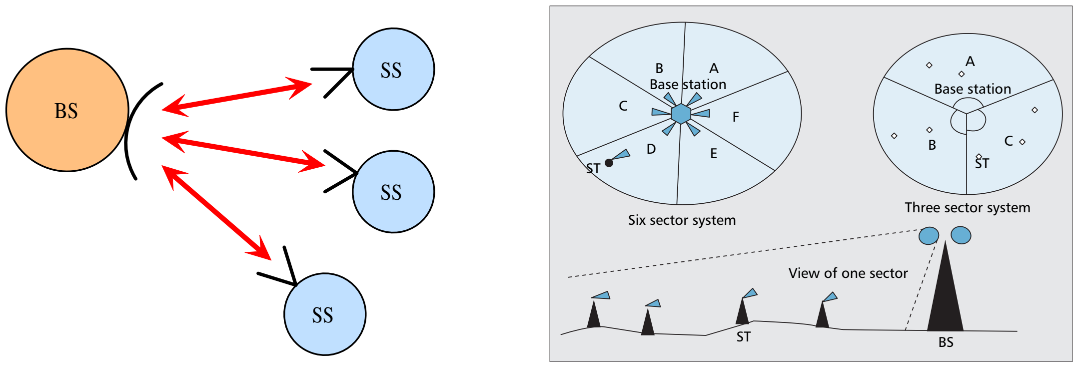
Related Work

Past work on the design of multi-hop framing mechanisms is limited. In this chapter, we discuss SRAWAN MAC [12] and Wi-Fi-Re [13], which are 802.11-based single-hop framing mechanisms, with a view of extending them in a multi-hop scenario. We also discuss the proposals of IEEE 802.16j [14] working group which seeks to extend IEEE 802.16 [15] to a multi-hop scenario. Finally, we discuss BriMon [16] from which we have partially borrowed the time-synchronization mechanism.

2.1 SRAWAN MAC and Wi-Fi-Re

Both SRAWAN (Sectorized Rural Area Wireless Access Network) [12] and Wi-Fi-Re (WiFi Rural Extension) [13] are MAC protocols based on IEEE-802.16 [15] designed to operate on 802.11 PHY. They aim to provide low-cost Internet access to rural areas over tens of kilometres.

SRAWAN operates in a point-to-multipoint fashion as shown in Figure 2.1(a). It employs a base-station with a sectorized antenna and several subscriber stations with directional antennae oriented towards the base-station. Wi-Fi-Re too operates in a point-to-multipoint fashion, but in a star topology with the base-station having multiple sectorized antennae as shown in Figure 2.1(b). In both cases, time division duplexing (TDD) is used to support uplink as well as downlink traffic.



(a) SRAWAN. (Source:[12]) (b) Wi-Fi-Re. (Source:[13])

Figure 2.1: SRAWAN and Wi-Fi-Re

In both SRAWAN and Wi-Fi-Re, a frame is divided into uplink (UL) and downlink (UL) sub-frames and has UL-Map and DL-Map. The UL and DL maps specify when each node transmits and receives. In SRAWAN, each node listens all the time and simply discards the packets not intended for itself, and therefore, it does not have a DL-Map. The time synchronization is done by ranging to take into account the propagation delay. In our design, we do not intend to divide a frame into downstream and upstream parts. We leave this decision to the scheduler.

Wi-Fi-Re and SRAWAN provide us with an insight of TDMA-based networks. We study them with a view of extending their design to multi-hop scenario.

2.2 802.16j: Multi-Hop WiMax

802.16j [14] task force is working on extending 802.16 to a multi-hop scenario. They plan to achieve throughput and coverage enhancements with the use of relay stations. A use-case of 802.16j is shown in Figure 2.2.

The figure shows how use of relay stations can help in the enhancement of coverage. Throughput can be increased by having multiple relay stations within the coverage area of the base-station but using different frequencies. There are to be provisions for two kinds of relay station namely transparent and non-transparent.

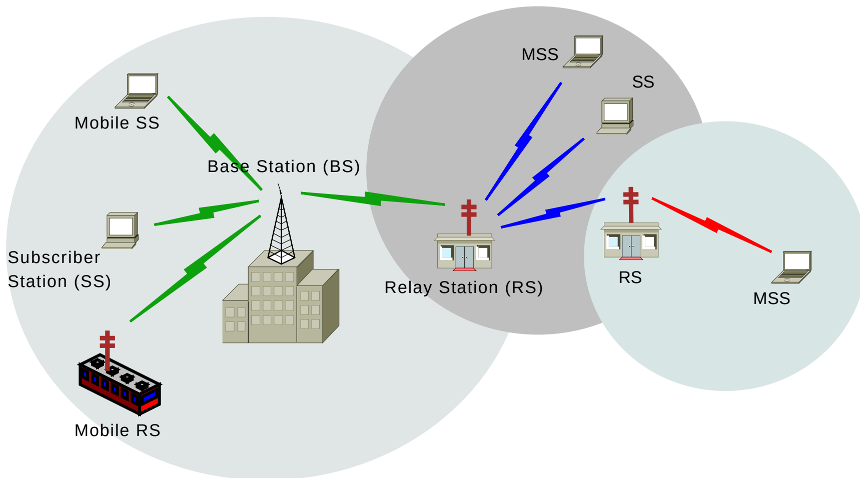


Figure 2.2: 802.16j Use-Case

The RS in transparent mode does not forward control messages forwarding only the data traffic. On the other hand, the RS in non-transparent mode forwards both data and control information.

802.16j is very closely related to our own work with similar set of requirements. However, it is still in a draft stage with no concrete design in place.

2.3 BriMon

BriMon [16] is an automated wireless sensor networks based system for railway bridge monitoring. Here, sensor nodes fitted with MEMS accelerometers are deployed on railway bridges for data collection as shown in Figure 2.3. The data collected by the sensor nodes must be time aligned for any meaningful analysis. For this purpose, the clocks of the nodes must be synchronized. The deployed sensor nodes form a multi-hop network, and therefore, a multi-hop time-synchronization mechanism is put in place.

We study this time-synchronization mechanism and use it, with a slight modification, in our implementation of a multi-hop framing and schedule dissemination on motes using TinyOS. This is discussed in detail in Section 4.4.2.

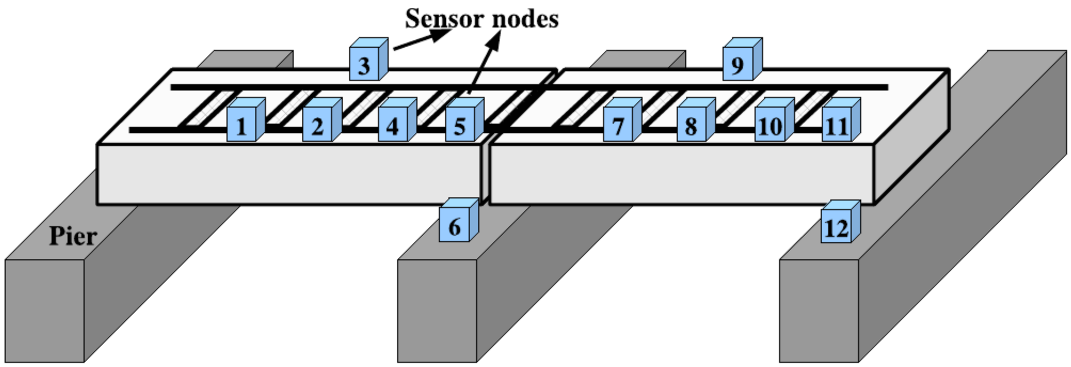


Figure 2.3: BriMon Setting (Source:[16])

The other similarity between BriMon and our work is that in BriMon too data is to be transmitted from the source to the head node over multiple hops. However, unlike our requirement, this transmission is not time-dependant, and therefore, the legacy CSMA/CA MAC protocol is used for transmission.

Chapter 3

Design of Multi-hop Framing and Schedule Dissemination

In this chapter, we present the design of the MAC protocol developed by us for multi-hop last mile wireless access. The design is not limited to any particular wireless technology. It could be applied to any suitable TDMA-based architecture. In our generic description below, we present the design of the various protocol messages and the various fields in packets which are transmitted. In this description, we have intentionally left open, details such as the number of bytes for specific packets: these will depend on the specific wireless technology to which the protocol is applied.

We will first look at the requirements of a multi-hop framing and schedule dissemination. We will then see how a data transmission is done. Based on the above we will then design the various types of packets and look at some aspects in the design. Finally we will revisit the transmission process with the exact type of packets exchanged at each step.

3.1 Requirements

The requirements of a centrally-controlled multi-hop framing mechanism is as follows:

- A TDMA-based network requires a schedule, which allots time-slots to links.

This schedule may be static or dynamic. In our setting, dynamic scheduling is required because the demands may keep changing. We assume that the central node runs an algorithm for the computation of the schedule. Therefore, we will not discuss schedule computation any further.

- In a multi-hop setting, a routing tree is required. The MAC protocol assumes an underlying routing tree, but since the network topology is not static, the routing algorithm for the construction of a routing tree would require the services of the MAC. Therefore, there is a cyclic dependency. Taking this into consideration, a routing algorithm is needed to run periodically using a flooding primitive.

- The timings mentioned in the schedule are as per a global clock. Therefore, a mechanism is required to synchronize the clocks on all nodes to the global clock. This synchronization mechanism itself has to also run on top of the TDMA MAC protocol. We will discuss the aspect of time-synchronization in greater detail.

- Once a schedule is computed, a mechanism to disseminate it to all the nodes of the network is required. The issue of schedule dissemination will be discussed in greater detail too.

- Since we wish to incorporate the idea of a flow to guarantee QoS in our protocol, we need flow and bandwidth request mechanisms. (The terminology used here is consistent with IEEE 802.16 [15]. Flow request is used for a connection setup defining the traffic requirements in terms of bandwidth and delay. Bandwidth request, on the other hand, is used to request for bandwidth in addition to the amount specified in the flow request. Bandwidth request is used to clear the existing transmit queue.)

- Finally, since the MAC protocol requires data transfer over multiple hops, we need a mechanism to relay data from the source to destination via (multiple) intermediate nodes.

Apart from the above stated requirements, there are also some others like provision for termination of an idle connection. These are implementation specific and will be explained while discussing the implementation details.

Since the requirements of time-synchronization and schedule dissemination form the central issues in the design of a TDMA-based MAC, it is only appropriate for us to discuss them in greater detail.

Time Synchronization

Time-synchronization is the crux to the design of a TDMA-based protocol. The approach to time-synchronization could be to synchronize all clocks to the fastest clock in the network, or, to synchronize the clocks to a particular global clock. In our case we chose the latter approach, using the central node's clock as the global clock. The reasoning behind this is explained with reference to Figure 3.1.

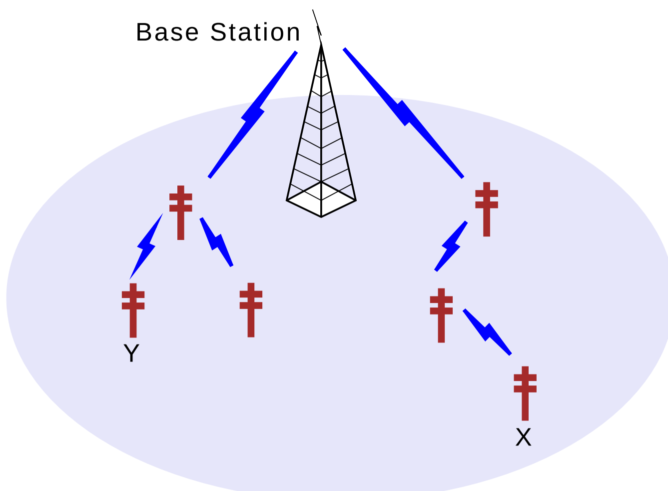


Figure 3.1: Example of a network

Consider the Figure 3.1. Let us assume that the network follows the time-synchronization algorithm where all the clocks are synchronized to the fastest clock in the network. If the clock of node X is the fastest in the network, then everytime the clocks of X and Y are to be synchronized, five clock exchanges are required as they are five hops away from each other. Now, in the same network, if we employ the time-synchronization algorithm where all the clocks are synchronized to the clock of the central node (irrespective of whichever clock is the fastest), a maximum of three clock exchanges are required for synchronization. Thus, by using the second approach and the central node's clock as the global clock, we are effectively reducing the time to synchronize the clocks, almost by a factor of two.

The time-synchronization information is passed to each node from its parent in the schedule messages.

Schedule Disseminition

The schedule sent by a node contains the transmitting information for all its descendants. A node, on receiving the schedule, extracts the transmitting information relevant to itself and to its children. It then follows the transmitting information relevant to itself and transmits the ones relevant to all its children. In this way, in addition to itself, a node also knows when its children would be transmitting.

In another approach, a node could also be informed of the transmission times of its parent in addition to that of its children. Using this information a node could remain in low power state for the duration in which it neither transmits nor receives. To facilitate the above, a schedule sent by node X (say) must contain the transmission times of X and the descendants of X. This approach is suitable for technologies which work on power constraint, e.g., sensor networks.

3.2 The Transmission Process

In this section, we list in a step-wise manner, the events that occur, when a new node joins the network and wants to transmit.

1. Node A (say) boots up.
2. It synchronizes its clock to the global clock.
3. It associates with the network (A node is deemed to be associated to the network if the central node has the routing information of that node).
4. It makes a flow request to the central node.
5. It transmits and if there is a build-up in the transmission queue, it requests for additional bandwidth.

6. It terminates the flow.

Based on the above steps, we design the types of packets required. After designing the types of packets, we will revisit the transmission process, stating the exact type of packet that will be transmitted at each step.

3.3 Types of Packets

All the types of packets share a common structure: there is a generic header, a specific header and the payload as shown in Figure 3.2. The structure of the "generic header" is common to all packets whereas the "specific header" is specific to each packet type. The payload (if any) too, depends on the type of the packet. Therefore, our discussion of a particular type of packet would involve discussing its specific header and the payload. The discussion is summarized in Table 3.1.

Figure 3.2: General Structure of a Packet

Before we describe the types of packets (and therefore, the specific headers) that are needed, it would be appropriate to discuss the generic header. The generic header houses the packet *type* field. This is used to identify the type of packet and also the payload.

In a multi-hop scenario, an intermediate node (which is neither root, nor leaf) is responsible for relaying packets towards both, the central node and the leaf nodes. Therefore, we also specify the *direction* in which the packet is being sent. This will enable a node A's (say) parent to discard packets intended for the children of node A and vice versa. The direction is specified as upstream when the packet is travelling towards the central node and downstream when the packet is travelling away from the central node. The *transmitter address* is also specified in the generic header. It is the address of the transmitter of that packet. For example, if a packet is sent from A to C via B, then the transmitter address during transmission from B to C is same as the address of B.

Adding destination address would only complicate the matter as each node would then be required to have the entire structure of its sub-tree rather than just a list of its descendants. Instead, packets could be transmitted to all neighbours and simply discarded at the destination if irrelevant to it. The combination of transmitter address and the direction specifies a set of possible destinations. Also, a *CRC* field is present in the generic header for error detection.

3.3.1 Schedule and Schedule-Fragment Packets

In a TDMA-based network, each node must know exactly when to transmit. This information is specified in the schedule. The schedule also contains the *time synchronization information* and the *start-time* of the schedule.

The payload of schedule is what we term as the *scheduling elements*. Each scheduling element specifies parameters for one transmission. There may be many such scheduling elements in a schedule. For example, to transfer data from central node to a hop-3 node, three transmissions are required (central node to hop-1, hop-1 to hop-2 and hop-2 to hop-3). Three scheduling elements are needed to specify these three transmissions. The *number of scheduling elements* present in a schedule is also specified in the schedule packet header.

Each scheduling element contains the transmitter address, the starting time of the transmission, the channel on which to transmit (if such support is to be provided), the duration of the transmission, the type of packet to be transmitted and any additional parameters associated with the packet type to be transmitted. For example, in case of data packet, these additional parameters are the values which uniquely identify a flow. We will describe these parameters in the sections that follow. The starting time of the transmission is specified relative to the start-time of the schedule. Duration of the transmission is specified in terms of number of time units, number of bytes or number of units of the packet payload. This is decided in advance for a particular implementation.

Maximum Transmission Unit (MTU) is the largest-sized (in terms of number of bytes) packet that can be sent over a network. It is possible for the schedule to be larger than the MTU for a particular technology. Keeping the above in mind, we must allow the schedule to be fragmented. We, therefore, introduce the schedule-fragment packet type. To allow fragmenting of a schedule, we need *control* field which is present in the schedule header as well as the schedule-fragment header. The control field consists of a sequence number, a fragment number and a "more" flag. The schedule-fragment header consists of only a control field and a field indicating the number of scheduling elements present in that packet.

3.3.2 Control Packet

When a node boots up, it has no knowledge of who its parent or children are. A routing protocol, therefore, needs to be run on top of the MAC protocol itself. Control packets are broadcast packets and slots specified for them in the schedule are contention-based. For these reasons, a routing algorithm can be implemented over control packets.

We do not specify any fields in a control packet or how the contention is resolved. These can be worked out on case-to-case basis. For example, a control packet itself could have many sub-types. One sub-type could be used for routing, another could be used to notify the nodes that the central node has booted, etc.

3.3.3 Flow-Request and Flow-Request-Aggregate Packets

Once the central node has the routing information of a node, that node is deemed to be "associated" to the network. The node, then, must make a flow-request to the central node in order to transmit the data. This flow-request specifies *source-address, destination-address, the requested flow number, the number of bytes requested* and the *interval of time*. If n bytes are requested and x is the time-interval, it means that the connection requires n bytes every x interval of time. Also, it is worth noting that flow number is requested rather than allotted by the central node. This is because a node can establish the confirmation of flow establishment by merely looking at the schedule rather than have a special packet sent to it to confirm flow setup.

Each node, on receiving a flow-request from its children, must relay it to the central node. For this purpose, a flow-request-aggregate packet is used. A node aggregates the flow-request packets received from its descendants, which is then included as a part of payload of the flow-request-aggregate packet. The header of a flow-request-aggregate packet contains the *number of flow-requests* included in it.

It is better to aggregate the flow requests rather than send them individually to the central node. This is because, sending individually will incur overhead of sending different packets. Also, many more scheduling elements will be required so specify when exactly each request has to be transmitted.

3.3.4 Data Packet

The data packet is used to transfer the user data between the central node and other nodes. Data is transferred in the form of flows, and therefore, a prior flow setup is required for data transfer. Each flow is uniquely identified by a combination of the source-address, destination-address and the flow number. The data packet header has as its fields the *source-address, destination-address, flow number* and the *data length.*

3.3.5 Bandwidth-Request and Bandwidth-Request-Aggregate Packets

It may so happen, that for a particular flow, a node requires to transmit more bytes than what it requested in the flow-setup process. This will result in building of the transmit queue at the node for that flow. To clear this queue, it sends a request to the central node asking for more bandwidth for that particular flow. This request is sent by the use of bandwidth-request packets. In addition to the *number of bytes requested*, the bandwidth-request specifies the flowID which is nothing but *source-address, destination-address* and the *requested flow number* taken collectively.

As in the case of flow-request, aggregation is required here too. For this purpose, we have a bandwidth-request-aggregate packet type. The header contains only the *number of bandwidth-requests* field as in the case of flow-request-aggregate packet.

A bandwidth-request packet may also be used for the purpose of terminating a flow. In the number-of-bytes-requested field, a pre-agreed value can be specified which indicates that the connection needs to be terminated.

3.3.6 Tree-Broadcast Packet

Many higher layer protocols require the services of a MAC broadcast mechanism. In a single-hop scenario, the central node simply sends a broadcast message and everyone in the network receives it. In a multi-hop scenario, however, the intermediate nodes must relay the broadcast message to their descendants. For this purpose, we have a tree-broadcast packet. As in the case of control type packet, we do not specify the fields for a tree-broadcast packet. The fields will need to be worked out on a case-by-case basis.

The difference between the tree-broadcast and a control packet lies in the fact that a tree-broadcast will follow the routing tree whereas the control packet does not assume the presence of a routing tree.

3.4 Other Aspects

3.4.1 Contention-based and Contention-Free Requests

Flow-request and bandwidth-request packets can be transmitted in a contention-based manner, or, slots can be specifically allotted for them in the schedule. The latter aspect is contention-free. We leave this, an open issue.

In a contention-free case, a bandwidth-request packet may be piggybacked with a data packet and flow-requests could be polled. In [17], the author concludes that the contention-free mode performs better than the contention-based mode in terms of throughput and delay. Therefore, in our implementation, as discussed in Chapter 4, we will use the no-contention mode.

Type of Header	**Fields Present**	**Payload Type**
generic	type direction transmitter-address CRC	specific packet
control	unspecified	unspecified
schedule	control number of scheduling elements time-synchronization data start-time	scheduling elements
schedule-fragment	control number of scheduling elements	scheduling elements
flow-request	source-address destination-address flow number number of bytes interval	none
flow-request-aggregate	number of flow-request	flow-request
data	source-address destination-address flow number length	user data
bandwidth-request	source-address destination-address flow number number of bytes	none
bandwidth-request-aggregate	number of bandwidth-request	bandwidth-request
tree-broadcast	unspecified	unspecified

Table 3.1: Summary of Packet Types

3.4.2 Packet Error and Losses

Packet losses and packet error are treated in similar fashion. The CRC field present in the generic header is used to detect if the packet is received without errors. If packet contains errors, it is simply discarded. There is no error correction mechanism. From this point onwards, the terms packet loss and packet error are used interchangeably.

A loss of a control packet is handled by the protocol which makes use of it. For example, if a lost control packet contained routing information, then its the responsibility of the routing protocol to handle it.

If any of the fragments of the schedule is lost, the entire schedule is considered lost and all other fragments are ignored. A loss in the schedule results in the node waiting for the next schedule doing nothing in this period. In technologies where multiple data rates are supported, it might be a good idea to transmit the schedule at the lowest data-rate to minimize its chances of being lost.

A loss of data packet is handled by the higher layer. We do not employ a retransmission mechanism.

A loss of request-aggregate packet is considered same as loss of multiple request packets. In case a flow-request packet is lost, there will be no connection-setup and therefore, no provision in the schedule to transmit. The flow-request is then re-sent. Similarly, in case a bandwidth-request packet is lost, there will be no allocation of extra bandwidth. A bandwidth-request is sent with possibly increased requirements.

Most protocols using the services of a broadcast mechanism do not assume reliability. Therefore, a loss in the tree-broadcast packet is not handled.

3.4.3 Central Node Reset

It is possible that a failure occurs in the central node and a reboot is needed. In such a case, the routing tree and all existing connections need to be terminated. A “reset” packet is used to implement this mechanism. The reset packet could be a sub-type of the control packet, therefore not requiring a packet type of its own. When the central node reboots, it broadcasts the reset packet and other nodes on receiving it simply forward it and reset their state.

3.5 The Transmission Process: Re-visited

Keeping in mind the various types of packets, we take a look at the transmission process again.

1. The node A (say) boots up.
2. It waits until it receives a schedule packet.
3. It receives a schedule and uses the timing information to synchronize its clock to the global clock. It now also knows when a control packet is to be transmitted and runs the routing algorithm over control packets.
4. Once the central node receives node A's routing information and adds it to the routing tree, node A is deemed to be associated to the network.
5. Once node A is associated, the subsequent schedule would contain a slot in which A could sent a flow request.
6. It sends a flow-request packet.
7. Once the central node receives node A's flow-request, it assigns node A a transmission slot in the schedule.
8. Node A transmits according to the schedule. Its queue builds up and it sends a bandwidth-request to clear this queue.
9. It is allotted more bandwidth in the next schedule and transmits accordingly.
10. Node A terminates the flow by sending a bandwidth-request packet with number of bytes request set to some pre-decided value.

The present design of the MAC protocol is very basic in nature and offers a lot of room for further enhancement. For example, a mechanism to for retransmission at the MAC layer may be devised for packet losses.

Chapter 4

Implementation on TinyOS

In this chapter, we will discuss the implementation of our protocol on TinyOS. We will begin with an overview of wireless sensor networks and of the platform used, then we will move on the designing of the components and finally we will look at some of the implementation details.

4.1 Overview of Wireless Sensor Networks

A wireless sensor network (WSN) is made of many sensor nodes. These sensor nodes consist of a processor, sensors, radio and battery. They have low processing capabilities. A sensor node is commonly called a mote. Examples of motes include Mica2, telos and tmote-sky. The motes are collectively used to monitor certain physical and environmental conditions such as temperature, pressure, vibration, etc. A sensor node is generally very small as this facilitates easy deployment. Cost of motes are variable, with low-cost (around $70) ones being available too.

With the increasing popularity of the motes, it is possible to develop application for them which are non-traditional. For example, low-cost motes with audio sensors attached to them can be used for voice communication [9].

Development of applications for motes involves considerably less effort than making similar modifications in a 802.11 compliant device driver. This is the reason why we have used mote as the platform for the implementation and evaluation of our protocol.

4.2 Overview of the Platform

We will now discuss the platform used for our implementation. The hardware platform used was tmote-sky [10] as it was easily available to us. We used TinyOS [11] as the operating system platform. We used the Boomerang version 2.0.4 which has tinyos-1.x.

4.2.1 Tmote-sky

Figure 4.1: Tmote (Source:[10])

Tmote-sky is a mote platform manufactured by Moteiv Corporation. It has a 8 MHz Texas Instruments MSP430 microcontroller with a 10 KB RAM and 48 KB flash. There is also a 1 MB external flash for data storage. It also has a 250 kbps, 2.4 GHz IEEE-802.15.4 compliant Chipcon wireless transceiver CC2420, and an integrated on-board antenna with 50m range for indoors and 125m range for outdoors. Tmote-sky has support for TinyOS. A tmote is shown in Figure 4.1. Further details can be found in the Tmote-sky datasheet [10].

4.2.2 TinyOS

TinyOS [11] is an open-source operating system and platform developed at University of California at Berkeley (UCB) for embedded sensor nodes. It is the most widely used operating system for motes. TinyOS facilitates development of concurrency-intensive applications which are data driven and work on limited memory and power requirements.

Applications developed on TinyOS have a modular framework with a set of components and interfaces. An application "wires" together the interfaces of the set of components. An interface is a set of commands and events. A command is a sub-routine which performs some action. An event is also a sub-routine which is signaled on the completion of a request. An event can be bound to a hardware interrupt. A component provides some interfaces and uses some interfaces. An interface provider implements the commands of that interface and the user implements the events.

TinyOS is written in nesC programming language which is a dialect of the C-programming language. Here, only the application specific component gets compiled and is transferred to the motes. This facilitates low memory usage and code-reuse.

4.3 Component Design

Before proceeding further, please note that we did not implement the central node. For the purpose of our evaluation we created a stub for the central node, which sent the schedule and data. This stub too runs on a mote. However, the design for the central node should essentially be the same as that of any other node with the exception of presence of a scheduler in the central node. Therefore in the implementation of the central node, a large chunk of the code will be re-used from the implementation of the non-central nodes.

The component design is shown in Figure 4.2. There are four components: SchedulerM which is present in the central node only, MHFramingM which is the principal component, SendAtTimeM which is the packetizer and performs the transmissions, and UserM which is the "higher-layer" component. We will now discuss the

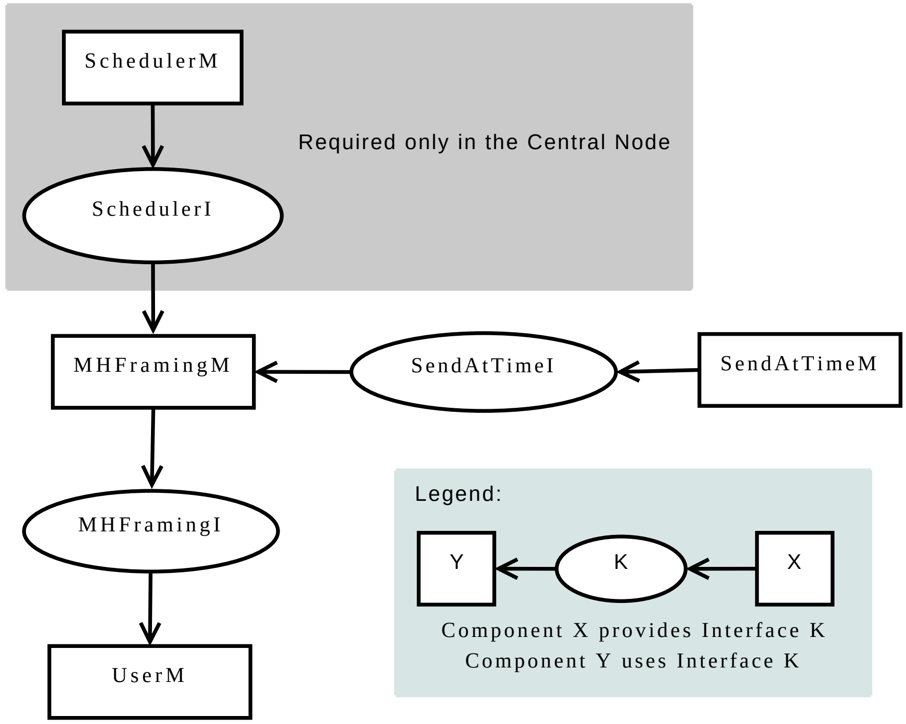


Figure 4.2: Component Diagram

tasks performed in each of the components.

4.3.1 SchedulerM

This component houses the scheduler, and therefore, it is required in the central-node only. It computes a schedule based on the traffic requirements of the network and passes it to the MHFramingM for dissemination. Since it computes the schedule, it needs to have access to the information on existing flows. Also, the flow and bandwidth requests are passed to this component to accommodate the changing traffic requirements of the network. As stated earlier, *we have not implemented the central-node code, and therefore, this component is not implemented.*

4.3.2 MHFramingM

This is the principal component. It co-ordinates the functioning of other components. The following are the functions performed by it:

- Initializes the state and buffer variables on boot-up.
- Handles received packet:
  - Accepts only the relevant packets.
  - Stores schedule, processes it to extract only the scheduling elements relevant to itself and to its descendants.
  - Passes the data packet to the UserM if the data packet are intended for itself, or, stores it if they are to be relayed.
  - Receives flow and bandwidth requests from children and stores them in a queue from which they can be transmitted in aggregation. In case of the central node, these flow and bandwidth requests are passed to SchedulerM.
- Handles flow setup and termination requests and data received from UserM (higher-layer). It also generates bandwidth request when required.
- Stores the parameters and state variables of existing connections.
- Checks periodically, for inactive connections and terminates them.
- Schedules transmissions: The schedule for this is received from parent in case of client nodes, and from ScheduleM, in case of the central node. When an alarm for a transmission is fired, the information to be transmitted (data or management) is sent to SendAtTimeM.

4.3.3 SendAtTimeM

This component is the “packetizer”. It receives the data or the management information from MHFramingM and transmits it over the radio. Before transmission,

```
interface MHFramingI {
    command flowreq(src, dst, flowid, numBytes, interval);
    /* Request a flow with numBytes every interval of time */

    command terminate_flow(src, dst, flowid);
    /* Request a flow to be terminated */

    command sendData(src, dst, flowid, *buf, buflen);
    /*Request for data to be queued for sending */

    event receiveData(src, dst, flowid, *buf, buflen);
    /* Handle received data */

    event flow_established(src, dst, flowid);
    /* Notification of flow establishment */

    event flow_terminated(src, dst, flowid, status);
    /* Notification of flow termination */
}

interface SendAtTimeI {
    command sendAtTimeT32KHz(type, *sendbuf, src, dst, direction, length, other_params);
    /* Request packetizing and transmission */

    event sendDoneAtTime(type, other_params);
    /* Notify of transmission complete */
}

interface SchedulerI {
    command getFlowRequest(flow_request);
    /* Pass flow request to scheduler */

    command getBandWidthRequest(bandwidth_request);
    /* Pass bandwidth request to scheduler */

    event newSchedule(*schedule_buf, length);
    /* Notify that a new schedule is ready */
}
```

Table 4.1: Pseudo-code for Interfaces

however, the information is encapsulated into packets by adding the relevant headers. The maximum transmission unit (MTU) in the CC2420 chip is 128 bytes out of which 12 bytes are used by the AM header in TinyOS. Therefore, the size of the packets created by this component does not exceed 116 bytes. If the amount of data (or management information) is more than what can be handled in a single packet, then multiple packets are created and are sent one after another. After sending, this component notifies MHFraming component of successful transmission.

4.3.4 UserM

This is the "higher layer" interface. An application wishing to use the MAC protocol developed by us should use this component as an interface. Apart from sending and receiving data, it also requests flow setups and terminations to MHFramingM.

Apart from the components, there are the interfaces namely MHFramingI, SchedulerI and SendAtTimeI. The pseudo-codes for these are specified in Table 4.1.

4.4 Selected Details

In this section, we will discuss some of the selected details in the implementation of our protocol on TinyOS. Other details, if required, can be obtained by looking at the source code.

4.4.1 Packet Formats and Disabling Random Backoff

The packet structure and header formats in our implementation are specified in Appendix A.2. Only the headers which are used in evaluation are shown. The rest can be obtained by looking at the source code.

Also, before moving ahead with the implementation it was necessary to disable the random backoff which is part of the inherent CSMA/CA in TinyOS. The details for this is specified in Appendix A.1.

Please note that in our implementation, we have generously used a number of bytes for the packet headers. It is possible to increase the throughput slightly by taking a more miserly approach. The design, however, remains the same.

4.4.2 Time Synchronization

Our time synchronization mechanism is borrowed from BriMon [16] with only a slight modification. As in BriMon, we maintain an offset variable at each node which has the difference between the global and the local clocks. Therefore, each time the global clock value is needed, we simply have to add the offset value to the local clock. The value of this offset variable is set to 0 in the central node since the global clock is assumed to be that of the central node.

A schedule packet carries the sender's timestamp and its offset value. On the receiver's side, timestamping is done according to the receiver's clock when a schedule arrives. The offset and the global clock values are calculated as follows:

$$\begin{aligned} receiverOffset &= (senderTimeStamp + senderOffset) - receiverTimeStamp \\ globalClock &= localClock + receiverOffset \end{aligned}$$

In BriMon, if a node misses an update, it uses the previous five updates to come up with the next update and uses it. We do not have any such mechanism. This is the only difference between BriMon's and our time-synchronization scheme. In our case, the time-synchronization information is carried by the schedule. If a schedule is missed, a node simply waits for the next schedule.

4.4.3 Routing

In our implementation, we have not implemented a routing protocol. Instead, we have hard-coded the routing information in each node. The routing protocol, if implemented, is to be run using the CONTROL packet for which we have provision in the multi-hop framing mechanism.

4.4.4 Termination of Idle Connections

In our implementation, there is a provision of termination of idle connections. For each flow, an idle count is maintained. This idle count is initialized to 0. On the arrival of a new schedule the idle count of each flow in the flow table is incremented by 1. When data is received or transmitted through a particular flow, its idle count is set to 0. As soon as the idle count reaches a certain threshold value, the flow is deemed to have been terminated.

A similar mechanism is used when requesting a flow. If a requested flow is not established within a certain number of schedules, the flow request is resent. If, even after certain number of retries, a flow is not established, it is deemed as a failure and notified to the UserM component.

Having discussed the implementation details, we will now discuss the evaluation of our protocol.

Chapter 5

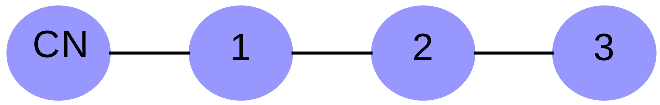
Evaluation of the Implementation on TinyOS

In this chapter, we will present the evaluation of the MAC protocol implementation on the tmote platform. We will first describe the types of experiments and their setup, then we will discuss the evaluation procedure and finally, present the various results and their analysis.

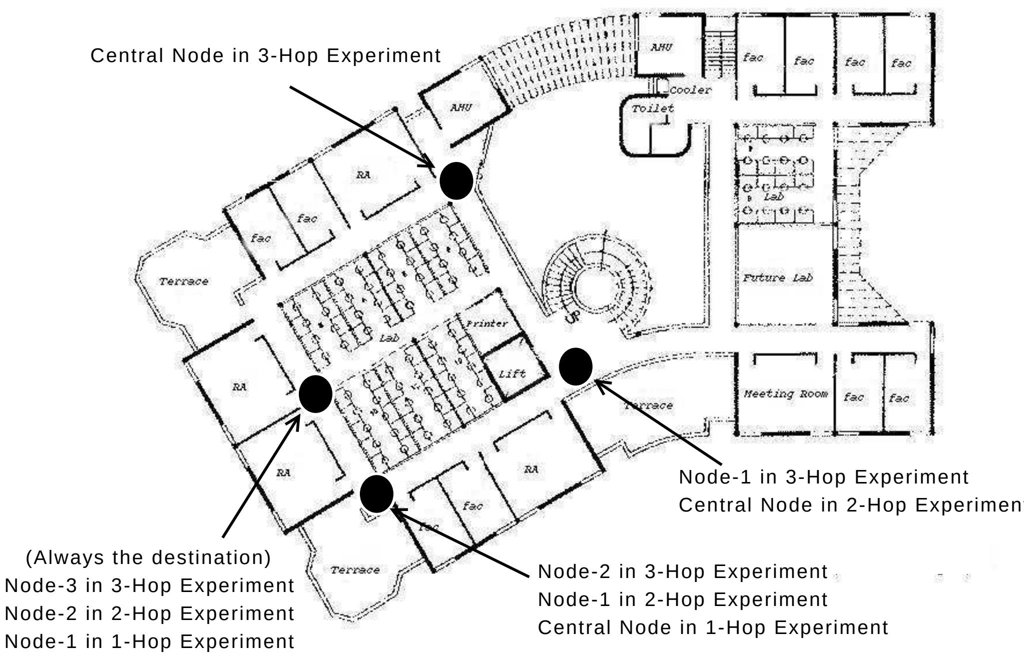
5.1 Experiment Types and Setup

For the purpose of our evaluation, we constructed a network with a linear structure involving four nodes as shown in the Figure 5.1(a). Such a topology was forced by hard-coding the parent, children and descendant information into the nodes. A particular node rejected all packets which were not received from parent or child. At the head of the network was the central node. On the above described network we conducted experiments to measure the delay, data-throughput and system-throughput.

Before proceeding further, we define the term, *frame*. *A frame is the number of bytes transmitted cumulatively, by all the nodes in the network (including the central node), from the start-time of a schedule till the start-time of the next schedule.* For example, let us consider the figure 5.1(a). If there exists only one flow which is from CN to node-3, then a frame contains the following: schedule transmitted from CN to node 1, data transmitted from CN to node 1, schedule transmitted from node 1 to node 2, data transmitted from node 1 to node 2, schedule transmitted from node 2 to node 3 and data transmitted from node 2 to node 3. The definition of the term frame is consistent with its usage in 802.16 [15].



(a) Network Topology



(b) Node Placement Plan for WGTE-MF

Figure 5.1: Experiment Setup

We now discuss the different types of experiment conducted by us.

5.1.1 Back-to-Back Experiment (BTBE)

In this experiment, the data was sent in a flow from the source (which was the central-node) to the destination (node-1, node-2 or node-3). For example, if the data was sent to node-2, then the frame consisted the following transmissions: central-node sent the schedule, followed by data to node-1, node-1 sent the schedule, followed by data to node-2.

However, the sent schedule was just a dummy. It was not followed in the network. Intermediate nodes transmitted, both, schedule and data, as soon as they finished receiving the same from their parent (and therefore the name back-to-back). At each intermediate node, the time at which data was received, was recorded. The obtained time recordings were used to construct an actual schedule, which was then used in experiment that we describe later.

For example, for n bytes of data sent from the central-node to node-2, the following events took place:

1. Central-node sent the schedule followed by n bytes of data.

2. Node-1 recorded the time t_1 at which it finished receiving n bytes of data.

3. Immediately after the above step, node-1 sent the schedule followed by n bytes of data.

4. Node-2 recorded the time t_2 at which it finished receiving n bytes of data.

Please note that the values t_1 and t_2 were with respect to the start-time of the schedule.

5.1.2 With-Guard-Time Experiment (WGTE)

In this experiment too, the data was sent from the source to the destination, but the intermediate nodes transmitted according to the schedule. In the schedule the values used for transmission times of the intermediate nodes, were the ones obtained from the BTBE. Also, a guard time of 1 slot (which is the minimum guard possible) was inserted to take into account, the synchronization error. As per our implementation, one slot was equivalent to 10 clock ticks even though the synchronization error was not more than 1-2 clock ticks.

5.1.3 WGTE with Multiple Frames and Hidden Node Scenario (WGTE-MF)

In the previous two experiments, all the nodes were kept within radio range of each other. In this experiment, we kept the nodes such that only the neighbours, as per the topology shown in Figure 5.1(a), were within the radio range of each other. Such a scenario was set on the top floor of the Computer Science and Engineering Department at IIT Kanpur. The layout is shown in Figure 5.1(b).

On the setting described above, we ran the WGTE, sending multiple frames continuously instead of just one. On the completion of one frame, the central-node initiated the next frame without any additional delay. We then recorded the time taken for the transmission of intended number of frames and averaged out this value over the number of frames. The resultant, average time taken for transmission of one frame, was used to calculate the data and the system throughputs. For example, if f frames were sent and the total time taken was t_f, then the average time per frame was $\frac{t_f}{f}$.

5.2 Evaluation Procedure

Before going any further, it is worth mentioning that in our implementation of the protocol on tinyOS, the maximum size of the payload in a data packet was 106 bytes (the MTU in tinyOS is 128 bytes). Therefore, in the experiments which we conducted, we have used the number of data bytes as multiples of 106.

5.2.1 Experiments' Order

In the experiments, we assumed that there existed only one flow in the network which was between the central-node and node-i. This we called the i-hop experiment. The values of i used were i = 1, 2, 3. In every i-hop experiment, each frame transferred j-bytes of data from the central node to node-i. The values of j used were j = 106, 212, 318, 424, 530, 1060, 1590, 2120, 2650, 3120. For each of the 30 possible values of (i, j), the following steps were followed:

1. The BTBE was run 800 times and the timings recorded. The 99-th percentile values of the recorded timings were then computed.
2. A schedule was computed based on the 99-th percentile values computed step-1.
3. The schedule was then used in 500 runs of WGTE and the end-time of each frame (with respect to the start-time of the schedule) was recorded. We then computed the 99-th percentile values of the end-time.
4. WGTE-MF was conducted using 10 frames for a 1-hop experiment and 50 frames for 2-hop and 3-hop experiments. The schedule used here was same as that computed in step-2. The time interval used between each frame was the same as that computed in step-3. Each WGTE-MF was conducted 20 times and time taken to transmit the stipulated number of frames was recorded in each case. Timings were considered only for those runs of the experiment in which all the stipulated number of frames were received successfully, others were simply discarded. We then took the 95-th percentile value of these recordings and averaged it out over the number of frames. The resultant average value, which was the per frame delay, was then used for the computation of the data and system throughputs. The obtained delay, data and system throughputs are tabulated in Section 5.3.

5.2.2 Measurement of Optimal Throughput

Apart from the above, we also carried out experiments to estimate the maximum possible throughput in tmote-sky running tinyOS platform. These experiments were carried out without the implementation of our protocol because the processing delays involved in our protocol would have resulted in lower throughput values.

We used two nodes and synchronized their clocks by the same mechanism as the implementation of our protocol. After synchronization, we sent multiple MTU-sized (128 bytes) packets between them, without adding any delay between successive packets. The timings before sending and after receiving were recorded. The difference between these times were used for measurement of optimal system throughput (OST). In our implementation, since each packet of 128 bytes can contain a maximum of 106 bytes of data, the optimal data throughput (ODT) is

$$ODT = \frac{106}{128} \cdot OST$$

The values of OST and ODT obtained by us are shown in the Table 5.1

Optimal System Throughput (OST)	∼124.7 kbps
Optimal Data Throughput (ODT)	∼103.3 kbps

Table 5.1: Optimal Throughput Values

5.3 Results and Discussion

In this part, we discuss the results obtained from the experiments conducted by us.

5.3.1 Delay

Table 5.2, shows the per frame delay in sending data bytes from the central node to the destination in WGTE-MF. For a particular destination (hop), the delay increases in a linear fashion on increasing the number data bytes. Also, for a particular number of data bytes, the delay in transmitting from the central node to a destination follows the expected pattern. The delay, is about thrice at 3-hop node and twice at 2-hop node, when compared to destination which was 1 hop away from the central node. These observations are illustrated by the graph in Figure 5.2

Number of Data Bytes in a Flow Per Frame	Per Frame Delay (in milliseconds)		
	1-Hop Experiment	2-Hop Experiment	3-Hop Experiment
106	19.21	40.88	63.76
212	28.04	58.57	90.59
318	37.15	75.96	116.82
424	45.98	93.34	143.05
530	54.54	111.03	168.38
1060	98.16	198.57	300.14
1590	141.77	285.80	430.69
2120	185.39	373.04	562.14
2650	229.00	460.28	692.98
3180	272.68	547.52	824.13

Table 5.2: Per Frame Delay for Each Experiment (from WGTE-MF)

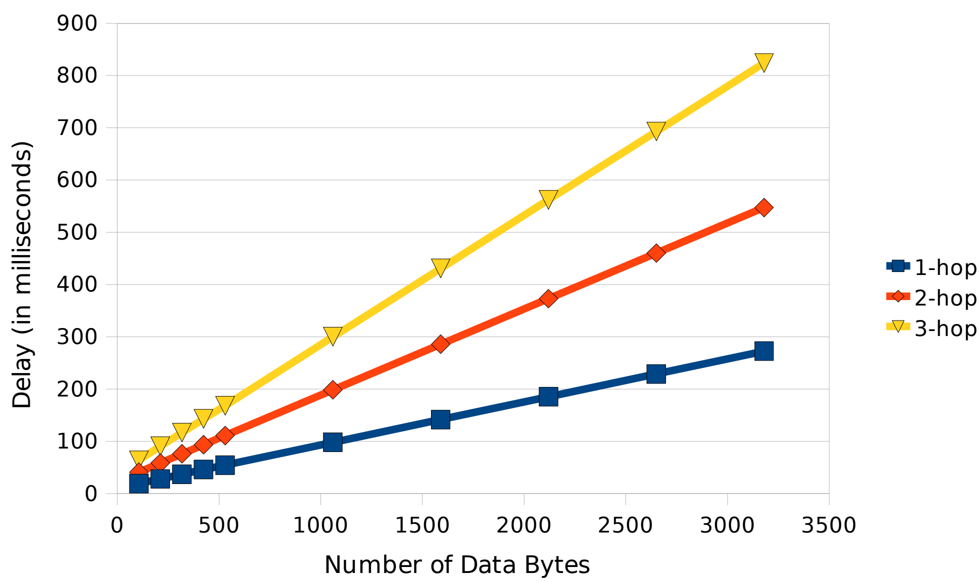


Figure 5.2: Graph: Number of Bytes in a Flow Vs Per Frame Delay

5.3.2 Data Throughput

Table 5.3 shows the data throughput for data sent in each experiment. Data throughput is calculated as follows:

$$Data\ throughput\ (in\ kbps) = \frac{Data\ in\ a\ flow\ per\ frame\ (in\ bits)}{Delay\ (in\ milliseconds)}$$

The value of the delay used to compute the throughput is taken from Table 5.2.

From Table 5.3, we see that for a particular number of bytes of data, the data throughput for a 1-hop experiment is twice that of a 2-hop experiment and thrice that of a 3-hop experiment. This is obvious since sending of data from the central node to a hop-2 node involves two transmissions and that to a hop-3 node requires three transmissions.

Number of Data Bytes in a Flow Per Frame	Data Throughput (in kbps)		
	1-Hop Experiment	2-Hop Experiment	3-Hop Experiment
106	44.15	20.74	13.30
212	60.48	28.96	18.72
318	68.48	33.49	21.78
424	73.77	36.34	23.71
530	77.74	38.19	25.18
1060	86.39	42.71	28.25
1590	89.72	44.51	29.53
2120	91.48	45.46	30.17
2650	92.58	46.06	30.59
3180	93.30	46.46	30.87

Table 5.3: Data Throughput for Each Experiment

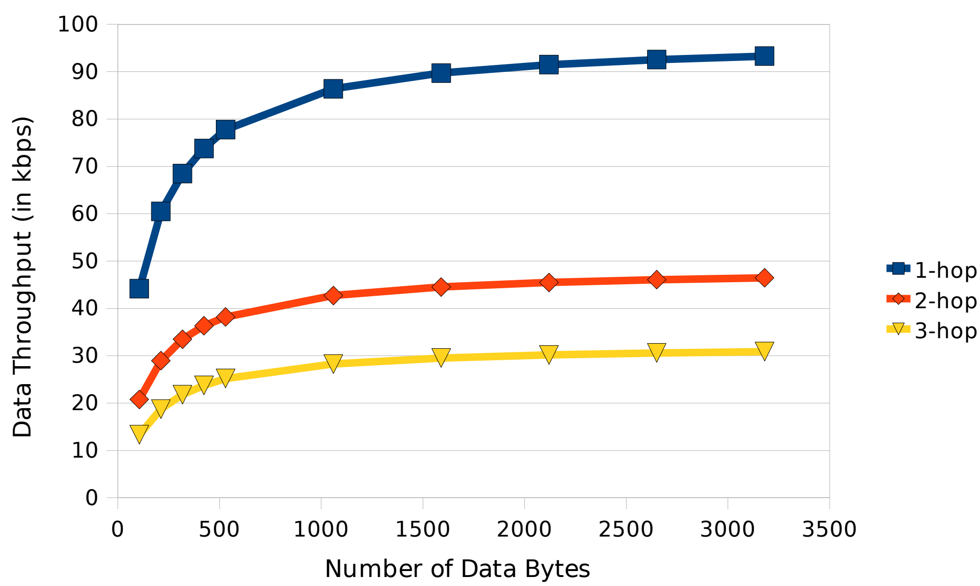


Figure 5.3: Graph: Number of Bytes in a Flow Vs Data Throughput

For a particular destination, however, with linear increase in the number of data bytes, there is a non-linear increase in the data throughput. This is explained as follows. There is a fixed overhead of the schedule irrespective of the number of data bytes. As we increase the number of data bytes, there is a decrease in the ratio of the overhead due to the schedule with respect to the data bytes. Therefore, the throughput increases. Ideally, as the number of data bytes approaches infinity, this ratio approaches zero and therefore the data throughput approaches its maximum value.

For 3-hop experiment, the data throughput varies from about 13 to 31 kbps for data bytes between 106 and 3180 per frame. All the experiments were conducted with a single flow only. With increase in the number of flows, the data throughput per flow will fall by the same factor. The GSM speech codec operates at 13 kbps, therefore, even across three hops, voice communication between two nodes could be possible.

The data throughput varies from 21 to 46 kbps for 2-hop experiment, and, from 44 to about 93 kbps for a single hop experiment. These values seem good enough for multiple voice calls simultaneously.

Also, it is worth noting that the maximum data throughput of about 93 kbps achieved by us is comparable to the optimal data throughput of about 103 kbps as stated in Table 5.1. The difference can be attributed to the processing delay in the microcontroller.

5.3.3 System Throughput

Table 5.4 shows the total number of bytes sent in a frame (cumulatively by all the nodes in the network) for each experiment. The total number of bytes includes, data bytes, data packet header, schedule and the tinyOS header. Using the total number bytes as specified here and the delay as specified in Table 5.2, the system throughput is calculated as follows:

$$System\ throughput\ (in\ kbps) = \frac{Total\ transmission\ per\ frame\ (in\ bits)}{Delay\ (in\ milliseconds)}$$

Number of Data Bytes in a Flow Per Frame	Total Number of Bytes Transmitted Per Frame and System Throughput (in kbps)					
	1-hop Experiment		2-hop Experiment		3-hop Experiment	
	Total Bytes	System Throughput	Total Bytes	System Throughput	Total Bytes	System Throughput
106	158	65.80	340	66.54	546	68.50
212	286	81.59	596	81.41	930	82.12
318	414	89.15	852	89.73	1314	89.98
424	542	94.30	1108	94.97	1698	94.96
530	670	98.28	1364	98.28	2082	98.92
1060	1310	106.77	2644	106.52	4002	106.67
1590	1950	110.04	3924	109.84	5922	110.00
2120	2590	111.77	5204	111.60	7842	111.60
2650	3230	112.84	6484	112.70	9762	112.70
3180	3870	113.54	7764	113.44	11682	113.40

Table 5.4: Total Bytes Transmitted and System Throughput for Each Experiment

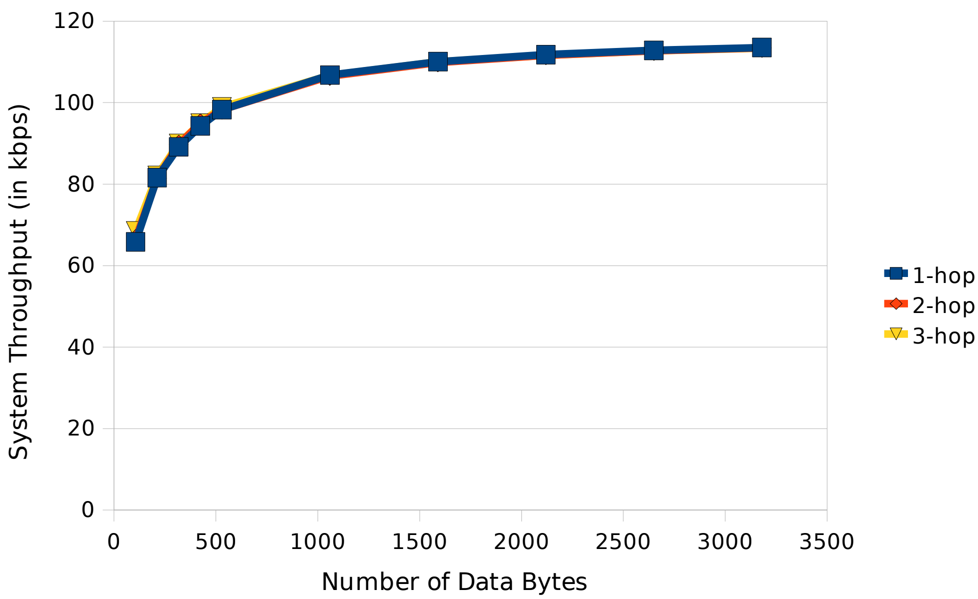


Figure 5.4: Graph: Number of Bytes in a Flow Vs System Throughput

We see that when the data is sent from the central node to a particular node, the system throughput increases non-linearly with linear increase in the number of data bytes. This is because, processing is done on per flow basis. So, for a particular flow, the amount of time spent on processing is same irrespective of number of bytes transmitted for the flow. Therefore, when less bytes are sent, more fraction of the total time is spent on processing and less on transmitting giving a lower throughput. Ideally, when the number of bytes to be sent by a flow approaches infinity, the system throughput will be maximum because the fraction of time spent on processing approaches zero.

We also see that for a specific number of bytes per flow, the system throughput is similar for experiments with different number of hops. This is illustrated by the fact that in Figure 5.4, the lines are almost co-incidental for different experiments. The above observation can be attributed to the fact that all the intermediate nodes contribute equal resources in terms of processing and transmission. Thus, for more number of hops, there is no additional processing delay at each hop.

We would like to highlight that the achievable throughput was close to the maximum. This is an indicator of the fact the we have managed to keep the processing delay in our implementation to a minimal.

Finally, to conclude the chapter, we would like to mention, that though the values obtained in our experiments are specific to the implementation of our design on TinyOS for tmote-sky, the pattern of the values should remain same across various wireless technologies. Therefore, the shape of the graphs obtained from experiments performed on other platforms should be identical to ours. In a nutshell, we do not expect the behaviour of the protocol to change on a different platform.

Chapter 6

Feasibility on WiFi

In this chapter, we will present a study on the feasibility of the implementation of our protocol on commodity 802.11 hardware. First, we will discuss about the possibility of high-precision time-synchronization on multi-hop WiFi networks. We will then calculate the achievable throughput over multiple hops assuming the use of 802.11b hardware. Finally, we will discuss whether is it feasible and worth the effort to actually implement our protocol on WiFi hardware.

6.1 Multi-Hop Time-Synchronization on WiFi

In a TDMA-based protocol, time-synchronization is a crucial aspect. Precision required is of sub-packet duration. The duration of a packet varies from a few hundreds of microseconds to a few milliseconds. Therefore, only a time-synchronization error of a few tens of microseconds is acceptable. As per our design, all the clocks in the network must be synchronized to the central node.

6.1.1 Overview of Time-Synchronization in IEEE-802.11

IEEE-802.11 standard [3] allows a maximum skew of 4 microseconds among the clocks in a basic service set (BSS). Beacon packets carry sender's timestamp and are, therefore, used for the purpose of time-synchronization. The time-synchronization algorithm varies for different modes of operation in a 802.11 network.

Time-Synchronization in Managed Mode

In the managed mode, the network is of a single-hop only with the access point (AP) at the centre. The AP sends out beacon packets periodically. A client node on receiving these beacon packets, sets its clock as per the received clock value of the AP, specified in the beacon.

Time-Synchronization in Ad-Hoc Mode

In ad-hoc mode, a multi-hop network is also possible. Here too, the clocks are synchronized using the beacons. In the network, only one beacon is sent out every beacon interval time. This is done using a distributed beaconing algorithm. Here, every node enqueues a beacon packet. At the beacon interval, all the nodes perform a random back-off before transmitting the beacon. If a node receives a beacon before its own back-off timer has expired then it simply discards its beacon from the transmit queue. The clocks in an ad-hoc network are synchronized to the oldest clock. Therefore, for a beacon, only if the sender's timestamp is greater than the receiver's timestamp, the clock of the receiver is synchronized to that of the sender.

6.1.2 Multi-Hop Time-Synchronization on Existing 802.11 Hardware

Here we will describe the multi-hop time-synchronization mechanism that has been devised by us. While it may not be possible to incorporate our mechanism in its current form in a practical implementation of the MAC protocol, it shows that high-precision time-synchronization over multiple hops to a central clock using the WiFi hardware is indeed possible.

The Approach

Since the ad-hoc mode already supports multi-hop networks, we build on top of it. We assume that the routing tree routed at the central node is known and therefore, each node has the MAC address of its parent. Each non-root node has a MAC filter using which it accepts beacons only from its parent. The time-synchronization mechanism works as follows:

1. Received beacon is accepted only when received from the parent (the root does not accept any beacons).
2. If the sender's timestamp is different from the receiver's timestamp,
    (a) The clock at the receiver is reset.
    (b) When the next beacon arrives, the sender's timestamp will be older than the receiver's, whose was just reset. Therefore, according to the synchronization mechanism of ad-hoc mode, the receiver's clock will be synchronized to that of the sender.

By repeating the above for all nodes, the root's clock is propagated down the tree.

Since we have used the open-source MADWiFi driver with a proprietary hardware abstraction layer (HAL), our approach to time-synchronization is influence by the fact that it is not possible to write values to the clock register without access to the HAL. It is, however, possible to reset the clock value to 0.

Implementation Details

We carried out the implementation on MADWiFi[1] [18]. It is the Linux driver for 802.11a/b/g cards with Atheros chipsets [19]. MADWiFi has a two-layered MAC: the HAL is proprietary and is provided in binary form only by Atheros, and net80211 stack which is hacked from FreeBSD and modified and maintained by the open-source community at [18].

[1]**M**ulti-band **A**theros **D**river for **Wi**reless **Fi**delity

The implementation was carried out on MADWiFi version 0.9.3.2 which comes with HAL version 0.9.18.0. We have tested our implementation on linux with kernel versions 2.6.11, 2.6.20 and 2.6.22. To facilitate our implementation and evaluation of the time-synchronization mechanism, the functionalities required by us were as follows:

- Filtering: We needed to ensure that only beacon packets received from the parent was accepted. The MAC address of the parent was hard-coded in the driver. For every packet received, the MAC address and the type/subtype of the packet was checked and it was accepted or rejected accordingly. A MAC filtering mechanism for managed mode is already present in MADWiFi but works only sporadically.
- TimeStamp Conversion: By default, the receiver's timestamp is a 32-bit value which wraps around every 32ms. For evaluation, we need a larger timestamp which can be tracked easily.
- For evaluation we needed the per packet header information and received timestamp which exists in the kernel space. We added code to pass this information from the kernel space to the user space using the *proc* filesystem.

Figure 6.1 shows the flow of incoming packets. It shows the order of invocation of various functions and the changes made in them. Details of each functions can be obtained from the source code which is available at [18].

Evaluation

To measure the synchronization error, we setup a testbed with 4 nodes with the topology as shown in Figure 6.2. All the four nodes were within radio range of each other. Due to the topology enforced by us (by hard coding the MAC address of the parent in the driver), each node accepted beacon packets only from its parent. However, there was no restriction on receiving data packets. Data packets could be accepted from any source.

After synchronization, one of the nodes broadcasted UDP packets to all other nodes. At each node, the receiver timestamp was stored. The receiver timestamps

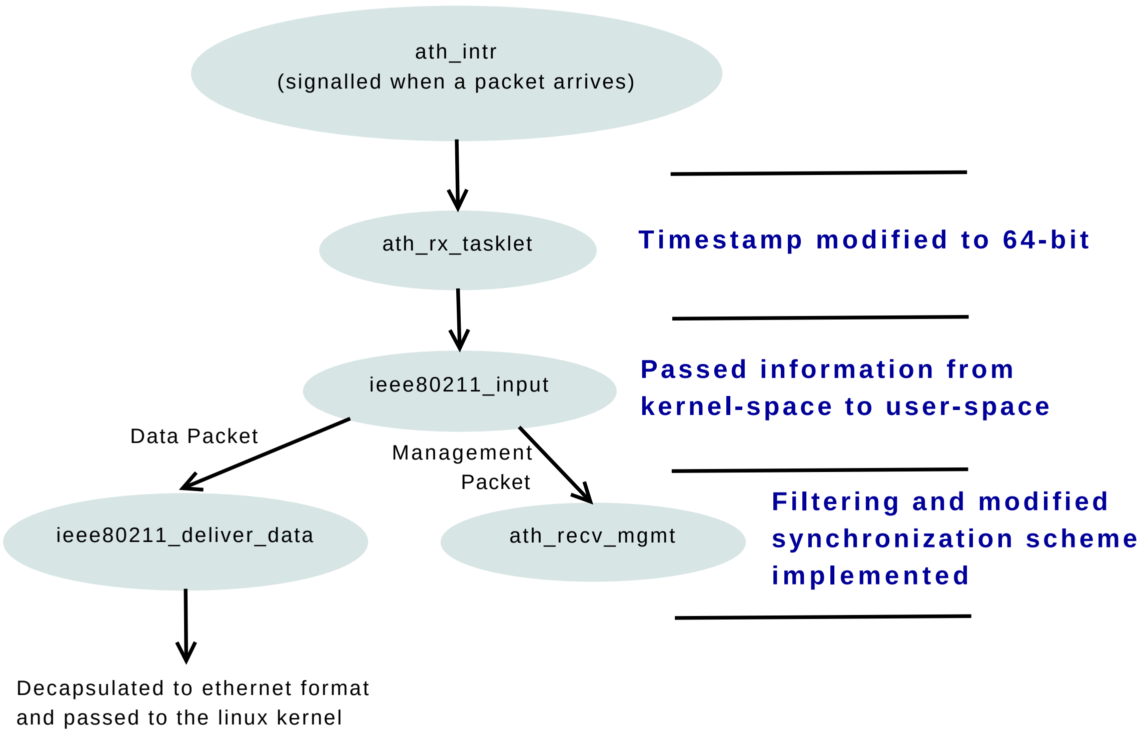


Figure 6.1: Flow of Incoming Packets

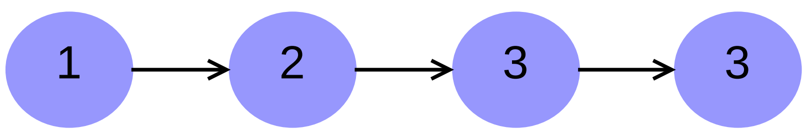


Figure 6.2: Evaluation Testbed

across the network is then compared. This is the measure of synchronization error since there is negligible propagation delay.

We conducted the above described experiment twice: once with node-1 as the UDP broadcaster and the other with node-4 as the UDP broadcaster, sending about 4500 packets each time. The first experiment allows us to measure the relative synchronization error among nodes 2, 3 and 4 and the second allows the measurement among nodes 1, 2 and 3. We then computed the PDF of the synchronization error. This is shown in Table 6.1.

As it is evident from Table 6.1, most of synchronization error are around one of the three values: -25, 0 and +25 microseconds. Ideally, the values for 1-Hop should

PDF of Synchronization Error Values (%)				
Error Range (in microseconds)	Node4 Broadcasts		Node1 Broadcasts	
	Node2-Node1 (1 Hop)	Node3-Node1 (2 Hops)	Node3-Node2 (1 Hop)	Node4-Node2 (2 Hops)
$-50 < err < -26$	0.02	0.00	0.00	0.02
$-26 \leq err \leq -23$	4.79	2.55	19.70	20.81
$-2 \leq err \leq 2$	6.42	8.70	58.52	57.01
$24 \leq err \leq 26$	88.74	88.70	21.71	22.11
$26 < err \leq 50$	0.03	0.05	0.07	0.05

Table 6.1: Synchronization Error

be identical and that of 2-Hops should be identical. However, this is not the case and no pattern is found. We conclude that the error values recorded by us are due to a timestamping delay on the receiver side. This is supported by the following:

- There is no pattern of increasing or decreasing synchronization error with time. Therefore, the reason for inconsistent values could not be due to clock drift.
- There was no pattern of increasing errors with number of hops.
- The 802.11 standard specifies the maximum limit for synchronization errors to be 4 microseconds. Assuming the hardware complies with the standard, the ± 25 microseconds errors that we see are not due to time-synchronization.

Thus, from the above, it can be concluded that the time-synchronization error achieved by us is about a few tens of microseconds or less. This is good enough for the implementation of a TDMA-based protocol.

6.2 Achievable Throughput

In this section, we will make an estimation of the achievable UDP throughput in our MAC protocol implementation on WiFi hardware. The frame formats with size assumed for this is provided in Section A.3. The calculations done by us assume the following:

- Use of 802.11b hardware:

  - Minimum bit-rate = 1 Mbps
  - Maximum bit-rate = 11 Mbps
  - SIFS = 10 μsecs
  - PHY header = 192 μsecs (24 bytes at 1Mbps)

- No random backoff since ours is a TDMA-based protocol.
- No propagation delay.
- Data packets sent at highest bit-rate (11 Mbps).
- Schedule packets sent at either lowest bit-rate or same bit-rate as the data packets.
- Guard time = 25 μsecs to account for synchronization error.
- UDP payload = 1400 bytes

Due to the presence of a TDMA-based MAC, a DIFS between subsequent transmissions is not necessary. However, we have assumed that consecutive packets are transmitted with a SIFS between them. An additional guard time is also required when transmissions are taking place across different nodes. Transmission sequence in a simple frame is shown in Figure 6.3. The figure illustrates sending of a single data packet in a frame from the central node to node-3.

Using the above transmission pattern and topology as shown in the figure, we have calculated the delay and UDP throughput for a 1-hop, 2-hops and 3-hops data transfer. In each case, we performed separate calculations for different number of data packet to be sent from source to the destination in a frame. The whole exercise was then repeated for two cases: one in which the schedule is transmitted at 1 Mbps and one in which the schedule is transmitted at 11 Mbps. The data packets are always transmitted at 11 Mbps.

In case of loss of schedule, the entire frame is lost. Therefore, we believe that in a practical implementation, the schedule should be sent at the lowest data-rate to provide extra reliability.

Our calculations are tabulated in Table 6.2. We will now show how we arrived at the figures by taking one example.

(a) Estimation of Delay

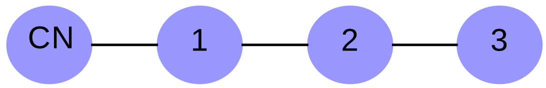
Number of Data Packets Per Frame	Delay (in milliseconds)					
	1-Hop Transfer		2-Hop Transfer		3-Hop Transfer	
	X=1	X=11	X=1	X=11	X=1	X=11
1	1.78	1.49	4.07	3.05	6.83	4.65
2	3.04	2.75	6.59	5.57	10.61	8.43
3	4.30	4.01	9.11	8.09	14.39	12.21
4	5.56	5.27	11.62	10.61	18.17	15.99
5	6.82	6.53	14.14	13.13	21.95	19.77
10	13.12	12.83	26.74	25.72	40.84	38.66
15	19.41	19.12	39.33	38.31	59.73	57.55
20	25.71	25.42	51.93	50.91	78.62	76.44
25	32.01	31.72	64.52	63.50	97.52	95.33
30	38.31	38.01	77.12	76.10	116.41	114.23

(b) Estimation of UDP Throughput

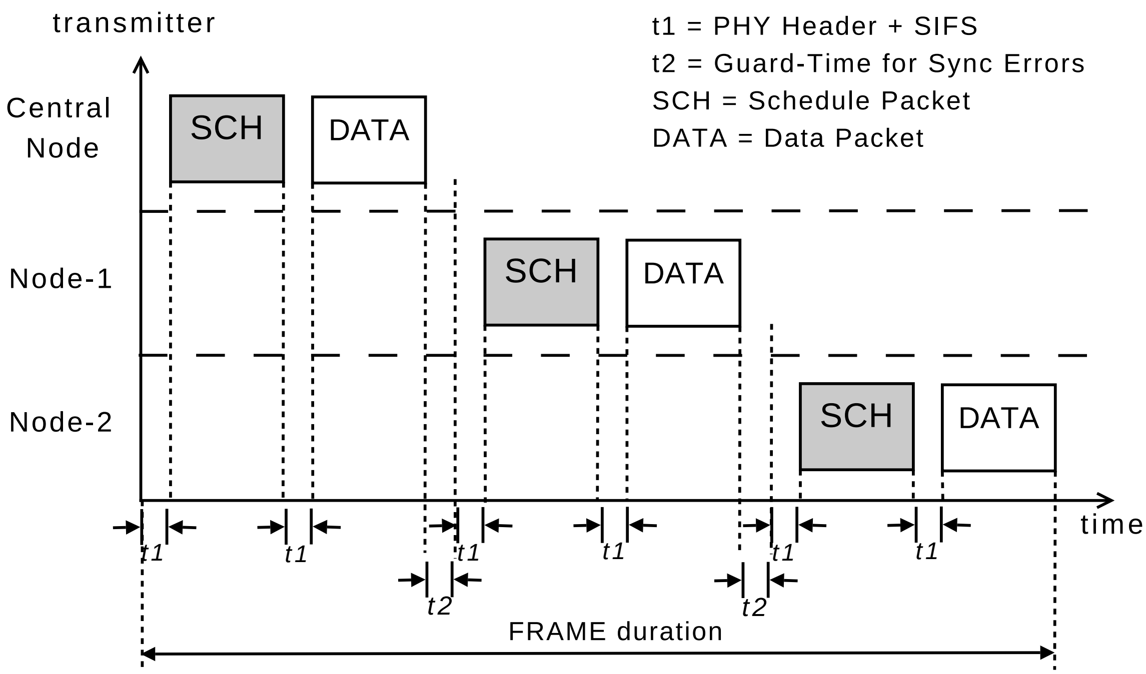
Number of Data Packets Per Frame	UDP Throughput (in Mbps)					
	1-Hop Transfer		2-Hop Transfer		3-Hop Transfer	
	X=1	X=11	X=1	X=11	X=1	X=11
1	6.29	7.51	2.75	3.67	1.64	2.41
2	7.37	8.15	3.40	4.02	2.11	2.66
3	7.81	8.38	3.69	4.15	2.33	2.75
4	8.06	8.50	3.85	4.22	2.47	2.80
5	8.21	8.58	3.96	4.27	2.55	2.83
10	8.54	8.73	4.19	4.35	2.74	2.90
15	8.65	8.79	4.27	4.38	2.81	2.92
20	8.71	8.81	4.31	4.40	2.85	2.93
25	8.75	8.83	4.34	4.41	2.87	2.94
30	8.77	8.84	4.36	4.42	2.89	2.94

Note:
X is the data rate (in Mbps) at which the schedule is transmitted
Each data packet has 1400 bytes of UDP payload
Data is transmitted at 11 Mbps

Table 6.2: Theoretical Performance Estimated for a WiFi Implementation



(a) Assumed Topology



(b) Packet Sequence

Figure 6.3: Packet Sequence in a Frame

Example

Let us consider the case where 3 data packets are to be transferred from the central node to node-3 in a single frame.

Guard time, $t_G = 25\mu secs$

PHY layer overhead and SIFS, per packet, $t_{oh} = 192 + 10 = 202\mu secs$

Total number of bytes in a schedule packet with n scheduling elements,

$$\begin{aligned} S &= (Generic\ header) + (Schedule\ header) + n * (Schedule\ Element\ Size) \\ &= 8 + 32 + (n * 30) = 40 + 30n \end{aligned}$$

At node	CN	1	2
Number of Scheduling Elements (n)	4	2	0
Size of schedule (in bytes) (S)	160	100	40
Time to transmit schedule at 1 Mbps (in $\mu secs$) $S * 8/1 + t_{oh}$	$t_{1,CN}$ 1482	$t_{1,Node1}$ 1002	$t_{1,Node2}$ 522
Time to transmit schedule at 11 Mbps (in $\mu secs$) $S * 8/11 + t_{oh}$	$t_{11,CN}$ 318.36	$t_{11,Node1}$ 274.73	$t_{11,Node2}$ 231.09

For UDP packet of 1400 bytes payload, 8 bytes of UDP header, 20 bytes of IP header, 18 bytes of data header and 8 bytes of generic header is required. Therefore total number of bytes is 1454.

Time required to transmit 1454 bytes at 11 Mbps,

$$t_a = \frac{1454 * 8}{11} = 1057.45 \mu secs$$

Therefore, total time to transmit one data packet with 1400 bytes of UDP payload,

$$t_D = t_a + t_{oh} = 2202 + 1057.45 = 1259.45 \mu secs$$

For transferring 3 packets from central node to node-3, total time required:

when schedule sent at 1 Mbps,

$$\begin{aligned}
t_1 &= (t_{1,CN} + 3 * t_D) + t_G + (t_{1,Node1} + 3 * t_D) + t_G + (t_{1,Node2} + 3 * t_D) \\
&= (1482 + 3 * 1259.45) + 25 + (1002 + 3 * 1259.45) + 25 + (522 + 3 * 1259.45) \\
&= 14391.05 \mu secs = 14.39 ms
\end{aligned}$$

UDP throughput $= 3 * 1400 * 8/t_1 = 33600/14391.05 = 2.33 Mbps$

when schedule sent at 11 Mbps,

$$\begin{aligned}
t_{11} &= (t_{11,CN} + 3 * t_D) + t_G + (t_{11,Node1} + 3 * t_D) + t_G + (t_{11,Node2} + 3 * t_D) \\
&= (318.36 + 3 * 1259.45) + 25 + (274.73 + 3 * 1259.45) \\
&\quad +25 + (231.09 + 3 * 1259.45) \\
&= 12209.23 \mu secs = 12.21 ms
\end{aligned}$$

UDP throughput $= 3 * 1400 * 8/t_{11} = 33600/12209.23 = 2.75 Mbps$

These values are shown in Table 6.2, under 3-hop transfer for number of packets per frame equal to 3.

6.3 Discussion

We saw that high-precision time-synchronization needed for the implementation of a multi-hop TDMA protocol is possible. We have also shown by calculations that using 802.11b hardware, the achievable UDP throughput over a single hop can go up to 8.8 Mbps. This value is more than what the existing CSMA/CA protocol can provide.

The achievable UDP throughput over 3-hops, varies from 1.64 to 2.94 Mbps. A video flow requires bandwidth of 384 kbps. Therefore, in practical implementation as per our design it should easily be possible to support 3-4 video flows over three hops. This number would increase significantly for lesser number of hops.

Also, the throughput does not drop significantly if the schedule is sent at low data-rate. This should definitely be considered as it would lower the loss-rate and increase the throughput, especially in a noisy environment.

Considering the above discussed factors, it is feasible to implement our design on the existing 802.11b hardware.

Chapter 7

Conclusion and Future Work

7.1 Conclusion

In this work, we have designed a framing and schedule dissemination mechanism for multi-hop TDMA-based wireless networks. We carried out the implementation for 802.15.4-based platforms using TinyOS environment. We carried out our evaluation on this implementation, measuring throughput and delay for various frame sizes. We saw that as the frame size increases, the throughput increases but the delay increases too. Thus, there is a trade-off between throughput and delay. Also, as the number of hops increases, the throughput decreases by a factor equal to the number of hops.

The GSM full-rate speech codec operates at 13 kbps. Assuming this is used for development of push-to-talk applications on motes using our protocol as the underlying MAC, 1-2 simultaneous voice calls is likely possible over 3 hops. This number increases if the number of hops decrease. Therefore, if the requirement is such that more than 2 connections over 3 hops is required simultaneously then our protocol may not be suitable. However, voice calls over motes is targeted for rural areas, and so, it is unlikely that more than 2 calls will be active simultaneously.

We have also conducted a study to see whether an implementation of our protocol on WiFi hardware is feasible. Since time-synchronization is the crux to any implementation of a TDMA-based protocol, we implemented a multi-hop time-synchronization mechanism which shows that it is indeed possible to achieve high-precision time-synchronization over multiple hops using WiFi hardware. We also performed theoretical calculations for the achievable throughput assuming 802.11b hardware. The results show that a WiFi implementation is feasible.

7.2 Future Work

With the kind of results obtained, we envision the following as future work:

- Optimization of our implementation on TinyOS to increase throughput. This can be done by reducing the number of bits/bytes per field in the headers. The design would remain the same.
- Since the evaluation we conducted was based on downstream traffic, it would also be good to conduct some studies based on upstream traffic. This could be done with variable bit-rate video flows.
- The evaluation we have done is for a single flow in a linear topology. This could be repeated for multiple flows in a more complex network topology to gain a better understanding.
- Design and implementation of a push-to-talk voice application for 802.15.4-based platforms.
- Since we have shown theoretically, that our protocol would perform well when implemented on WiFi, an implementation could be carried out.

Appendix A

A.1 Disabling Random Backoff on TinyOS-1.x

Random backoff can be disabled in the Boomerang version of TinyOS by making changes to the file *CC2420RadioM.nc*. This file can be found in the directory */opt/moteiv/tos/lib/CC2420Radio/*

The events *MacBackoff.congestionBackoff* and *MacBackoff.initialBackoff*, when signaled return the number of time periods to backoff. Replacing the signaling of these events with a 0 value, will cause the backoff period to be 0. In other words, backoff will be *disabled*.

A.2 Packet and Header Formats in TinyOS Implementation

Packet format and selected headers in our implementation of the protocol on TinyOS is specified here.

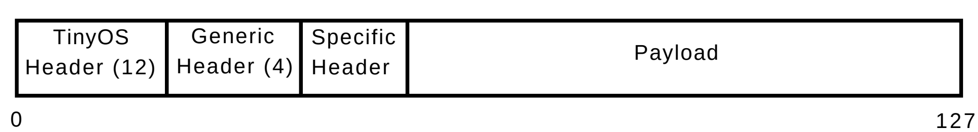
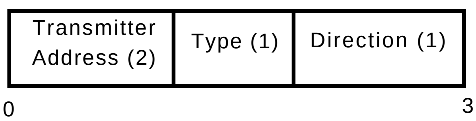
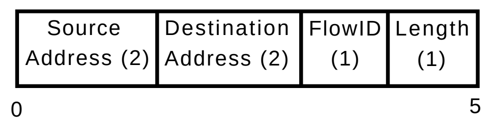


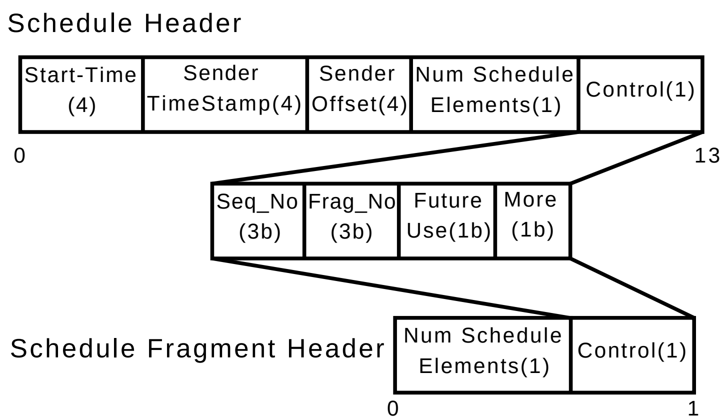
Figure A.1: General Structure of a Packet in TinyOS implementation



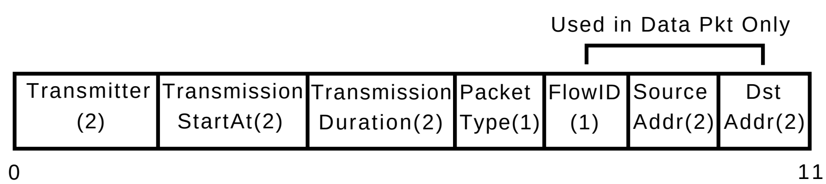
(a) Generic Header(4)



(b) Data Header(6)



(c) Schedule Header(14) and Schedule Fragment Header(2)

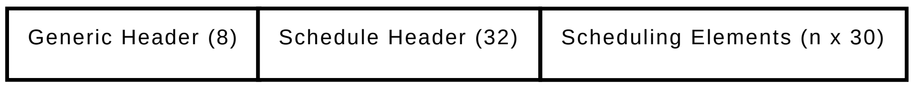


(d) Schedule Element(12)

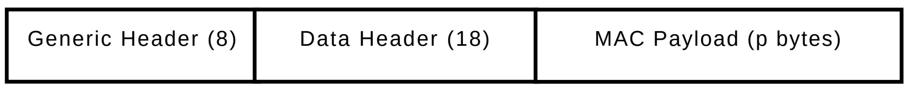
Figure A.2: Header Formats (in bytes) in TinyOS implementation

A.3 Packet and Header Formats for WiFi Calculations

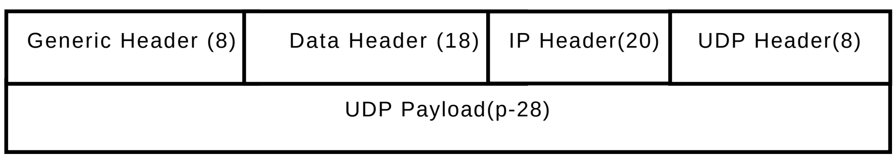
Header and packet formats assumed by us for theoretical computation of delay and UDP throughput values in Section 6.2, are shown in Figures A.4 and A.3.



(a) Schedule Packet



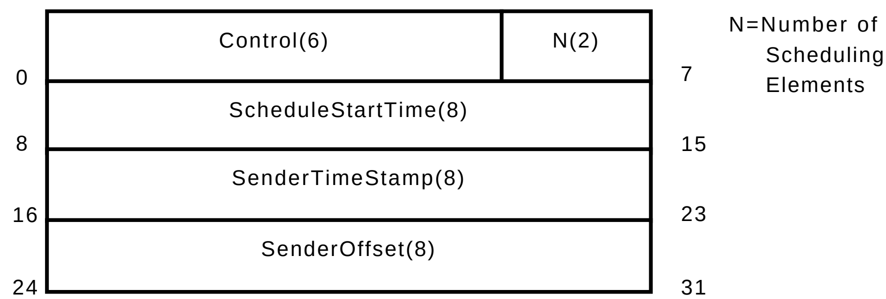
(b) Data Packet (MAC Layer)



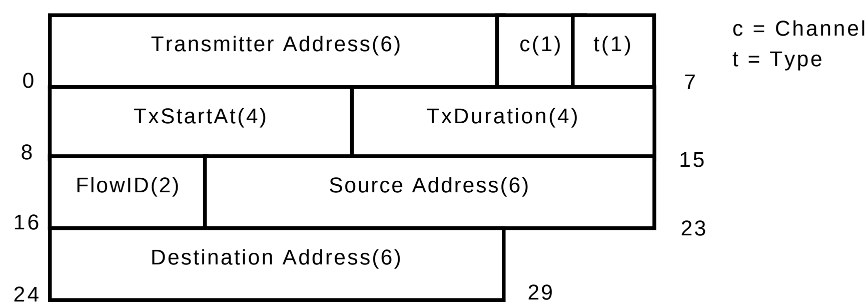
(c) Data Packet (UDP)

Figure A.3: Packet Formats Assumed in WiFi

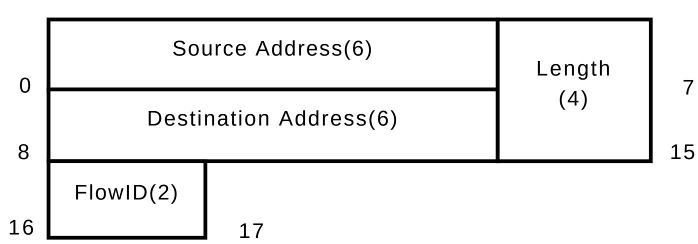
(a) Generic Header (8)



(b) Schedule Header (32)



(c) Schedule Element (30)



(d) Data Header (18)

Figure A.4: Header Formats (in bytes) Assumed in WiFi

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