Self-organizing Wireless Broadband Multihop Networks with QoS Guarantee

- Protocol Design and Performance Evaluation

Von der Fakultät für Elektrotechnik und Informationstechnik der Rheinisch-Westfälischen Technischen Hochschule Aachen zur Erlangung des akademischen Grades eines Doktors der Ingenieurwissenschaften genehmigte Dissertation

vorgelegt von

Master of Science
Bangnan Xu

aus Zhejiang

Berichter: Universitätsprofessor Dr.-Ing. Bernhard Walke
Universitätsprofessorin Dr. rer. nat. habil. Carmelita Görg
Tag der mündlichen Prüfung: 11. July 2002

ABSTRACT

New multimedia services accessible via the Internet are strongly driving the demand for the residential and commercial broadband wireless networks.

In this thesis the author proposes a channel-oriented solution - W-CHAMB (Wireless CHannel-oriented Ad-hoc Multihop Broadband) network - for self-organizing broadband multihop networks. In comparison with HiperLAN/2, the advanced features of W-CHAMB are its multihop ability and decentralized Medium Access Control (MAC). In comparison with IEEE 802.11a, W-CHAMB achieves a much higher network efficiency for short to medium size packets. More importantly, IEEE 802.11a has no means to support QoS of real time services in a multihop environment. W-CHAMB is a convergence of HiperLAN/2 and IEEE 802.11a.

The design concept based on the ISO/OSI reference model provides the guideline of the protocol design. After the hidden station problem and the unsuitability of existing MAC schemes are discussed, the thesis presents new ideas and solutions of W-CHAMB air interface for self-organizing multihop networks with QoS guarantee.

For the purpose of performance evaluation a simulation tool based on the prototypic implementation of protocols in SDL is developed. The multihop traffic performance of W-CHAMB, IEEE 802.11a and HiperLAN/2 is intensively evaluated stochastically using the simulation tool. In addition to the computer simulation, W-CHAMB air interface is mathematically modeled using probability theory and the two dimensional Markov Chain. The analytical results are compared with the simulation results to validate each other.

KURZFASSUNG

Die neuen Internet Multimediadienste haben eine starke Nachfrage nach kommerziell und zu Hause nutzbaren drahtlosen Breitbandnetzen geweckt.

In dieser Arbeit schlägt der Autor eine kanalorientierte Lösung - W-CHAMB (Wireless CHannel-oriented Multihop Ad hoc Breitband) Netz - für selbstorganisierende Breitband Multihop Funknetze vor. Im Vergleich mit HiperLAN/2 sind die besonderen Eigenschaften von W-CHAMB die Multihopfähigkeit und die dezentralorganisierte Zugriffsteuerung zum Funkmedium (MAC). Im Vergleich mit IEEE 802.11a erreicht W-CHAMB einen viel höheren Netz-Durchsatz für kleine bis mittelgroße Pakete. Außerdem verfügt IEEE 802.11a über keine Möglichkeiten, die Dienstgüte von Echtzeit-Diensten in einem Multihop Netz zu unterstützen. W-CHAMB ist aus der Konvergenz von HiperLAN/2 und IEEE 802.11a entstanden.

Das Konzept des Protokollentwurfes basiert auf dem ISO/OSI Referenzmodell. Nach der Diskussion des Problems der versteckten Station und der mangelnden Eignung der bisher standardisierten MAC Protokolle damit umzugehen, stellt der Autor die neuen Ideen und Lösungen von W-CHAMB vor.

Für die simulative Leistungsbewertung wird ein Simulator entwickelt, der auf der prototypischen Implementierung von Protokollen in SDL basiert. Die Multihop Dienstgüte von W-CHAMB, IEEE 802.11a und HiperLAN/2 werden intensiv stochastisch mit dem Simulator untersucht. Außer durch Computersimulation wird die W-CHAMB Funkschnittstelle durch Wahrscheinlichkeitstheorie und mit zweidimensionalen Markovketten mathematisch modelliert und analysiert. Die analytischen Ergebnisse werden durch die Simulationsergebnisse validiert.

CHAPTER 1

Introduction

The future wireless network will feature broadband transmission, multimedia support, self-organization and QoS guarantee. New multimedia services accessible via the Internet are strongly driving the demand for the residential and commercial broadband wireless networks. Meanwhile, the development of the high performance radio technology has made the broadband wireless networking technology feasible.

Due to the high bandwidth needed for the broadband transmission, the 5 GHz band that has been opened for the personal communication as license-exempt frequency spectrum is the most promising frequency spectrum for broadband wireless networks. Figure 1.1 shows that 455 MHz *license-exempt* spectrum in the 5 GHz band has been allocated in Europe [41, 44, 115]. 300 MHz of *unlicensed* spectrum is available in the USA, and 100 MHz in Japan. Specifications for Power Control (PC) and Dynamic Frequency Selection (DFS) exist in Europe for the efficient use of this spectrum band. The values given for the Equivalent Isotropic Radiated Power EIRP in Figure 1.1 relate to the maximum average transmit power in Europe. PC or DFS is not required in the USA. There the EIRP values relate to the maximum peak transmit power. In Japan *carrier sensing* every 4 ms is required for the operation at 5 GHz.

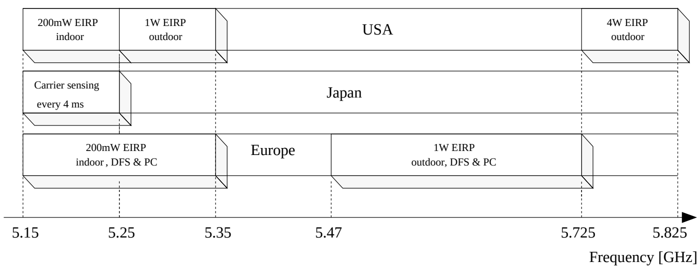


Figure 1.1: License-exempt frequency spectrum at 5 GHz

As 5 GHz or higher frequencies have very unpredictable propagation characteristics and very limited ability to penetrate an obstruction, the communication zone is irregular in shape and there exist severe shadowing problems. In most cases, communication is only possible between wireless stations with line of sight connections. Thus, multihop transmission must be considered to overcome the shadowing problem and to achieve a reasonable communication coverage. Self-organization is a very attractive feature for broadband wireless networks with multihop capability [129].

In Europe, HiperLAN/2 was standardized by the ETSI project BRAN in April 2000. In the U.S., the standard IEEE 802.11a has been developed to extend IEEE 802.11. IEEE 802.11 is a WLAN standard that provides detailed medium access control (MAC) and physical layer (PHY) specifications [61]. Both systems, ETSI HiperLAN/2 and IEEE 802.11a, operate at the 5 GHz frequency band and provide transmission rates up to 54 Mb/s [27, 28, 60].

IEEE 802.11a uses the MAC protocol of IEEE 802.11 that was originally developed for wireless Ethernet networks with a transmission rate of 1 to 2 Mb/s [61]. The MAC protocol of IEEE 802.11 is not suited for broadband wireless networks due to its inefficiency for transmission of short data packets and no QoS guarantee [8, 127, 132]. Supporting multimedia services with QoS guarantee is essential for the future broadband wireless networks.

ETSI HiperLAN/2, on the other hand, can support high performance multimedia services with QoS guarantee. But its centralized MAC solution is not suited for self-organizing networks [122, 123]. Due to the propagation conditions at the 5 GHz license-exempt frequency band the application scenarios of HiperLAN/2 are limited. It is important to extend the HiperLAN/2 to be a decentralized self-organizing network with QoS guarantee.

The objective of this thesis is to design and evaluate a self-organizing broadband wireless network with multihop transmission capability and QoS guarantee for high performance multimedia services.

The thesis is organized as follows:

Chapter 1 is a short introduction of the motivation to design a self-organizing wireless broadband multihop network with QoS guarantee. Multihop transmission and self-organization are viewed as two important issues of the future broadband wireless networks.

Chapter 2 describes three broadband wireless systems, HiperLAN/2, IEEE 802.11 and W-CHAMB (Wireless CHannel Oriented Ad hoc Multihop Network [132]), that are expected to take a significant role in the next generation wireless communications.

Chapter 3 presents the design concept based on the ISO/OSI reference model. The hidden station problem and the unsuitability of existing MAC schemes for self-organizing broadband networks are reviewed.

Chapter 4 presents the main contributions of this thesis namely the design of protocols and algorithms of the W-CHAMB network.

Chapter 5 describes details of the simulation tool that is based on the prototypic implementation of protocols in SDL.

Chapter 6 evaluates the W-CHAMB traffic performance. The W-CHAMB MAC protocol is modeled and analyzed using probability theory and the two dimensional Markov chain method. The analytical results are compared with the simulation results. The mathematical model reveals insights of the W-CHAMB air interface behavior. Using the stochastic simulation tool the integrated W-CHAMB traffic performance is studied intensively and more insights of the W-CHAMB traffic performance are revealed.

Chapter 7 is dedicated to the comparison of W-CHAMB with IEEE 802.11a and HiperLAN/2. The multihop traffic performance of the three broadband networks is intensively compared.

Chapter 8 concludes the thesis.

CHAPTER 2

Broadband Wireless Networks

After the standardization of the 3G (3rd generation) systems, researchers are beginning to develop concepts and technologies for the wireless systems beyond 3G. In the ITU/R international wireless system standardization organization, WP-8F was organized in November 1999 to investigate and standardize the wireless systems beyond 3G. In [85], a concept and possible technologies for the systems beyond 3G were presented. Higher transmission rate and lower system cost are regarded as the key requirements of the system beyond 3G. In [10], a vision for the future of wireless communications beyond 3G, which consists of a combination of several optimized access systems connected to a common IP based medium access and core network platform is proposed.

Major applications of the system beyond 3G will require a transmission rate of at least 20 times higher than that of 3G. This chapter introduces three broadband wireless networks, ETSI HiperLAN/2, IEEE 802.11a and W-CHAMB, that are expected to take a significant role in the future wireless communications beyond 3G.

2.1 ETSI HiperLAN/2

HiperLAN/2 is a reservation oriented high performance radio technology specified by the European Telecommunication Standards Institute (ETSI) [20, 21, 28]. The system operates at the 5 GHz license-exempt frequency band [27]. HiperLAN/2 provides a uniform broadband wireless access platform to carry any type of user data, such as Ethernet frames, ATM cells and IP packets [18, 19, 23, 24, 25, 26].

2.1.1 HiperLAN/2 MAC protocol

In HiperLAN/2, a fully centralized MAC protocol based on Dynamic Slot Assignment (DSA++ [121]) is specified [20, 21]. A central controller assigns radio resources within a HiperLAN/2 MAC frame. The assignment of resources may change from frame to frame. The fixed length of a HiperLAN/2 MAC frame ($t_{frame} = 2ms$) consists of three major phases as shown in Figure 2.1. The broadcast phase is used to transport the control information and carries the Broadcast Control Channel (BCCH), the Frame Control CHannel (FCCH) and the Random Access Feedback Channel (RFCH). The BCCH transmits control information through the Broadcast CHannel (BCH) protocol data unit (PDU) in each MAC frame. It provides information about transmission power levels, starting point and length of the Frame CHannel (FCH) and the Random CHannel (RCH). The BCCH is 15 bytes long and is transmitted using the most robust modulation scheme available, i.e. BPSK 1/2 [27]. The RFCH provides information through the access feedback CHannel (ACK) on access attempts made by WSs in the RCH of the previous MAC frame. The downlink, direct link and uplink phases are used to transport data PDUs between the central controller and WSs or between WSs directly. The random access phase is used for the initial access to the network, for handover indication and for requesting radio resources.

Two kinds of PDUs, the long PDU (LCH PDU) and the short PDU (SCH PDU) are specified. A LCH PDU is 54 bytes long and contains 48 byte payload, which is the same as that of an ATM cell. An SCH PDU is 9 bytes long and contains 52 bits for signaling data. In order to reduce physical overhead, all LCH and SCH PDUs in one MAC frame belonging to connections of the same MT are combined to a PDU train. A detailed description of the HiperLAN/2 MAC protocol can be found in [20].

2.1.2 HiperLAN/2 HEE

The communication between home devices including the support of IEEE 1394 is another purpose of HiperLAN/2. The HiperLAN/2 Home Environment Extension (HEE) specifies an ad hoc network configuration to allow plug-and-play operation [22]. The HiperLAN/2 HEE uses the same MAC protocol as described in Section 2.1.1. The central control function is per-

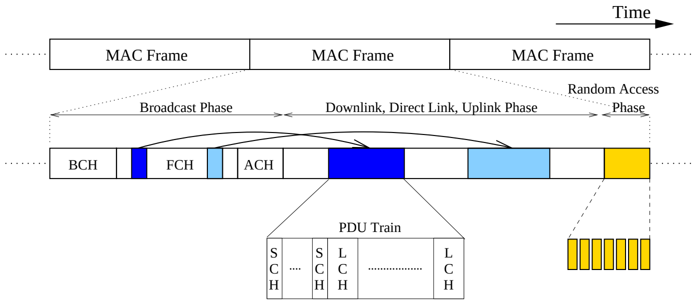


Figure 2.1: HiperLAN/2 MAC frame

formed by a Central Controller (CC). The network is self-organized by two functions: CC selection and CC handover.

The HiperLAN/2 HEE is designed for one single cluster. A multihop networking concept is presented in [54] to extend HEE to a multi-cluster and multihop network. Terminals of two different clusters may communicate via terminals that are able to participate in both networks. A terminal can participate in two clusters at the same time, if it is in the transmission range of both CCs in the respective clusters. In [91] a Multiple-Frequency Forwarder concept is presented to realize such a terminal.

To extend the HEE to a multihop network can be viewed as a centralized solution for a self-organizing multihop network. CC selection and CC handover may be suited for a home environment, but appear to be too complicated for a large outdoor network. Moreover, Multiple-Frequency Forwarder and multi-cluster operation suffer from the inherent spectrum inefficiency.

2.1.3 HiperLAN/2 with FMT

To enhance the communication range of HiperLAN/2, a Forwarding Mobile Terminal (FMT) concept is presented in [38]. A Remote Mobile Terminal (RMT) is an MT that cannot communicate directly with the AP on the one hop link but needs a forward link for two hop communications. An MT associated with the AP and located at the edge of the AP coverage area may perform the function of a forwarder. In [38] a possible implementation of the time shared forwarding concept without large modification to the existing

HiperLAN/2 specification is proposed. A HiperLAN/2 MAC sub frame is generated by the FMT to communicate with the RMTs associated with it. The uplink phase capacity assigned to the FMT in the HiperLAN/2 MAC frame, see Figure 2.1, is exploited by the FMT to define the sub frame and to transmit its own uplink traffic if any. The structure of the sub frame is identical to that of the MAC frame. The limitation of this solution is that the MAC sub frame causes too much protocol overhead. An increase of the number of FMTs degrades the traffic performance substantially.

2.2 IEEE 802.11a

IEEE 802.11a is the physical layer specification of IEEE 802.11 at 5 GHz [60]. In this thesis IEEE 802.11a refers to the IEEE 802.11 WLAN working at 5 GHz. In contrast to HiperLAN/2, IEEE 802.11 uses a distributed access scheme that consists of DCF and PCF [61]. In the IEEE 802.11, DCF (Distributed Coordination Function) is the fundamental mechanism of the medium access control (MAC). DCF is based on Carrier Sense Multiple Access with Collision Avoidance (CSMA/CA) and shall be implemented in all stations. DCF can work without any AP and can realize multihop communication. PCF (Point Coordination Function) is an optional access method that is implemented on top of the DCF. PCF is a mechanism used in IEEE 802.11 WLAN to achieve a contention free access to support time-bounded service by a point coordination station. A high speed physical layer in the 5 GHz band, called IEEE 802.11a, has been specified as a supplement to IEEE 802.11. The data rates of IEEE 802.11a will be up to 54 Mb/s.

2.2.1 DCF

DCF has two mechanisms to access the medium, namely the basic access mechanism and the RTS/CTS mechanism. Both mechanisms use an exponential backoff scheme and an immediate transmission of a positive acknowledgment (ACK) by the receive station.

The reason of using the random backoff is to reduce the collisions owing to the transmissions by more than one station at the same time. This is the collision avoidance (CA) feature of the protocol. According to the

DCF (see Figure 2.2), a station must sense the medium before initiating the transmission of a packet. If the channel is idle for a period of time equal to a distributed inter frame space (DIFS), the station transmits. Otherwise, if the channel is sensed busy (either immediately or during the DIFS), the transmission is deferred and the backoff process is started. The station generates a random backoff interval before transmitting. It is based on the idea that after the channel was busy for some time, the probability of more than one station that wants to access the channel will be high. So the risk of collision will be high. Through the back off process, most collisions resulting from the simultaneous access can be avoided.

DCF employs a discrete-time backoff scale. The time immediately following an idle DIFS is slotted, and a station is allowed to transmit only at the beginning of each time slot. The slot time, τ, is set equal to the time needed at any station to detect the transmission of a packet from any other station. It accounts for the propagation delay, Rx_TX_Turnaround_Time and medium detection time. The station computes a random number uniformly distributed between zero and a maximum (w-1). w is called Contention Window (CW) and depends on the number of transmissions failed for the packet. At the first transmission attempt, w is set equal to a value CW_{min}. After each unsuccessful transmission, w is doubled, up to a maximum value $CW_{max} = 2^m CW_{min}$. After a successful transmission, w is reset to CW_{min}. The random number is multiplied by the slot time τ, resulting in the backoff interval used to set the backoff timer. This timer is decremented only when the medium is idle, whereas it is frozen when another station is transmitting. Each time the medium becomes idle, the station waits for a DIFS and then periodically decrements the backoff timer. The station transmits when the backoff timer expires.

Since a sending station cannot detect any ongoing collision in a wireless environment, an explicit transmission of an ACK by the receive station is used to notify the sending station that the transmitted frame has been successfully received. The ACK is transmitted at a time interval equal to the SIFS (short inter frame space) after the end of the reception of the previous one. If the sending station does not receive the ACK within a specified ACK_Timeout, it reschedules the frame transmission according to the given backoff rules.

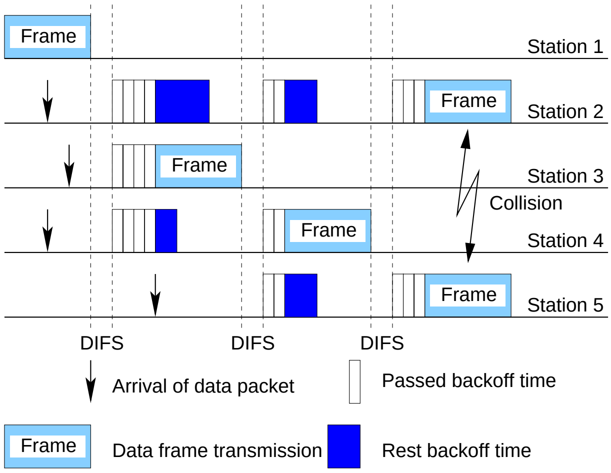


Figure 2.2: IEEE 802.11 basic access mode

As an example, Figure 2.2 illustrates the operation of the basic access mechanism. Five stations share the same radio medium. We assume that a packet arrives at station 2, station 3 and station 4 during the transmission period of station 1. As these stations sense the medium as being busy, they defer the transmission and execute a backoff process. Their backoff timers are decremented after the medium is idle for a time interval greater than DIFS. Station 3 transmits its packet earlier than other two stations as its backoff timer expires earlier. The timers of Station 2 and Station 4 are frozen during the transmission period of station 3 until the medium is idle for a time interval greater than DIFS again. The transmissions of Station 2 and Station 5 collide as their timers expire at the same time.

The limitation of CSMA/CA in comparison with CSMA/CD in IEEE 802.3 is that an unsuccessful transmission cannot be detected at the beginning. A sending station has to wait the ACK from the receiver to know whether its packet has been successfully received. If the packet cannot be successfully received, the channel resources are wasted. The larger the packet of an unsuccessful transmission, the more the channel resources wasted. There are two reasons that the packet cannot be received. (1) the receiver is out or the decode range of the transmit station. (2) The receive station is interfered.

To reduce the waste of bandwidth due to the unsuccessful transmission of large packets, DCF defines an additional four way handshaking technique. With the RTS/CTS mechanism, a station that wants to transmit a data frame, waits until the channel is sensed idle for a DIFS, follows the backoff rules explained above, and then, instead of a data frame, transmits a special short frame called Request to Send (RTS). When the destination station can receive the RTS packet, it responds, after an SIFS, with a Clear to Send (CTS) packet. The source station is allowed to transmit its data frame after an SIFS only if the CTS packet has been correctly received.

The RTS/CTS packet carry the information of the length of the data frame to be transmitted. Other stations that receive this information are able to update a network allocation vector (NAV) containing the information of the period of time in which the medium will remain busy.

2.2.2 PCF

As DCF has no means to support a time-bounded service, IEEE 802.11 defines PCF to permit a Point Coordinator (PC) to have priority access to the medium. Stations that are able to respond to polls by the PC are called Contention Free Pollable (CF-Pollable). Besides the AP, only CF-Pollables are able to transmit frames according to the PCF. PCF has no multihop ability.

2.3 W-CHAMB

The aim of this thesis is to develop a new air interface to form a W-CHAMB (Wireless CHannel-oriented Ad-hoc Multihop Broadband) network that is suited for the next generation wireless multimedia communications. The advanced features of W-CHAMB are self-organization, multihop transmission and QoS guarantee [127].

Figure 2.3 shows the W-CHAMB network architecture. Wireless Stations (WSs), e.g. Pocket Computers or Laptops, can communicate with each

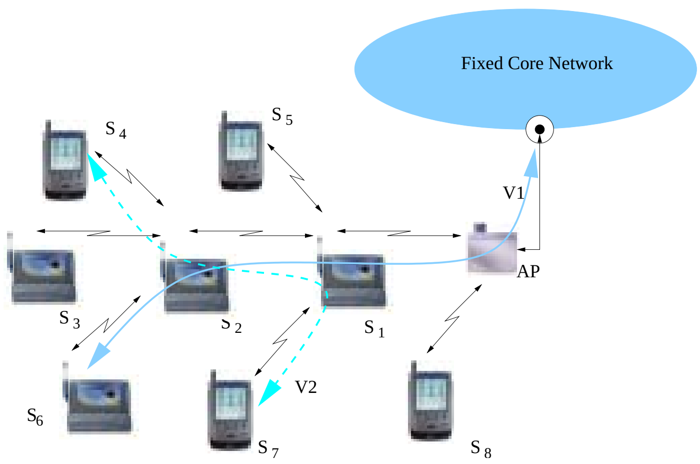


Figure 2.3: W-CHAMB network architecture

other directly or through one or more WSs acting as relays to reach their destinations. If an access point (AP) to a fixed core network is available, the W-CHAMB network WSs can be integrated with the fixed core network. With a number of APs, a large communication zone can be achieved to provide a public/private radio access network with a full coverage of an area of any size.

The most challenging task to realize such a self-organizing multihop network with QoS guarantee is to design an efficient MAC protocol. In addition to general requirements of a MAC protocol, such as simplicity, efficiency and reliability, the design challenges of a MAC protocol for a self-organizing broadband wireless network are provisioning different QoS requirements to various services and realizing statistical multiplexing of bursty traffic in a self-organizing multihop networking environment.

2.3.1 QoS Provisioning

Most MAC schemes rely on a central control to allocate radio resources to meet the QoS requirements for various services [30, 59, 65, 94, 99]. These kinds of MAC schemes can function well if all the WSs can successfully receive the control information from the central control and the central control can receive the resource requirements from the WSs. In the broadband wireless networking environment at the 5 GHz unlicensed frequency spectrum with hidden station and shadowing problems, such kind of assumption is too optimistic. Without a central control, provisioning QoS to different services must be realized in a fully distributed manner. A distributed algorithm is necessary to give urgent data packets higher priority to access the common wireless channel.

2.3.2 Statistical Multiplexing

As the broadband multimedia traffic is bursty in nature, statistical multiplexing among these traffic streams must be considered to use radio resources efficiently [51, 58, 63, 75, 82]. This requires a WS to release its channel reservation at the end of an information burst and to reserve the resource again after the arrival of the next information burst. A WS needs to contend with other WSs to reserve the channel resources dynamically.

2.3.3 Asymmetric Traffic

In the future broadband wireless networks, asymmetric traffic will dominate in the traffic flows, mainly from IP applications. It will be very beneficial to separate the two directions of a bidirectional communication connection. This feature makes it impossible to piggyback an ACK (acknowledgment) in the data packet for the purpose of the ARQ (automatic repeat request) protocol. A new method needs to be developed to acknowledge a received data packet.

2.3.4 Synchronization

It is much more difficult to maintain synchronization in a self-organizing network than in a centralized one hop network. As the W-CHAMB MAC protocol is based on a frame structure, the synchronization issue needs to be solved. Three approaches may be used to maintain synchronization in a self-organizing network. In the IEEE 802.11 standard [61], all concerned stations are synchronized to a common clock using a timing synchronization function. The timing synchronization function in an ad hoc (self-organizing) network is implemented via a distributed algorithm that is performed by all concerned stations. All stations shall transmit Beacons according to an algorithm specified in [61]. In [89], another synchronization scheme to maintain a global clock is proposed. This scheme is adopted in [5]. The synchronization in a self-organizing network can also be realized by an external radio clock which has a large coverage [133]. Further experimental study is necessary to decide which synchronization scheme is best suited for W-CHAMB networks.

2.3.5 Shadowing Problem

Broadband wireless networks operating at 5 GHz frequencies have severe shadowing problems. The multihop transmission capability is needed to solve the shadowing problem to achieve a reasonable communication coverage.

2.3.6 Hidden Station Problem

The most challenging problem in designing a MAC protocol for a self-organizing broadband wireless network is the existence of hidden stations. The hidden station problem is addressed in Section 3.3 in details.

CHAPTER 3

Design Concept of Self-organizing Wireless Broadband Networks

In this Chapter the design concept based on the ISO/OSI reference model is presented. There are a number of common issues involved in the design of self-organizing wireless broadband networks. These include efficient methods for sharing the common radio channel, methods for determining connectivity and using that connectivity to route data through the network, methods for achieving reliable communications in a noisy radio environment, and methods for managing and controlling the distributed network. The function to be performed may be organized into a linear hierarchical structure, as defined in the ISO/OSI reference model. This allows the discussion of design issues underlying a packet communication to be done by focusing on one layer at a time.

3.1 ISO/OSI Reference Model

The International Standardization Organization (ISO) specified a generally accepted model, the ISO/OSI reference model, for Open Systems Interconnection (OSI). The model refers to almost all the communications systems in use today [67]. The ISO/OSI reference model defines the hierarchical layers of the entire communication process. Each layer except the top one offers services to the layer directly above it. The hierarchical model facilitates the protocol development. If a change is undertaken in one of the layers, it does not affect the others. Furthermore, the structure of the layers makes it easier for protocols to be implemented and standardized.

The ISO/OSI reference model is based on different principles. Each layer carries out a precisely defined function, and each function has been stipulated in line with internationally standardized protocols. The boundary lines between the individual layers have been established to minimize the

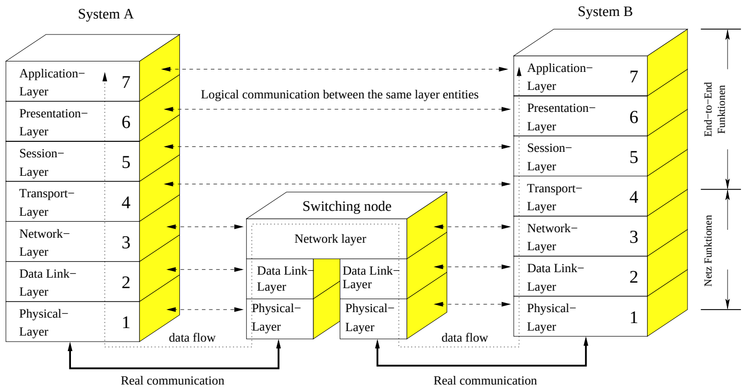


Figure 3.1: ISO/OSI reference model

data flow over the interfaces. Each higher layer represents a new level of abstraction from the layers below it. To keep the number of layers and interfaces to a minimum, several different functions have also been added to the same layer. A seven layer model (see Figure 3.1) was created as a result of these considerations.

A brief description of the different tasks of the seven layers of the ISO/OSI reference model is the following [115]:

Physical layer: The bit transmission layer (physical layer) provides the basis for communication and facilitates the transmission of bits over a communications medium. Layer 1 describes the electrical and mechanical characteristics, e.g., standardized plugs, synchronized transmission over cable or radio channels, synchronizing techniques, signal coding, and signal levels for the interface between terminal equipment and line termination.

Data link layer: The task of the data link layer is to interpret the bit stream of layer 1 as a sequence of data blocks and to forward them error free to the network layer. Error-detection or correction codes are used to protect data from transmission errors. Thus, for example, systematic redundancy that is used at the receiving side for error detection is added by the transmitter to the data, which is transmitted in blocks (frames).

These frames are transmitted sequentially between peer entities of layer 2. If a transmission error is detected then an acknowledgment mechanism initiates a retransmission of the block and guarantees that the sequence will be maintained.

The data link layer adds special bit patterns to the start and to the end of blocks to ensure they are recognized. Because of flow control on both sides, the logical channel can be used individually by the communicating partner entities.

Layer 2 contains the access protocol for the medium and the functions for call set-up and termination with regard to the operated link.

Network layer: The network layer is responsible for the setting up, operation and termination of network connections between open systems. In particular, this includes routing and address interpretation, and optimal path selection when a connection is established or during a connection.

Layer 3 also has the task of multiplexing connections onto the channels of the individual subnets between the network nodes.

Transport layer: The transport layer has responsibility for end-to-end data transport. It controls the beginning and the end of a data communication, carries out the segmentation and reassembly of messages, and controls data flow. Error handling and data security, coordination between logical and physical equipment addresses and optimization of information transport paths are also within the range of this layer's tasks.

The transport layer represents the connecting link between the network-dependent layers 1-3 and the totally network-independent overlaid layers 5-7, and provides the higher layers with a network-independent interface. The transport layer provides a service with a given quality to the communicating applications processes, regardless of the type of network used.

Session layer: The session layer controls communication between participating terminals, and contains functions for exchange of terminal identification, establishing the form of data exchange, dialogue management, tariff accounting and notification, resetting to an initialized logical checkpoint after dialogue errors have occurred, and dialogue synchronization.

Presentation layer: The presentation layer offers services to the application layer that transform data structures into a standard format for transmission agreed upon and recognized by all partners.

It also provides services such as data compression as well as encryption to increase the confidentiality and authenticity of data.

Application layer: The application layer forms the interface to the user or an applications process needing communications support. It contains standard services for supporting data transmission between user processes (e.g. file transfer), providing distributed database access, allowing a process to be run on different computers, and controlling and managing distributed systems.

3.2 Design Concept Based on the ISO/OSI Reference Model

The ISO/OSI seven-layer architecture model is a widely accepted, useful framework for network design and provides a natural basis for description of the design concept of the self-organizing network [76]. The network consists of mobile nodes that are geographically dispersed over distances that may be beyond line of sight. In particular, a node has no a priori knowledge of the location of other nodes, the connectivities of the network, nor even of its own neighbors. Thus the first task of the network design is to provide the means by which the existing connectivities among the nodes can be discovered. This is a basic issue that resides in the physical layer of the ISO/OSI reference model.

Once this is achieved, the next step is to determine how to transform the discovered connectivities into reliable communication links. This is the issue that resides in the data link layer of the ISO/OSI reference model. Actually, in a wireless network in which the communication medium is a multiple access channel the situation is somewhat complicated by the potential interference between simultaneously transmitted signals. Thus, new issues arise, such as when and how to activate or enable the links determined by the discovered connectivities.

Assuming the issue of link utilization gets resolved, the next natural question is how to route messages through the network whose destinations are more than one hop away, which is clearly a network layer issue.

3.2.1 The Physical Layer

Self-organizing broadband wireless network operating at the 5 GHz license-exempt frequency spectrum shall use OFDM (Orthogonal Frequency Division Multiplexing) schemes as its physical layer to increase the robustness against frequency selective fading or narrow-band interference [17, 27, 34, 60, 110]. In a single carrier system, a single fade or interference can cause the entire link to fail, but in a multi-carrier system, only a small percentage of the sub-carriers will be affected. Error correction coding can then be used to correct for the few erroneous sub-carriers. OFDM is an efficient way to deal with multi-path fading. For a given spread, the implementation complexity is significantly lower than that of a single carrier system with an equalizer. In addition, using the DFT-based multi-carrier technique, frequency-division multiplex is achieved not by bandpass filtering but by baseband processing.

The word orthogonal indicates that there is a precise mathematical relationship between the frequencies of the carriers in the system. In a normal frequency-division multiplex system, many carriers are spaced apart in such a way that the signals can be received using conventional filters and demodulators. In such receivers, guard bands are introduced between the different carriers and in the frequency domain which results in a reduction of spectrum efficiency. In an OFDM signal, the sidebands of the individual carriers overlap. To receive the signals without adjacent carrier interference the carriers must be mathematically orthogonal.

The basic principle of OFDM is to split a high-rate data stream into a number of lower rate streams that are transmitted simultaneously over a number of sub-carriers. Because the symbol duration increases for the lower rate parallel sub-carriers, the relative amount of dispersion in time caused by multi-path delay spread is decreased. Intersymbol interference is eliminated almost completely by introducing a guard time in every OFDM symbol. In the guard time, the OFDM symbol is cyclically extended to avoid inter carrier interference.

The physical layer using OFDM schemes establishes digital link connectivity among nodes in the network so that information paths may be established from traffic sources to their destinations.

After the nodes in a self-organizing network discovered each other, an overall network organization structure that can be flexibly used to accommodate the overall network operation is needed. Link connectivity can be determined using a distributed algorithm. Distributed algorithms are of importance for the survivability, robustness, and freedom from reliance on any single node for network control. As links are discovered, a structure should be formed that will permit:

1. communication between any pair of nodes;
2. avoidance of the hidden station problem;
3. robust recovery from node or link losses and other topological changes.

After the existing connectivities, each of which represents a potential communication link, are identified, a highly nontrivial question arises, namely: How should these links be used? How should a node transmit over any of these links, given the distributed nature of the network and the potential of interference? These questions are issues of the second layer of the ISO/OSI architecture.

3.2.2 Data Link Control

The communication resource available is the channel. This resource must be shared by, and therefore allocated to the users of the network. This issue is addressed in the Medium Access Control (MAC) sublayer of the Data Link Control (DLC) layer. The traditional way of allocating bandwidth is via frequency division multiple access (FDMA), under which each user is provided a different portion of the frequency band for its exclusive use. The time-domain counterpart of FDMA is time division multiple access (TDMA) under which each user has exclusive rights for the use of the entire channel during predetermined, periodically recurring time slots [14, 113]. In a multihop network it is also possible to reuse the communication resource in a distant part of the network.

The performance of self-organizing networks depends on the way of the use of the scarce channel resources. The key to an effective resource-sharing scheme is the ability to share or reuse the resource as efficiently as possible.

There are three main domains in which we can achieve channel sharing: frequency, time and space.

Time reuse and its management is usually referred to as the channel access protocol. We can classify channel access protocols in terms of how much knowledge about the system state is used in making decisions. The simplest protocols (e.g. ALOHA [2]) require no system state information to operate; more complex protocols (e.g., CSMA [70]), which tend to have better performance, typically require additional state knowledge (such as the state of the neighbors, obtained by the listening to the channel for CSMA). The key point in spatial reuse is that when a node is transmitting in some part of the network, it is possible to reuse the same frequency and time in another part of the network with no significant interference.

To allocate the bandwidth in time and space to the nodes in the network, four techniques are available, all of which may coexist: frequency division, time division, code division, and spatial re-utilization of the bandwidth resources.

In self-organizing broadband wireless networks, multiple access control takes a very important role. The purpose of the MAC is to allocate the limited radio channel resources among the wireless stations (nodes).

Conceptually, the multiple access in a self-organizing network can be viewed as a distributed queuing system, see Section 6.1.2.1. Each wireless station (WS) has a number of queues of packets to be transmitted. Packets that have the same destination and service type are put in the same queue. The common server of this distributed queuing system is the access channel. Ideally, the server should view all the waiting packets as one combined queue to be served by an appropriate queuing discipline so that the most urgent real time packets will be served at first and the channel resources can be shared efficiently and fairly among the wireless stations in the network. Unfortunately, the server does not know which wireless station has packets to send; Similarly, wireless stations are unaware of packets at other wireless stations. Thus, the knowledge about the state of the queue is fully distributed.

The simplest way for a WS to send packets is that each WS, upon receiving a new packet, transmits it immediately. If a packet is involved in a collision, it is retransmitted after a random delay. This scheme is known as pure ALOHA. The pure ALOHA and its improvement called slotted

ALOHA were widely used in wireless computer network in the 70s. The difference between ALOHA and S-ALOHA is that WSs in S-ALOHA system is synchronized on a slot basis and send a packet only at the beginning of a time slot whose duration is exactly equal to the transmission time of a single packet. The advantage of ALOHA and S-ALOHA systems is its simplicity. It worked well in the early wireless networks, which had a small amount of data packets to transmit. Its disadvantages are low throughput and instability under high load. The maximum throughput of S-ALOHA is 36%, which is double as much as that of an ALOHA system. The reason is that in the S-ALOHA system when two packets conflict, they will overlap completely rather than partially, providing an increase in channel efficiency.

To improve the network throughput, CSMA or listening before talking, was developed. In this scheme a WS attempts to avoid collisions by listening to the carrier indicating another user's transmission. For a system with small propagation and detection delay, CSMA achieves much higher throughput than an ALOHA system. But CSMA has the key disadvantage that its performance degrades greatly if there exist hidden stations [107].

ALOHA, S-ALOHA and CSMA are three basic schemes for the multiple access in the early packet radio networks. The basic ideas of these scheme are still very important in the design of MAC for the modern wireless networks. However, these basic schemes can no longer be directly used for self-organizing wireless broadband multihop networks as the requirements of such networks have been changed significantly. To design a MAC protocol for self-organizing broadband wireless networks, we face many challenges as described in Section 2.3.

The other part of the DLC is the Logical Link Control (LLC) sublayer, which is mainly responsible for the error control, such as Automatic Repeat Request (ARQ) used in W-CHAMB. The general concept of ARQ is to detect PDUs with error at the LLC entity and then to request the source LLC entity to repeat the erroneous PDUs. Three basic ARQ schemes are stop-and-wait, go-back-N and selective-repeat [6, 42, 43, 73, 125].

The stop-and-wait ARQ scheme is the simplest type of retransmission protocol. The basic idea is to ensure that each PDU has been received correctly before initiating transmission of the next PDU. The source LLC entity sends a PDU to the destination LLC entity and waits for an acknowledgment. If a positive acknowledgment (ACK) is received, the source LLC entity sends out the next PDU; if a negative acknowledgment (NAK) is received, the PDU is retransmitted. The performance of the stop-and-wait ARQ is strongly dependent on the delay of the reception of the associated ACK/NAK message.

The go-back-N ARQ scheme is the most widely used type of ARQ protocol in narrow-band wired network. In contrast to the stop-and-wait ARQ scheme, several successive PDUs can be sent without waiting for the ACK/NAK. The destination LLC entity accepts PDUs only in the correct order and sends the request number (RN) of the next expected packet back to the source LLC entity. The effect of a given RN is to acknowledge all PDUs prior to RN and to request transmission of the next PDU. The go back number $N >= 1$ in a go-back-N protocol is a parameter that determines how many successive PDUs can be sent in the absence of a request for a new PDU. The go-back-N protocol has a kind of inherent inefficiency as all PDUs following an erroneous PDU are discarded, regardless of whether or not they are error-free.

The selective-repeat (SR-ARQ) is the most efficient ARQ protocol. The basic idea of SR-ARQ is to accept out-of-order PDUs and to request retransmissions only for those PDUs that are not correctly received.

The fast ARQ scheme developed for W-CHAMB networks is as efficient as SR-ARQ and as simple as stop-and-wait ARQ, see Section 4.4.4. The design of the W-CHAMB DLC protocol is presented in Chapter 4 in detail.

3.2.3 Network Layer

A self-organizing network features fully automated network management. It is self-configuring upon network initialization, reconfigures upon gain or loss of WSs, and has dynamic routing. The network operation simply requires that a WS has at least one other WS within its communication range. The system discovers the radio connectivity between WSs and organizes routing strategies dynamically on the basis of this connectivity. After initialization, communication networks maintain a static topology in most cases. A unique feature is that the network topology can be altered without affecting the user's ability to communicate. Although the connectivity may change in an unexpected way as WSs move about, the automated network management procedures used are capable of sensing the existing connectivity in real time and then exploiting this connectivity in order to continuously transport data and control packets.

After a WS has been powered on, it begins the process of establishing and maintaining local connectivity. As long as the WS is powered up, it announces its existence and information about the network topology from its perspective periodically. Each WS gathers and maintains information about network topology so that it can make independent decisions about how to route data through the network to any destination. The best route is the shortest route with good connectivity on each hop.

Routing is an important issue for multihop networks. Before a connection can be established, a route between the source and destination must be found. The existing routing schemes designed for multihop wireless networks can be classified either as proactive or reactive [53]. Proactive schemes attempt to continuously evaluate the routes within the network so that the route will be known and can be immediately used when it is needed. The family of Distance-Vector schemes [92] is an example of a proactive scheme. Reactive schemes, on the other hand, invoke a route determination procedure on demand only. The family of classical flooding algorithms is a reactive scheme.

The advantage of proactive schemes is that, once a route is queried, there is little delay until a route is determined. In reactive schemes, route information may not be available at the time a route is needed. The delay to determine a route can be quite significant. Moreover, the global search procedure of reactive protocols requires significant control traffic. Because of long delay and excessive control traffic, pure reactive routing schemes may not be applicable to real-time communication. However, pure proactive schemes are likewise not appropriate for the self-organizing networks, as they continuously use a large portion of network capacity to keep the routing information up-to-date.

A trade-off between the proactive scheme and the reactive scheme results in many hybrid proactive/reactive routing schemes. Zone Routing Protocol (ZRP) [53] proposed for ad hoc networks is one example protocol. ZRP limits the scope of the proactive procedure only to the WS's local neighborhood. If the destination is out of the local neighborhood of the source WS, the search is done globally by querying the selected WSs in the network.

In ZRP, a WS's local neighborhood is defined as a routing zone. A routing zone of radius n includes the WSs whose minimum distance in hops from the WS in question is at most n hops. Each node is assumed to maintain routing information only to those WSs that are within its routing zone. The topology update is performed locally. The amount of update traffic required to maintain a routing zone is not dependent on the total number of WSs. A WS learns its zone through a proactive scheme, which is called IntrAzone Routing Protocol (IARP). The IARP maintains routes only for those WSs that are within the coverage of the routing zone. For reactively discovering routes to destinations located beyond a WS's routing zone, The IntErzone Routing protocol is developed. The IERP is distinguished from standard flooding-based query/response protocols by exploiting the structure of the routing zone. The idea behind the ZRP is that querying can be performed more efficiently by transporting queries to the periphery of a routing zone, rather than flooding the queries. However, problems arise once the query leaves the initial routing zone. Because the routing zone heavily overlap, a WS can be a member of many routing zones. It is very probable that the query will be forwarded to all the WSs. As the query is sent along a path equal to the zone radius n, $n >= 1$, the IERP will result in much more traffic than flooding itself! Figure 3.2 shows an example of the ZRP protocol. The radius of the routing zone is two hops. Assume that the source station S needs a route to destination D. As D is out of S's routing zone, IERP is invoked. S sends a route query to the selected WSs, *i.e.* F1, F2, F3 and F4, that are at the boundary of its routing zone. Although D is found in the routing zone of F1, ZRP cannot terminate the route query and F2, F3 and F4 delivery the route query to their border WSs, F5 - F13 further. Such a global search throughout the whole network generates too much control traffic. So ZRP is not suited for W-CHAMB networks.

The design objective of W-CHAMB networks is that more than 80% of the connections shall have fewer than or equal to 3 hops. Connections with 4 hops should be less than 20 %. Deploying the network architecture of W-CHAMB networks, an iterative routing protocol (IRP) called W-CHAMB IRP is developed, see Section 4.5.1.

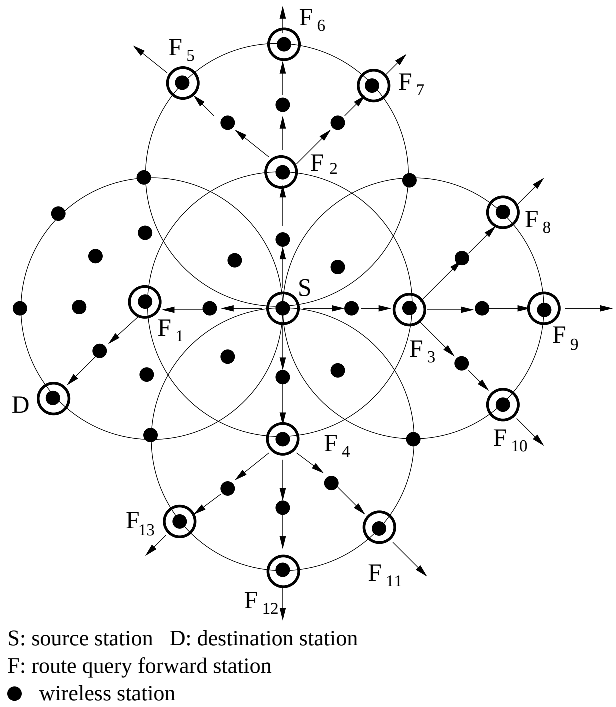


Figure 3.2: An example of the ZRP protocol

3.3 The Hidden Station Problem

Hidden stations may result in extreme inefficiency in self-organizing wireless networks. A hidden station is a station that cannot sense the transmission of the sending WS, but will cause interference to the receiving WS if it transmits. Hidden stations can be caused by an obstructer, see, Fig. 3.3. Assuming that S_2 is receiving data from S_1, S_3 cannot sense the transmission of S_1 because of the obstructer. If S_3 transmits at the same slot as used by S_1, S_2 will be interfered. S_3 is a hidden station in this case. Hidden stations may be caused by the multihop environment. Even if there is no obstructer, hidden stations can result from the different distances among wireless stations. Assuming that S_2 is receiving from S_1 again. As S_5 is out of the detection range of S_1, it cannot sense the transmission of S_1. So S_5 is a hidden station which may cause interferences to S_2. Hidden stations may degrade the network performance substantially [107]. There are two basic approaches to solve the hidden station problem:

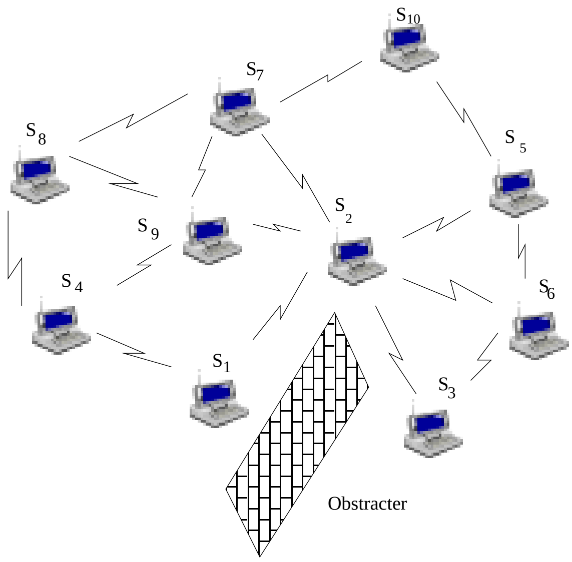


Figure 3.3: Self-organizing broadband wireless network

1. The busy tone solution was firstly proposed in [107] to combat hidden stations in CSMA systems. A busy tone signal is sent by the receiving station on a narrow band channel to make a hidden station aware of an ongoing transmission and prevent it from transmitting and interfering. The limitation of the busy tone solution is the need of a separate channel, the need of additional hardware for the receiver to transmit the busy tone while receiving on the data channel, and the large delay of detecting the busy tone in a narrow band channel.

2. The RTS/CTS (request to send/clear to send) mechanism as specified in IEEE 802.11 is able to combat the hidden station problem to some extent. A station that intends to send a data packet sends an RTS packet to the receiver. After reception, in turn, the receiver sends a

CTS packet to indicate it is ready to receive data. Other stations that receive the RTS and/or CTS packet will defer their access for a period of time according to the transmission duration information contained in the RTS/CTS packets. The goal of the RTS/CTS mechanism is that hidden stations should receive the CTS packet and would cooperate then. But the RTS/CTS mechanism does not solve the hidden station problem completely. Some cases remain where hidden stations cannot receive the CTS packet. For example, see Fig. 3.3, S_{10} cannot receive the CTS packet from S_2 because it is out of the decode range of S_2. But S_{10} can cause interference to S_2 as S_{10} is still in the interference range of S_2. The interference range is usually much larger than the decode range. In some cases, even though a station is in the decode range of the receiver, it may not be able to receive the CTS packet because of the interference of other stations. For example, S_9 is in the decode range of S_2. But if S_8 is sending while S_2 transmits the CTS packet, S_9 cannot receive the CTS of the S_2. So S_9 may access the channel after S_8 ended the transmission, which causes interference to S_2.

The hidden station problem is solved using energy signals in W-CHAMB networks.

3.4 Review of the Existing MAC Schemes

3.4.1 HiperLAN/1 EY-NPMA

EY-NPMA (Elimination Yield Non Preemptive Priority Multiple Access) is specified as the channel access procedure for the HiperLAN/1 standard [40]. HiperLAN/1 aims to support fully distributed ad hoc networking, multihop transmission and multimedia applications with real time requirements. Up to five frequency channels in the 5.15-5.30 GHz range are being provided for HiperLAN/1. One channel provides a bit rate of 23.5249 Mbit/s. It seems that HiperLAN/1 has the required features of self-organizing broadband networks. So it is interesting to discuss the EY-NPMA multiple access protocol to find the reason why HiperLAN/1 has not been succeeded at the market.

The EY-NPMA channel access mechanism defines three activity phases: the prioritization phase, the contention phase, and the transmission phase. A combination of an elimination phase and a yield phase is used for the contention phase. The priority phase is 1 to m_{CAP} slots long. If a packet to be transmitted has channel access priority n, the station detects the channel for the duration of n-1 slots (priority detection). If the radio channel is free at this time, the station immediately sends a burst (priority assertion, PA). However, if a burst has been sent by another station before the n-1 prioritization slots have been finished, this station ends the access cycle immediately and waits for the next cycle. The priority phase ends with the transmitted burst PA. We can see that the priority phase can support m_{CAP} kinds of priorities. The smaller priority number means higher priority.

The priority phase is followed by a contention phase during which a selection of stations takes place if a number of stations want to transmit data packets with the same priority. The decision is made about which stations are allowed to transmit their packets over the radio channel. the contention phase consists of an elimination and a yield phase. Details of the operation of these phases can be found in [7, 40, 115].

Although the EY-NPMA mechanism has many interesting features, it suffers many shortcomings which made the HiperLAN/1 not successful. Since the standardization of HiperLAN/1 in 1996, there is still no product at the market. The main problems of the EY-NPMA are follows:

1. EY-NPMA is very inefficient for the transmission of short packets, such as ATM cells which contain 53 bytes only [55, 56]. A large overhead which results from priority phase, contention phase and acknowledgment is necessary to transmit every data frame.

2. No effective means is available in EY-NPMA to overcome the hidden station problem. The sophisticated priority phase and contention phase cannot guarantee the transmission of a data frame free of collision in the existence of hidden stations that are quite usual in a self-organizing networking environment. The performance of HiperLAN/1 degrades significantly due to the interference from the hidden stations.

3. The support of QoS in HiperLAN/1 is very limited. The reasons for that are as follows:

(a) Access priority becomes invalid in the existence of hidden stations.

(b) No means of call admission control is considered. The network can easily experience an overloaded situation, at which no QoS can be supported.

(c) The channel access cycle is non preemptive. A delay sensitive packet cannot acquire the channel immediately.

One lesson we learn from the EY-NPMA mechanism is that QoS support cannot be achieved with packet-oriented multiple access in a self-organizing network. Even with very sophisticated priority and contention phases that cause the unacceptable protocol overhead for short packets, the packet transmission can still be interfered by the hidden stations.

The idea to use energy signals to realize a distributed access priority in the EY-NPMA mechanism is interesting and can be used in the MAC design for self-organizing broadband networks to achieve a prioritized access. The priority phase can be improved further by a combination of several energy signals, see Section 4.3.2.4.

3.4.2 DCF of IEEE 802.11

As the Distributed Coordination Function (DCF) of IEEE 802.11 can be operated in a self-organizing multihop networks, it is interesting to review its suitability for a self-organizing broadband wireless network.

Due to the worldwide great success of Ethernet, the packet-oriented CSMA protocol is pervasive in the area of LANs. So the WLAN standard IEEE 802.11, viewed as a wireless extension of Ethernet, has also adopted the packet-oriented CSMA/CA (Carrier Sense Multiple Access with Collision Avoidance) access scheme [61]. A high speed physical layer in the 5 GHz Band, called IEEE 802.11a, has been specified as a physical layer supplement to IEEE 802.11. The data rate of IEEE 802.11a is up to 54 Mb/s [60].

Although the DCF of IEEE 802.11 is fully decentralized and self-organizing, it is not, however, suited for self-organizing broadband wireless networks due to its inefficiency and no means to guarantee QoS.

The success of the Ethernet is due to its simplicity. The throughput of the Ethernet with the CSMA/CD protocol can be achieved as high as 82%. The increasing bandwidth requirements have been met by the introduction of the high speed Ethernet of 100 Mb/s and 1 Gb/s. IEEE 802.11 WLAN, however, has a different transmission medium. It is impossible for IEEE 802.11 to apply CSMA/CD because the sending station cannot detect any ongoing collision. So IEEE 802.11 can only use a CSMA/CA protocol. To achieve collision avoidance (CA), a large protocol overhead, such as backoff, is necessary. In addition, the existence of hidden stations in wireless environments makes the CSMA-like protocol very inefficient [107]. To overcome this problem, IEEE 802.11 has specified the RTS/CTS mechanism that increases the protocol overhead. It is worth mentioning that RTS/CTS cannot solve the hidden station problem completely (see Section 3.3). Moreover, one of the main advantages of the packet-oriented CSMA protocol namely that the transmission of large packets can achieve high efficiency, is no longer valid in the wireless environment because the packet error probability will be higher for larger packets. Assume that the packet error rate (PER) of a short packet is 3 %, then the PER of a large packet of 10 short packets length will be $1 - (1 - 0.03)^{10} = 26\%$, which is no longer acceptable. So large packets have to be fragmented to short packets to achieve sufficient transmit reliability. Such a fragmentation increases the protocol overhead.

The limitations of DCF to be used in self-organizing broadband multihop multimedia wireless networks are as follows:

1. The transmission of a data frame in IEEE 802.11 is packet-oriented. An explicit ACK is used to acknowledge the successful reception of the data frame. This results in inefficiency for the transmission of short packets, such as ATM cells.

2. The hidden station problem cannot be solved by the RTS/CTS mechanism completely. In a multihop environment, there exist many cases at which hidden stations cannot receive RTS/CTS packets [35].

3. DCF does not support provisioning of different QoS requirements to different services. It cannot guarantee QoS for real time services.

4. To realize collision avoidance, DCF uses the backoff process. This backoff process increases the protocol overhead. The higher the data rate in the air interface, the more significant this overhead is.

In our research, we have found that provisioning QoS requirements to high performance multimedia applications in a packet-oriented self-organizing wireless network appears to be impossible. The current attempt to extend the DCF to support QoS guarantee [101] will suffer from the same problems as that of HiperLAN/1 described in Section 3.4.1.

3.4.3 ETSI HiperLAN/2

ETSI HiperLAN/2 uses a fully centralized MAC protocol. A central controller assigns the radio resources within the HiperLAN/2 MAC frame. The assignment of resources may change from frame to frame. The fixed length HiperLAN/2 MAC frame ($t_{frame} = 2ms$) consists of three major phases. The broadcast phase is used to transport the control information. The downlink, directlink and uplink phase is used to transport data PDUs between the central controller and the WSs or between WSs. The random access phase is used for the initial access to the network, for handover indication and for requesting radio resources.

Using a centralized solution like HiperLAN/2 MAC protocol for self-organizing broadband multihop multimedia wireless networks may suffer from the following inherent problems:

1. To be self-organizing, most wireless stations should have an in-built central control function. As a central controller in a broadband wireless network needs very high computing capacity, the hardware requirements on the wireless stations increase dramatically.

2. The network will be complicated and vulnerable. The failure or departure of the selected central controller will cause temporary chaos in the whole network.

3. Direct mode and multihop communication cannot be realized efficiently as communication is possible only under a central control.

4. The scarce frequency spectrum cannot be used efficiently. Neighboring central controllers must use different frequencies. Dynamic channel allocation which is inherent in decentralized networks is not easy to perform.

5. A wireless station may not be able to associate because it may not be able to receive the information from the central controller or the central controller cannot hear this wireless station.

3.4.4 DECT

DECT (Digital Enhanced Cordless Telecommunications) is a standard specified by ETSI in 1992 [39]. A DECT network is a micro-cellular digital mobile radio network for high user density and primarily indoor communication.

The transmission capacity of each frequency is divided into 10 ms long periodically recurring frames, each of which has a length corresponding to the duration of 11.520 bits. A frame comprises 24 time slots, which are used as either full slots, double slots or half slots. In a basic connection the first 12 time slots are used for transmitting data from the base station to a mobile station (downlink), whereas the second part of the 24 slots is reserved for the direction from the mobile station to the base station (uplink). TDD (Time-Division Duplexing) mode is used for a duplex connection. If the base station occupies slot k for the down link, slot k+12 is reserved for the uplink [115].

The most advanced feature of DECT is the Dynamic Channel Selection (DCS) by mobiles according to the measured signal level RSSI (Radio Signal Strength Indicator). Each mobile station maintains a channel list with RSSI values. The quietest channel is selected to establish a connection. DCS means that basically the entire frequency spectrum with all channels is available in each cell, and a mobile is able to select a suitable channel for itself. So the system is capable of adjusting independently to changes in traffic load. Furthermore, because of the DCS technique, DECT systems do not require frequency planning.

The limits of DECT solution to be used for broadband self-organizing networks are:

1. no method to realize statistical multiplexing of bursty traffic;
2. no means to provision different QoS requirements to different services;
3. not suited to transport asymmetric traffic.

3.5 W-CHAMB - A Solution for Self-organizing Broadband Networks

Inspired from DECT and GPRS concepts, we developed a channel-oriented solution - W-CHAMB (Wireless CHannel-oriented Ad-hoc Multihop Broadband) network - for a self-organizing broadband wireless network. W-CHAMB adopts the key idea of GPRS, that is statistical multiplexing of bursty traffic through packet reservation, and the most advanced feature of DECT, that is dynamic channel selection according to the measured signal level RSSI. It differs from the GPRS and DECT with its ability to operate in a multihop environment and in a fully distributed manner. The most significant feature of W-CHAMB is that it meets QoS demands for various services and realizes statistical multiplexing of bursty traffic in a fully distributed and efficient manner.

In comparison with HiperLAN/2, the advanced features of the W-CHAMB MAC protocol are its multihop ability and decentralized control. The centralized MAC solution of HiperLAN 2 will reach its limitation soon if users in 5-6 GHz license-exempt frequency spectrum increase. In comparison with IEEE 802.11, W-CHAMB achieves a much higher network efficiency for short to medium size packets. More importantly, the DCF of IEEE 802.11 has no means to support QoS of real time services. W-CHAMB can be viewed as a convergence of HiperLAN 2 and IEEE 802.11. It takes the advantages of IEEE 802.11 and HiperLAN 2, but overcomes their shortcomings.

CHAPTER 4

W-CHAMB Networks

This chapter describes protocols and algorithms of W-CHAMB (Wireless CHannel Oriented Ad-hoc Multihop Broadband) networks. Ideas and Solutions to design a W-CHAMB air interface for self-organizing multihop networks with QoS guarantee are addressed. The W-CHAMB protocol stack based on the ISO/OSI reference model is presented in Section 4.1. The protocol stack for the W-CHAMB air interface includes the physical layer (W-CHAMB PHY layer), the medium access control sublayer (W-CHAMB MAC layer), the logical link control sublayer (W-CHAMB LLC layer) and the network layer (W-CHAMB NL). Section 4.2 defines the W-CHAMB PHY layer. The W-CHAMB PHY layer is based on the same modulation scheme as used in HIPERLAN/2 and IEEE 802.11a. After the definition of the W-CHAMB PHY layer, the W-CHAMB MAC layer is described in Section 4.3. The W-CHAMB MAC protocol uses dynamic channel reservation (DCR) to achieve statistical multiplexing of broadband bursty traffic services in a self-organizing network environment. The prioritized access and the reservation interruption scheme are used to provision QoS to different traffic services. Energy signals are used to acknowledge the correct reception of a PDU (protocol data unit) and to solve the hidden station problem. Based on the data service provided by the MAC function, the W-CHAMB LLC layer presented in Section 4.4 provides a reliable data transport between two neighboring wireless stations. A LLC scheduler is used to plan the bandwidth requirements for a PDU train. A fast ARQ scheme based on the MAC level acknowledgment is developed for a simple and efficient error control. The W-CHAMB NL describing the routing protocol and call admission control is discussed in Section 4.5. Routing protocol is necessary for W-CHAMB networks to establish a multihop connection. By exploiting the W-CHAMB network architecture a new efficient routing protocol called iterative routing protocol (W-CHAMB IRP) is developed. Finally, a distributed measurement-based call admission control algorithm is proposed to guarantee the QoS requirement of real time traffic services.

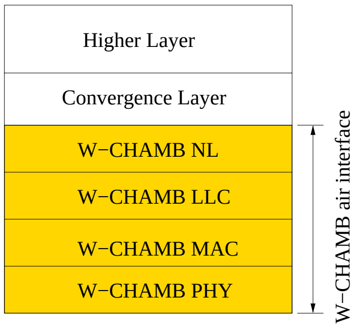


Figure 4.1: W-CHAMB Protocol stack

4.1 W-CHAMB Protocol Stack

Figure 4.1 shows the W-CHAMB protocol stack based on the three lower layers of the ISO/OSI reference model. The W-CHAMB PHY layer provides a basic data transport function for the W-CHAMB MAC layer. The W-CHAMB MAC layer establishes a real channel connection (RCC) for the W-CHAMB LLC layer that takes the responsibility for the reliable data transport of PDU trains coming from higher layers. The W-CHAMB NL performs the routing and the call admission control functions. The convergence layer (CL) is used to perform a segmentation and re-assembly (SAR) function to fit a higher layer data unit into W-CHAMB PDUs and adapts the wired core network to the W-CHAMB air interface on the access point (AP) side. For each supported core network and service a special CL is necessary. CL provides a software interface for W-CHAMB networks to support different kinds of services.

4.2 W-CHAMB PHY Layer

The W-CHMAB PHY layer provides services for the W-CHAMB MAC layer to transport PDUs and energy signals (E-signals). PDUs are sent via

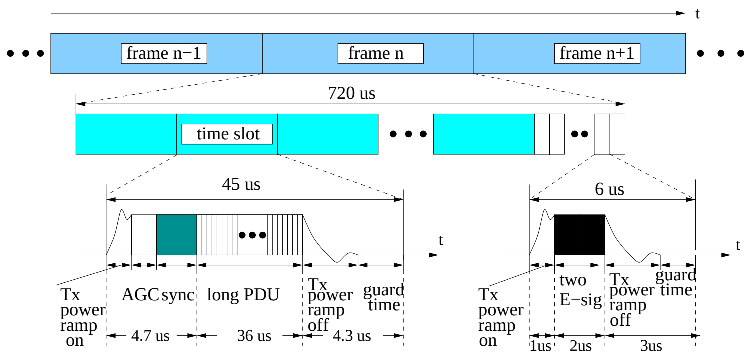


Figure 4.2: W-CHAMB frame structure

broadband channels (BCHs). E-signals are sent via a BCH or a dedicated narrow-band energy signal channel (NECH).

4.2.1 W-CHAMB Frame Structure

For the compatibility and possible coexistence of W-CHAMB with IEEE 802.11a and HiperLAN/2, W-CHAMB uses the same frequency bands and OFDM schemes as those of IEEE 802.11a and HiperLAN/2. The frequency bands are divided into several frequency channels and each frequency channel is shared among many wireless stations (WSs). In W-CHAMB networks, a frequency channel is shared among WSs using TDMA technology [36, 102, 109]. The transmission time scale is organized in frames, each containing a fixed number of time slots. Two time slots may be used to send energy signals. Figure 4.2 shows the W-CHAMB frame structure. WSs of the network are synchronized on a frame and slot basis, which is achieved by a distributed timing synchronization scheme, see Section 2.3.4.

A time slot has a duration of 45 us. The duration of one frame which consists of 16 time slots is 16 x 45 = 720 us. One time slot can be used to send one data PDU or several E-signals together with a short signaling PDU.

4.2.2 Broadband Data Channel

Figure 4.2 shows the structure of the broadband data channel. The necessary physical overhead to send a data packet is 9 us, including necessary time for Tx power ramp on (1.5 us), AGC and synchronization (3.2 us), Tx power ramp off (2.5 us) and guard time (1.8 us) according to [79]. The time left for the transmission of a data PDU in one time slot is 36 us that can be used to transport 9 OFDM symbols.

Data PDUs transmitted per data slot have different data length with different modulation schemes and coding rates, see Table 4.1. The modulation scheme and the coding rate can be selected by a link adaptation algorithm based on the actual radio signal quality, see Section 6.2.5.4. The PDU length is adapted by the convergence layer based on the selected channel type.

Table 4.1: PDU length of different modulation schemes and coding rates

Modulation	Coding rate	Bytes per Symbol	Bytes per Data slot	Data rate ([Mbit/s])
BPSK	1/2	3	27	6
BPSK	3/4	4,5	40.5	9
QPSK	1/2	6	54	12
QPSK	3/4	9	81	18
16QAM	1/2	12	108	24
16QAM	9/16	13.5	121.5	27
16QAM	3/4	18	162	36
64QAM	2/3	24	216	48
64QAM	3/4	27	243	54

4.2.3 Broadband Energy Signal and Short Signaling PDU

E-signals are proposed in W-CHAMB networks for different purposes. Access energy signals (ACC-E-signals) are used to reduce the access collision probability and to realize the prioritized random access. Busy energy signals (Busy-E-signals) are used to solve the hidden station problem. Acknowledgment energy signals (ACK-E-signals) are used to realize the MAC level acknowledgment for the fast ARQ scheme.

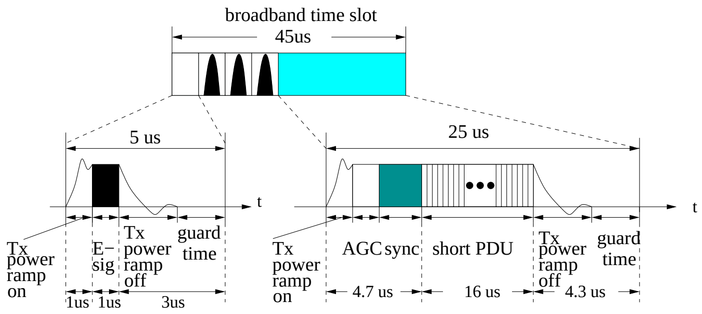


Figure 4.3: Broadband time slot

E-signals can be sent on the BCH. Figure 4.3 shows that four ACC-E-Signals together with a short signaling PDU (*s-PDU*) are sent on one BCH that is based on periodic broadband time slots. One broadband E-signal has a duration of 5 us, including necessary time for Tx power ramp on (1 us), energy signal (1 us), Tx power ramp off (1.5 us) and guard time (1.5 us). The time left for the short *s-PDU* is 25 us, in which a short PDU with 4 OFDM symbols can be sent.

For a small scale network that works at an unsaturated condition, four ACC-E-signals that correspond to a maximum access priority of 15 can achieve the required traffic performance [126]. But they are not enough for a large scale network with a large number of contending WSs to keep a low access collision probability and to support multiple priority types. In this case, one full broadband time slot may be used to support 9 ACC-E-signals. A long *s-PDU* with 9 OFDM symbols can be sent in the following time slot.

An ACK-E-signal is always transmitted together with a Busy-E-signal. The time needed for the transmission of one Busy-E-signal together with one ACK-E-signal is 6 us, see Figure 4.4. If a Busy-E-signal is sent alone, the duration of the energy signal is 1 us and the time to send this Busy-E-signal is 5 us that is the same as needed by an ACC-E-signal. If Busy-E-signal and ACK-E-signal are sent together, the duration of the energy signal is 2 us and the transmission time needed is 6 us. Each time slot except the access channel (ACH) in every MAC frame is associated with a Busy-E-signal as well as an ACK-E-signal. To support these 15 pairs of E-signals, two broadband time slots are needed. The cost of the frequency spectrum

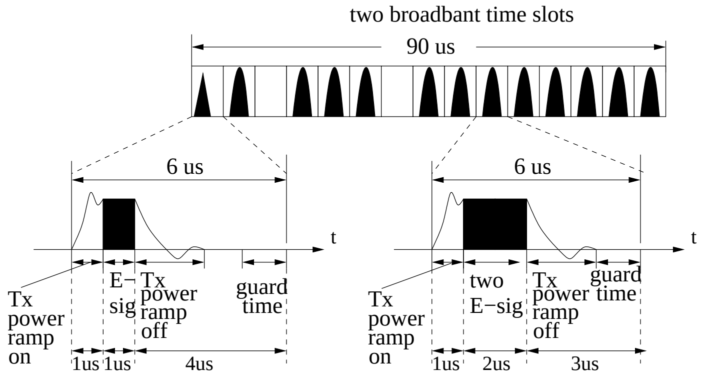


Figure 4.4: Broadband energy channel

to support the Busy-E-signals and the ACK-E-signals is then 12.5% in the case that the number of time slots in every MAC frame is 16.

4.2.4 Narrow-band Energy Signal Channel

To use the frequency spectrum more efficiently, E-Signals can be sent on a narrow band frequency channel, called narrow-band energy signal channel (NECH). Figure 4.5 shows the frame structure of the NECH and the relationship between the NECH and the BCH. The time slots of NECH is exactly one slot delayed in comparison with the BCH so that the received PDU on BCH can be acknowledged on the related NECH. One NECH slot is divided into three subslots, each can be used for one E-signal. The first subslot is used to transmit an ACC-E-signal. 16 ACC-E-signals that can be sent on the NECH are enough to keep the access collision probability as low as necessary. The second and the third subslots are used for Busy-E-signals and ACK-E-signals, respectively. The duration of a subslot is 15 us, in which an E-signal with a duration of 11 us can be sent. About 100 kHz frequency band is required to detect an E-signal with a duration of 11 us. As the frequency band of a broadband channel is 20 MHz, the cost of the frequency spectrum to support the NECH is about $100KHz/20MHz = 0.5\%$. It seems that sending E-signals on NECH is much more efficient than on BCH. The additional hardware required to send/receive a narrow-band en-

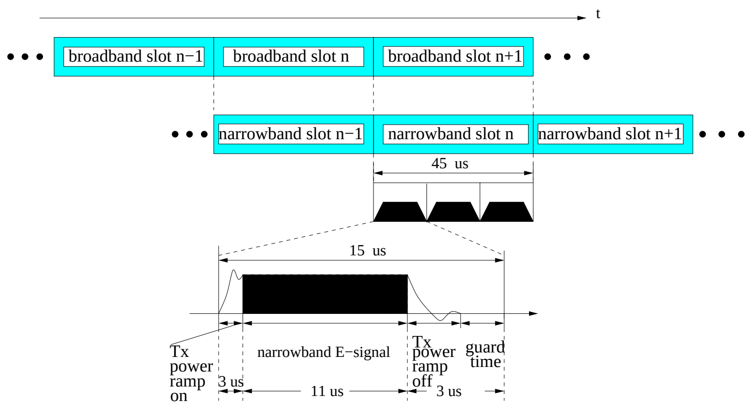


Figure 4.5: Narrow-band energy channel

ergy signal is very limited. Therefore, we prefer to transmit E-signals on the NECH rather than on the BCH.

4.3 W-CHAMB MAC Layer

This section introduces the W-CHAMB MAC protocol, the most important part of the W-CHAMB protocol stack. The traffic performance of a wireless network mainly depends on the MAC protocol and algorithm. Before describing the details of the W-CHAMB MAC protocol, the definition of the MAC data service is presented to build a better understanding of the W-CHAMB MAC operation.

4.3.1 W-CHAMB MAC Data Service

The W-CHAMB MAC data service provides the peer LLC entities with the ability to transport LLC protocol data units (PDUs). To support this service, the source MAC entity establishes a real channel connection (RCC) with a peer MAC entity using the underlying PHY layer services. The established RCC will be released immediately if there is no more LLC PDUs to send in the LLC entity. The successful reception of a PDU is acknowledged by the peer MAC entity. Unacknowledged transmission will be repeated on the next available transmission opportunity.

The W-CHAMB MAC protocol supports the following service primitives for the LLC entity:

- MAC_RCC.request
- MAC_RCC.indication
- MAC_RCC.confirm
- MAC_DATA.request
- MAC_DATA.indication
- MAC_RCC_STATUS.indication

The Message Sequence Chart (MSC) of the W-CHAMB MAC data service is shown in Figure 4.6.

The primitive MAC_RCC.request (*source id, destination id, channel type, maximum access delay, service type, vc id*) used by the LLC entity to request the MAC entity to establish an RCC with a peer MAC entity. It is generated by the LLC entity when an RCC is required by the LLC scheduler. The reception of this primitive causes the MAC entity to establish an RCC. The parameter *source id* specifies the source station identifier. The parameter *destination id* specifies the destination station identifier. The parameter *channel type* specifies the needed RCC bandwidth based on the QoS requirement of LLC PDU trains. The parameter *maximum access delay* specifies the maximum access delay that can be tolerated by the LLC PDU train. A PDU train has to be dropped if an RCC cannot be established before the access delay exceeds its maximum value. The parameter *service type* specifies the service type, e.g. real time service or non real time service, of the LLC PDU trains. The parameter *vc id* specifies the virtual connection (VC) identifier.

The primitive MAC_RCC.indication (*source id, destination id, channel type, rcc id, vc id*) is generated by the destination MAC entity to provide its LLC entity with the information about an incoming RCC. The parameters *source id, destination id, channel type and vc id* has the same meaning as specified in the primitive MAC_RCC.request. The parameter *rcc id* specifies the established RCC identifier.

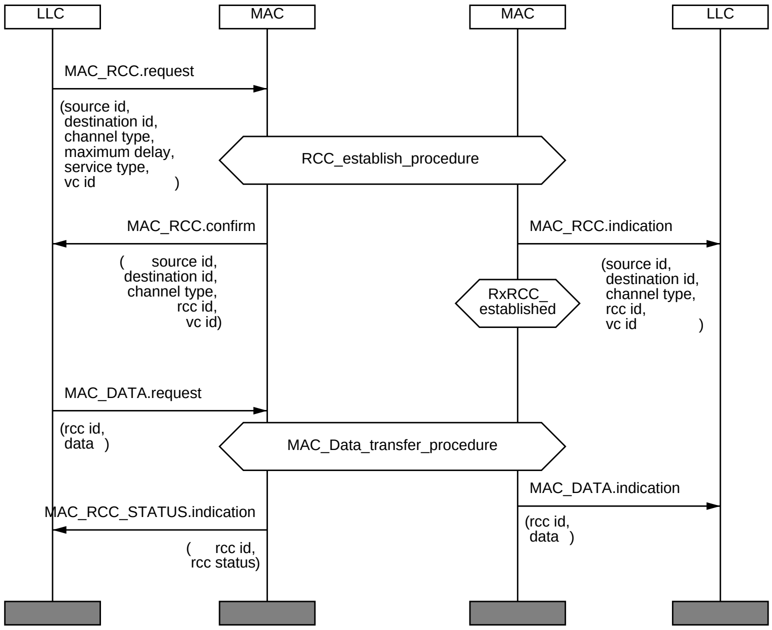


Figure 4.6: W-CHAMB MAC data service

The primitive MAC_RCC.confirm (*source id, destination id, channel type, rcc id, vc id*) is generated by the source MAC entity after the requested RCC is established. After the receipt of this primitive, the source LLC entity transfers data PDUs on the established RCC. The parameters of the primitive MAC_RCC.confirm are the same as specified in the primitive MAC_RCC.indication.

The primitive MAC_DATA.request (*rcc id, vc id, data*) used by the source LLC entity to transport a LLC PDU to the destination LLC entity. The parameter *data* specifies the LLC PDU to be transported.

The primitive MAC_DATA.indication (*rc id, vc id, data*) is passed from the MAC entity to its local LLC entity to deliver a received LLC PDU. The parameter *data* specifies the LLC PDU received by the local MAC entity.

The primitive MAC_RCC_STATUS.indication (*source id, destination id, rc id, rcc status*) is generated upon the receipt of the MAC level acknowledgment or after any change of the RCC. It has a local significance only and provides the LLC entity with the RCC status information. An unsuccessfully transmitted PDU is resent at the next transmission opportunity. The parameters *source id, destination id and rcc id* are the same as those in the primitive MAC_RCC.request. The parameter *rcc status* is used to pass the RCC information to the LLC entity. The RCC status may be one of the following:

- The previous transmission on the RCC is unsuccessful
- The previous transmission on the RCC is successful
- The RCC is released

4.3.2 W-CHAMB MAC Protocol Description

The main task of the W-CHAMB MAC protocol is to provide the LLC entities with the MAC data service by establishing an RCC when needed. As the wireless medium is shared decentrally among all WSs in W-CHAMB networks, the MAC protocol takes the most important role in the network efficiency and reliability.

The W-CHAMB MAC protocol is channel-oriented with many advanced features:

- High efficiency is achieved through statistical multiplexing among bursty traffic services by dynamic channel reservation (DCR).
- QoS for real time services are guaranteed without any central controller.
- The gain of frequency spatial reuse is increased by allowing the interfered station to send and the hidden station to receive.
- The hidden Station problem is solved completely by Busy-E-signals.
- The fast ARQ scheme is realized by MAC level acknowledgments with ACK-E-signals.
- Prioritized random access is realized by ACC-E-signals.

4.3.2.1 W-CHAMB Channel Structure

The transmission of data PDUs in W-CHAMB networks is channel-oriented. Figure 4.7 shows the MAC logical channel structure based on the frame structure of the physical layer, see Figure 4.3.

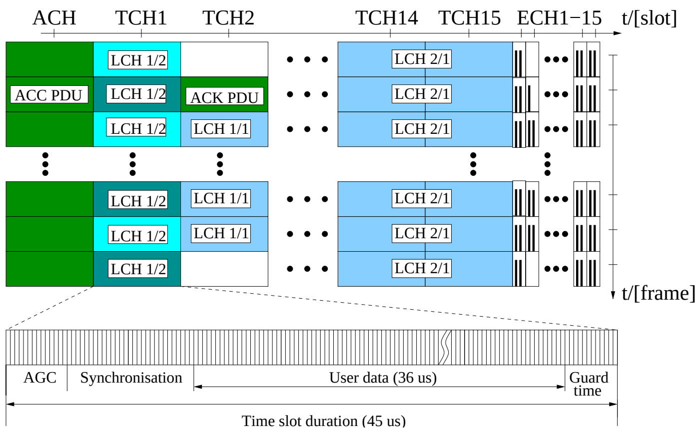


Figure 4.7: W-CHAMB MAC channel structure

The first slot of the MAC frame is used as an ACH to send an *acc s-PDU* [15]. Other slots in the frame are used as traffic channels (TCHs). Each TCH is associated with one Busy-E-signal as well as one ACK-E-signal. If a TCH is reserved, the receiver sends a Busy-E-signal on the related ECH. A successful reception of a data PDU is acknowledged by an ACK-E-Signal. Busy-E-signals and ACK-E-signals are sent on the related ECH to solve the hidden station problem and to acknowledge the successful reception of a data PDU. A duplex channel can be realized either by TDD (time division duplex) operation of one TCH [79] or by two independent TCHs with the related ECH.

To use channel resources more flexibly for heterogeneous applications, periodic slots are used as physical channels to provide transmit capacity for several LCHs. An LCHx/y is using x slot(s) per y frame(s). x gives the number of slots used per frame, and y is the repetition period of these slots counted in frames. E.g., an LCH1/2 defines a physical channel capacity according to one slot every second frame, whilst an LCH2/1 uses two slots in every frame, see Figure 4.7. By means of defining different slot-to-frame relations sub-multiplexing of the capacity represented by one TCH becomes possible. The smallest traffic channel capacity available is LCH 1/y where y is a design parameter that is chosen according to the traffic characteristic and QoS requirements.

4.3.2.2 Channel Occupation List

Like DECT systems [39], a channel occupation list (COL) that records TCH status is maintained in the MAC entity. According to the measured signal level called RSSI (Radio Signal Strength Indicator) on a TCH and its related ECH, a WS determines the usability of the TCH. A TCH may have one of the following TCH status:

- *Free*, it can be used to transmit or receive
- *Interfered*, it may be used to transmit
- *Hidden*, it may be used to receive
- *Busy*, it cannot be used to either receive or transmit
- *Reserved to receive*, it is reserved to receive by itself
- *Reserved to transmit*, it is reserved to transmit by itself

Table 4.2 is an example of a COL. For the purpose of easy explanation, it is assumed that TCH 1 to TCH 6 have different status. A TCH is viewed as *Free* if neither transmission on this TCH nor Busy-E-signal on the related ECH is detected. If a transmission on the TCH is detected, but no Busy-E-signal on the related ECH, the TCH status is *Interfered*. This TCH cannot be reserved to receive but can be used to transmit. The transmission on the TCH will not cause interference to other WSs because no related Busy-E-signal is detected by this WS. The TCH status is viewed as *Hidden* if a Busy-E-signal on the related ECH is detected, but no transmission on the TCH. Such a TCH may be reserved to receive. A TCH is marked as *Busy* if a transmission on the TCH as well as Busy-E-signal on the related ECH is detected. If a TCH is reserved to receive or transmit by the WS itself, the detection is unnecessary/impossible for this TCH.

Table 4.2: Example of channel occupation list (COL)

TCH No.	Tx on TCH	Busy-E-signal	TCH status
1	false	false	*Free*
2	true	false	*Interfered*
3	false	true	*Hidden*
4	true	true	*Busy*
5	-	-	*Reserved to receive*
6	-	-	*Reserved to transmit*
...			

In W-CHAMB networks, every WS has its own COL. To better understand the TCH status, Figure 4.8 shows an example of a W-CHAMB network. Before TCH statuses are discussed, we define the maximum communication range (R_c) as well as the the maximum detection range (R_d) of a WS. All WSs within the maximum communication range of a WS can communicate directly with this WS. All WSs within the maximum detection range of a WS can detect the transmission of this WS. R_d is usually much larger than R_c. Assuming the minimum carrier-to-interference-ratio (C/I) needed by a receiver to decode a PDU with an acceptable PDU error rate is γ, the relationship between the maximum communication range R_c and the maximum detection range R_d of a WS should meet the criteria $10log(\frac{R_d}{R_c})^{-\alpha} >= \gamma$ so that the receiver will not be interfered by transmissions of WSs that are out of the detection range of this receiver. α indicates the path loss parameter of the receiving power. R_d should be defined as small as possible to increase the frequency spatial reuse gain. For the explanation of the TCH status, it is assumed that TCH 2 is reserved by S_1 to transport PDUs to S_2, and S_2 transmits Busy-E-signals on the related ECH. The status of TCH 2 is *Reserved to transmit* in the COL of S_1, and *Reserved to receive* in the COL of S_2. According to Figure 4.8, S_{14} and S_{18} are out of the detection range of both S_1 and S_2, so the status of TCH 2 in COL of S_{14} and S_{18} is *Free.* S_{15}, S_{16} and S_{17} can sense the transmission of the S_1, but they are out of the detection range of S_2. So the status of TCH 2 in the COLs of these WSs is *Interfered.* TCH 2 may be used by one of these WSs to transmit PDUs to S_{18}. The status of TCH 2 in the view of S_{11}, S_{12} and S_{13} is *Hidden* as they cannot sense the transmission of S_1 on TCH 2, but if they send PDUs on TCH 2, S_2 is interfered. So they are hidden stations to S_2 concerning

TCH 2. As S_{11}, S_{12} and S_{13} can detect the Busy-E-Signal on the related ECH sent by S_2, which indicates that TCH 2 is reserved to receive, the hidden station problem is solved completely. However, TCH 2 may be used by S_{11}, S_{12} or S_{13} to receive PDUs from S_{14} as they are not interfered by the transmission of S_1. By allowing stations to receive on the TCH with *Hidden* status and to send on the TCH with *Interfered* status, the grade of frequency spatial reuse can be increased in a self-organizing network. For S_3, S_4, S_5 , S_6, S_7, S_8, S_9 and S_{10} TCH 2 is fully busy. TCH 2 cannot be used by these WSs.

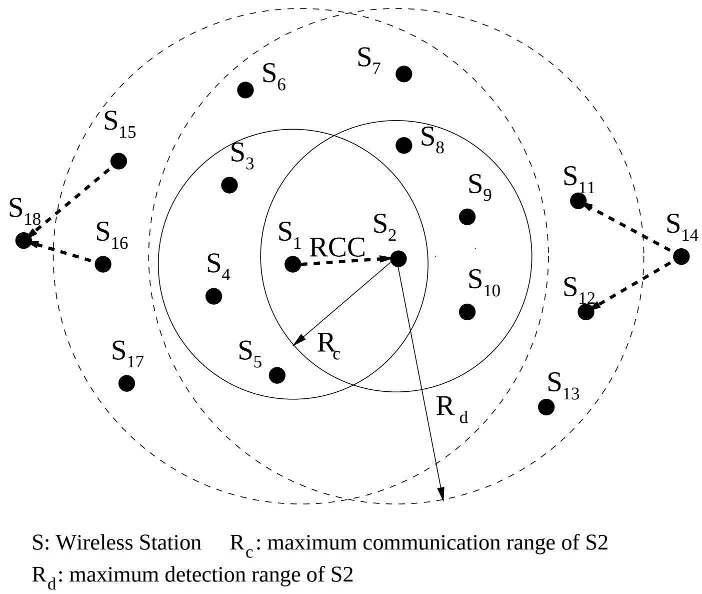


Figure 4.8: Example of a W-CHAMB network for TCH status

4.3.2.3 MAC Level Acknowledgment

The receipt of a PDU on the reserved TCH requires the receiver to respond with an acknowledgment. This technique is known as positive acknowledgment. For the purpose of a positive acknowledgment in a W-CHAMB network a receiver sends an ACK-E-Signal on the related ECH when a PDU is received on the reserved TCH.

Lack of receipt of an expected ACK-E-Signal on the related ECH indicates to the sender that an error on the reserved TCH has occurred. This information is passed to the LLC entity by the primitive MAC_RCC_STATUS.indication for the fast ARQ-scheme, see Section 4.4.4. The erroneous PDU is repeated on the next transmission opportunity.

An ACK-E-signal on the ECH is uniquely related with a reserved TCH by appropriately setting the decision signal level, called ACK decision signal level (ADSL). The ADSL should be much larger than the detection level, but smaller than or same as the decode level of the receivers. The ADSL might be adjusted according to the actual needs. To describe the operation of MAC level ACK with ACK-E-signals, we assume that S_1 has sent a PDU to S_2 on the reserved TCH 2. There are three possible results of this PDU:

1. S_2 receives the PDU successfully and sends an ACK-E-signal on the related ECH. S_1 detects this ACK-E-signal with a signal level higher than ADSL. The transmission of this PDU on TCH 2 is acknowledged correctly.

2. S_2 receives the PDU successfully and sends an ACK-E-signal on the related ECH. But S_1 cannot detect the ACK-E-signal due to heavy fading. The PDU shall be repeated by S_1. Thus, S_2 receives the correct PDU twice. The duplicated PDU is filtered out by the sequence number of the PDU. The probability that a PDU is received twice can be kept reasonably low by selecting an appropriate ADSL.

3. S_2 does not receive the PDU correctly. Then no ACK-E-signal is sent by S_2. The correct operation of the MAC level acknowledgment requires that S_1 does not detect any ACK-E-signal related with TCH 2. A false acknowledgment happens if S_1 detects an ACK-E-signal related with TCH 2 sent by other WSs. As no WSs other than S_2 can reserve TCH 2 to receive in the detection range of S_1 due to the transmission of the S_1, no ACK-E-signal related with TCH 2 will be sent by other WSs except S_2 in the detection range of S_1. WSs which are out of the detection range of S_1 may send ACK-E-signals for TCH 2 due to the spatial reuse of TCH 2. But those ACK-E-signals reach S_1 with a much lower signal level than the agreed ADSL. So no false

acknowledgment shall be received by S_1. The erroneous PDU sent on TCH 2 will be retransmitted until it is correctly received.

A fast ARQ scheme can be realized in the LLC layer using MAC level ACK-E-Signals. Another advantage using ACK-E-Signals is that WSs can use a TCH with the status *Interfered* to transmit and a TCH with the status *Hidden* to receive by the separation of the data channel and the acknowledgment channel. This increases the grade of the frequency spatial reuse and improves the system efficiency.

4.3.2.4 Distributed Prioritized Access

A distributed prioritized access (DPA) algorithm is developed to prioritize the random access and to reduce the collision probability on the ACH, see Figure 4.9. After receiving the primitive MAC_RCC.request from the LLC entity, the MAC entity attempts to reserve an RCC. The whole PDU train will be dropped if the MAC entity cannot establish an RCC before the access deadline D_{acc} to avoid transporting part of the PDU train that is useless for the application and will be dropped by the higher layer in the end user terminal. The access priority is decided based on the D_{acc} that is scheduled in the LLC entity based on the QoS requirement of the traffic service. As non real time traffic (ABR) is delay insensitive it is assigned a lower priority than real time traffic (rt-VBR) in any case.

The DPA is specified as following:

if $((D_{acc} - now()) <= T_{urgent}$

  then $P_{type} := P_{urgent}$

 else if $((D_{acc} - now()) > T_{urgent}$ and B_{type} is rt-VBR

  then $P_{type} := P_{rt-VBR}$

 else if B_{type} is ABR

  then $P_{type} := P_{ABR}$

$P_{acc} := Rand(P_{min}, P_{max})$

where T_{urgent} is a time duration. An access request is urgent if its allowed access delay is smaller than T_{urgent}; P_{acc} is the access priority calculated by this algorithm; P_{max} is the highest access priority with the largest priority value; P_{min} is the lowest access priority with the smallest priority value; P_{type} is the priority type. P_{urgent} is the urgent access priority type; P_{rt-VBR} is the real time traffic priority type; P_{ABR} is the non real time traffic priority type; $Rand(x, y)$ is a function that returns random integer value between x and y; B_{type} is the traffic type; T_{urgent}, P_{min} and P_{max} are system variables. Different priority types use different P_{min} and P_{max}. They meet the following rules: P_{min} of $P_{urgent} > P_{max}$ of P_{rt-VBR} and P_{min} of $P_{rt-VBR} > P_{max}$ of P_{ABR}.

DPA effectively assigns a higher priority to an urgent PDU train to support the QoS requirement. A random function $Rand(x, y)$ is used to generate the access priority to reduce the possibility that more than one WS sends an *acc s-PDU* on the same ACH, which results in a collision on the ACH if the addressed WSs are within the interference range of the sending WSs. The difference between x and y should be large enough to keep a low collision probability.

DPA is implemented using a combination of ACC-E-signals. For example, if a combination of four ACC-E-Signals is used and a WS contends for ACH with a priority, say 14, the WS sends three ACC-E-signals on the first three positions and listens for the fourth position. If there is no E-signal during the listening duration, the WS survives from the contention and transmits its *acc s-PDU*. The most significant ACC-E-signal is transmitted at first. So a contending WS withdraws its further contention for the right of transmission as soon as it detects an ACC-E-signal on the previous position. In Figure 4.9, S_1 survives from the contention with S_2 that has an access priority of 13 (1101). S_2 does not send its ACC-E-signal on the fourth position as it detects an E-signal on the third position. This implementation can be viewed as a much simplified vision of EY-NPMA (Elimination Yield Non Preemptive Priority Multiple Access) of Hiperlan/1 standard [40].

4.3.2.5 RCC Setup Procedures

After receiving the primitive MAC_RCC.request from the LLC entity, the MAC entity attempts to establish an RCC based on the parameters in

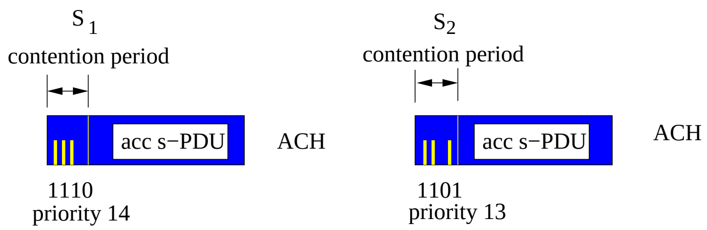


Figure 4.9: Distributed access priority

this primitive. The RCC establishment procedures (see Figure 4.7) can be described as follows:

1. The MAC entity of the source WS selects a number of TCHs that can be used to transmit according to the channel status in its COL. TCHs with the status *Free* are selected first. If there are not enough free TCHs, TCHs with the status *Interfered* can also be selected to transmit. In the case that there are not enough TCHs for an RCC required by the LLC entity, ABR RCC interruption procedures described in Section 4.3.2.6, are activated if the access request is urgent. If it is not urgent this access request is queued in the MAC entity. This step is repeated at the next access opportunity.

2. If enough free TCHs can be used to establish an RCC, the source WS contends for the right to transmit an *acc-s-PDU* on the ACH. The MAC entity of the source WS decides the access priority based on the DPA specified in Section 4.3.2.4. If there are no other WSs that contend for the transmission on the ACH with a higher priority, this WS is granted the right to send an *acc-s-PDU* on the ACH. The *acc-s-PDU* contains the information about the source STA ID (station identifier), destination STA ID, VC ID (virtual connection identifier), RC (real connection) type and a list of proposed TCHs.

3. Upon receipt of the *acc s-PDU*, the addressed WS looks up its COL to find if the TCHs proposed by the source WS can be reserved to receive. As described in Section 4.3.2.2, a TCH with the status *Free or Hidden* can be used to receive. If the addressed WS can find at least one of the proposed TCHs that can be used to receive, it responds to the source WS with an acknowledgment (*ack*) *s-PDU* and starts sending

a Busy-E-signal on the related ECH. The *ack s-PDU* is sent via the selected TCH if the selected TCH is not interfered in the view of the source WS. Otherwise, it is sent via the ACH with a higher priority type than that of the received *acc s-PDU*.

4. After the source WS receives the *ack s-PDU*, an RCC between the source WS and the addressed WS is established. The source LLC entity will be informed with the primitive MAC_RCC.confirm. LLC data PDUs are transported via the established RCC. All other WSs in the detection range of source WS and/or addressed WS will mark the reserved TCH in their COL. The hidden station problem is solved by the Busy-E-signal sent by the receiver.

5. If the source WS does not receive the expected *ack s-PDU*, the source MAC entity goes into the backoff process. There are three possible reasons that the source WS does not receive an *ack s-PDU*:

   - The addressed WS has not received the *acc s-PDU* due to a collision or fading.
   - The addressed WS cannot find any TCH out of the proposed TCHs that can be used to receive.
   - The source WS cannot receive the *ack s-PDU* due to a collision or fading.

   The backoff value is determined based on the access type to keep the backoff value of an urgent access as small as possible. Because of the distributed prioritized access that avoids most collisions on the ACH, the effect of the backoff process on the traffic performance is not as significant as with other systems, e.g. IEEE 802.11 [29]. The backoff value is decreased at the beginning of each frame. Once the backoff value is reduced to zero, this WS is allowed to contend for the transmission right on the ACH again.

4.3.2.6 ABR RCC Interruption Procedures

If a source WS cannot find enough TCHs for an RCC requirement with an access type P_{urgent}, it carries out the ABR RCC interruption procedures. The goal of the ABR RCC interruption is to release an ABR RCC to favor an urgent RCC requirement so that the dropping probability of rt-VBR PDU trains can be reduced.

There are four different ways to interrupt an ABR RCC:

1. If a WS has established an ABR RCC to transmit ABR PDUs to the same WS as the destination of the urgent rt-VBR PDU train, this ABR RCC is used to transport the rt-VBR PDU train directly. The transmission of ABR PDUs is resumed at the end of the rt-VBR PDU train.

2. If an ABR RCC that is established between the source WS and another WS that is not the destination of the urgent rt-VBR PDU train, RCC reassociation procedures are necessary. An RCC cannot be used directly by another pair of LLC entities because (1) any RCC is associated with two specific LLC entities at the time the RCC is established. (2) The reserved TCH may not be used to receive PDUs by another WS. As an example, Figure 4.10 shows that S_1 is sending ABR PDU trains to S_2 via RCC_1. RCC_1 is associated with the LLC entity responsible for the communication between S_1 and S_2. Assuming that S_1 has an urgent rt-VBR PDU train to S_3 and cannot find any free TCH for this PDU train, S_2 wants to use RCC_1 to transport this urgent rt-VBR PDU train. RCC_1 must be reassociated with the LLC entities responsible for the communication between S_1 and S_3 before RCC_1 can be used to transport the urgent rt-VBR PDU train to S_3.

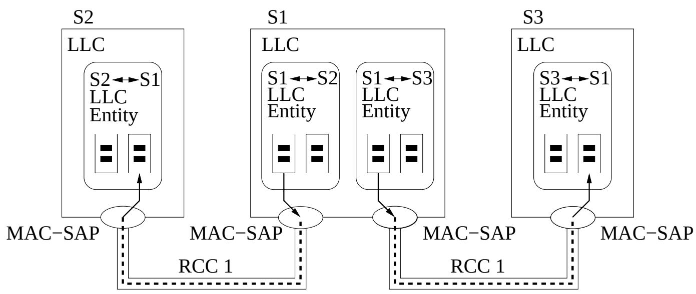


Figure 4.10: RCC reassociation

The RCC reassociation procedures are shown in Figure 4.11 and described as follows:

(a) S_1 contends for the right to send an RCC reassociation *s-PDU* to S_3 on the ACH and starts a timer to wait for the reply from S3. This RCC reassociation *s-PDU* includes parameters of RCC_1. The access priority is decided by the same way as to send an *acc s-PDU*.

(b) If S_3 receives this RCC reassociation *s-PDU*, it checks the reserved TCHs of RCC_1. If these TCHs are not interfered in the view of S_3, S_3 responses to S_1 with an RCC reassociation *ack-s-PDU* on the ACH.

(c) After the receipt of the RCC reassociation *ack-s-PDU* from S_3, S_1 sends an RCC deassociation *s-PDU* to S_2 on the reserved TCH of RCC_1. Upon receipt of a positive acknowledgment of the RCC deassociation *s-PDU* from S_2, RCC_1 is reassociated with the LLC entities responsible for the communication between S_1 and S_3. After that S_1 starts sending data PDUs via RCC_1 to S3. S3 is informed implicitly about the completion of the reassociation procedures by the first data PDU addressed to it.

(d) If S1 does not receive any reply from S3 after the timer is out, the interruption procedure is repeated after a backoff process.

3. If the source WS has not any ABR RCC established by itself, it tries to decode PDUs transmitted by its neighbors. If the source WS can find TCHs on which ABR PDUs are sent, it sends an RCC release request *s-PDU* via the ACH to its neighbors. After receipt of the release request *s-PDU*, WSs that are sending on these TCHs release their RCCs accordingly. After that, S_1 uses the RCC establishment procedures to establish an RCC for its urgent rt-VBR PDU train.

4. If the source WS cannot find any TCH reserved by its neighbor for ABR PDU transmission, it sends an RCC release request to the destination of its urgent PDU train. The destination takes the responsibility to ask its neighbors to release TCHs on behalf of the source WS having an urgent rt-VBR PDU train.

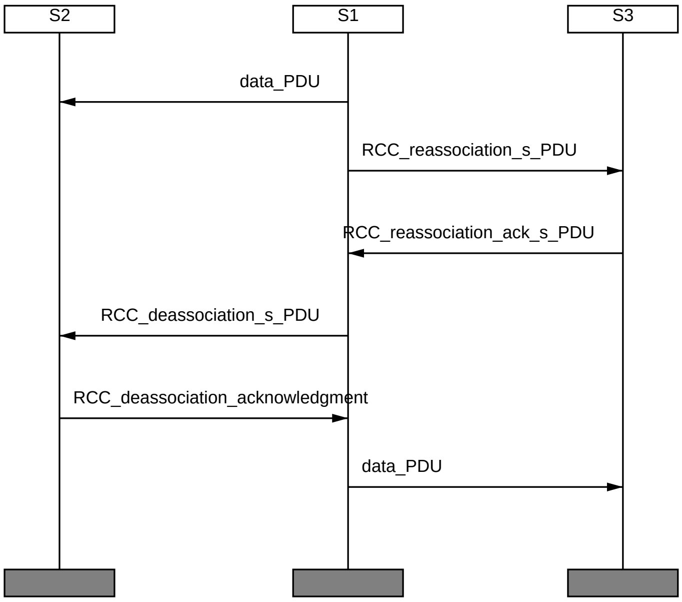


Figure 4.11: MSC of RCC reassociation

If S_1 cannot establish an RCC for its rt-VBR PDU train before its access deadline D_{acc} after all above attempts are used, the rt-VBR PDU train has to be dropped. The dropping probability must be kept very low to meet the QoS requirement of rt-VBR services. This can be achieved via a distributed measurement-based call admission control (CAC) to avoid an overloaded situation, see Section 4.5.3.

4.3.2.7 ABR RCC Bandwidth Adjusting

The bandwidth of the established ABR RCC is dynamically adjusted according to the load situation of the network as well as the source WS itself. Upon receipt of an RCC bandwidth increase request from the LLC entity that adjusts the bandwidth of an ABR RCC based on the Dynamic Channel Allocation (DCA) algorithm specified in Section 4.4.2.3, the MAC entity forms a bandwidth increase *s-PDU* including a list of proposed TCHs that can be used for transmission. If the source WS does not receive an *ack-s-PDU* from the addressed WS, the MAC entity of the source WS does not queue this bandwidth increase request. It is the responsibility of LLC entity to reschedule the bandwidth requirement after a backoff process.

4.3.2.8 Data PDU Transport

After an RCC is established between a pair of peer to peer LLC entities, the source LLC entity begins to transport data PDUs using the primitive MAC_Data.request. As an RCC is associated with the LLC entities at the time of RCC establishment, the source STA ID and destination STA ID are not required to be included in the data PDU, hence the protocol overhead is reduced significantly. The only MAC level overhead is the PDU Type ID to identify *s-PDUs* sent via the RCC.

The receipt of data PDU on the reserved TCH is acknowledged using an ACK-E-signal sent on the related ECH. Unacknowledged data PDUs are retransmitted. It is possible that a LLC PDU is received more than once if the source WS fails to detect an ACK-E-signal. Duplicate data PDUs are filtered out within the destination LLC entity using a sequence number in the PDU.

If the number of successive unacknowledged PDUs exceeds a handover threshold, the reserved TCH is no long suited to be used to transmit/receive and must be handovered due to interference or loss of connectivity. So the source WS releases this reserved TCH and informs the LLC entity with the primitive MAC_RCC_STATUS.indication. A TCH handover *acc-s-PDU* is formed by the MAC entity to reserve a new TCH. The access deadline D_{acc} of the handover *acc-s-PDU* is decided based on the traffic type. Unsuccessful handover access is repeated after a backoff process. If the handover access is unsuccessful, the remaining data PDUs must be dropped.

4.3.2.9 Release of RCC

An RCC is released immediately if the LLC entity has no further PDUs to send. A reserved TCH is released implicitly when the source WS stops sending on it. All WSs in the detection range of source WS can detect the stop of transmission and mark the TCH in their COLs as free.

The destination WS regards the RCC as released if the signal level on the reserved TCH is lower than the channel holding threshold and stops sending the related Busy-E-signal. The WSs which are in the detection range of the destination WS can detect the stop of Busy-E-signal and update their COLs.

4.4 W-CHAMB LLC Layer

The structure of LLC entities are shown in Figure 4.12. An LLC entity is responsible for the communication with a neighbor WS. In each entity there is one or more transmit/receive queues, each for one virtual connection (VC). An LLC scheduler is used to plan the RCC requirements and to schedule the transmission of PDU trains. Train level scheduling instead of PDU level scheduling is used to reduce the computing power requirement and to provide QoS to PDU trains other than individual PDUs. This is based on the consideration that it is useless to deliver part of the PDU train that will be discarded in the destination. For an application, the end-to-end delay and dropping probability of a whole PDU train are more relevant measures of performance than ones specified for individual PDUs.

Each LLC entity has an ARQ buffer to back up the transmitted PDUs for the possible retransmission due to the channel error. The buffered PDUs are identified by an RCC ID. An unacknowledged PDU shall be retransmitted by the fast-ARQ-scheme described in Section 4.4.4.

4.4.1 The Flow Specification

Consider a traffic source that generates a sequence of application layer data units for network delivery to the same destination. Most of these data units will be too large to be carried in a single PDU and must be segmented in the W-CHAMB convergence layer for the wireless delivery. The sequence of PDUs encapsulating an application layer data unit forms a PDU train [63, 75]. To provide QoS to the whole PDU train, a traffic flow model is specified. In the traffic flow model, a flow is a sequence of trains, each of which is a sequence of PDUs that encapsulate an application layer data unit.

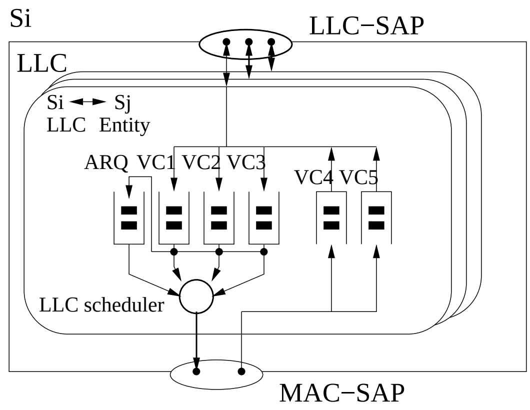


Figure 4.12: LLC layer

Since PDU trains vary greatly in length, the information on the length of a PDU train can be exploited to optimize the traffic performance.

The traffic flow, shown in Figure 4.13, is specified as follows:

- Four PDU types are specified: Begin of the Train (BOT), Continuing of the Train (COT), End of the Train (EOT) and Single PDU of the Train (SOT). The first and last PDU of a PDU train is uniquely identified by BOT and EOT.
- The first PDU of the train carries information on its length. The LLC scheduler exploits this information to plan an RCC based on the scheduling algorithm described in Section 4.4.2.

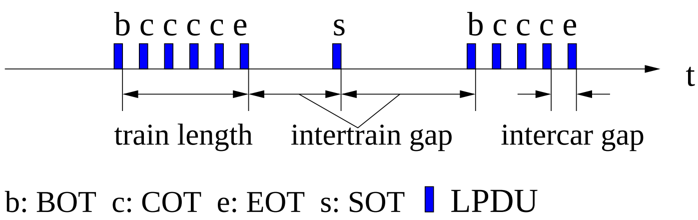


Figure 4.13: Traffic flow model

4.4.2 The LLC Scheduling Algorithm

Following the arrival of the BOT PDU of a PDU train, the LLC scheduler plans the channel resource requirement for this PDU train. The scheduling algorithms for rt-VBR and ABR traffic are described in Section 4.4.2.1 and Section 4.4.2.2.

4.4.2.1 rt-VBR Scheduling Algorithm

The principle of the rt-VBR scheduling algorithm is to keep the transmission duration of an rt-VBR PDU train within a time limit T_d. T_d is a design variable that should be selected based on the QoS requirement of rt-VBR services. The EOT PDU of the PDU train experiences a maximum delay that is the transmission duration of the PDU train plus the access delay to establish an RCC. The EOT PDU delay must be less than the maximum delay allowed by rt-VBR services. Assuming that L is the number of PDUs of a rt-VBR PDU train, r, r_1 and r_2 represent RCC bandwidths, the rt-VBR scheduling algorithm is specified as follows:

1. If there is an ABR RCC available and $L/r_1 <= T_d$, then the ABR RCC is used directly by the rt-VBR PDU train. r_1 is the bandwidth of the ABR RCC. $L/r1 < T_d$ means that the bandwidth of the ABR RCC is enough to transmit the rt-VBR PDU train. The ABR RCC is changed to rt-VBR RCC directly, and no new RCC requirement is generated. The transport of the ABR PDUs is interrupted by the scheduler and the rt-VBR PDUs are sent immediately. The statistical multiplexing among ABR and rt-VBR traffic can be realized most efficiently in this case. The transmission of ABR PDUs resumes at the end of the rt-VBR PDU train.

2. If there is an ABR RCC, but $L/r_1 > T_d$, which means that the bandwidth r_1 of the ABR RCC is not enough for the LLC entity to transport the rtVBR PDU train. In this case, an additional bandwidth r_2 is scheduled. r_2 meets the following relation:

   $L/(r1 + r2) <= T_d$

   The access deadline D_{acc} for the MAC layer to reserve the additional bandwidth r_2 is calculated as follows:

$D_{acc} = T_{life} - L/(r1 + r2) + now()$

T_{life} is the time left before the PDU train has to be discarded.

Anyway, the ABR RCC with rate r_1 is immediately used to transmit rt-VBR PDUs.

3. If no ABR RCC is available, the bandwidth required for the rt-VBR train is scheduled as follows:

   $L/r <= T_d$

   The access deadline D_{acc} is calculated as follows:

   $D_{acc} = T_{life} - L/r + now()$

The rt-VBR PDUs are transmitted via the RCC as soon as it is available. After the EOT or the SOT of an rt-VBR train is transported, the rt-RCC is either released or changed to ABR RCC if there is any ABR PDU available.

4.4.2.2 ABR Scheduling Algorithm

The ABR scheduling algorithm aims to use channel resources efficiently as well as fairly among WSs in the W-CHAMB networks. The LLC entity schedules the ABR PDU trains as follows:

1. If there is an ABR RCC available, nothing is done upon the arrival of a new ABR PDU train. The bandwidth of ABR RCC is adjusted by the dynamical channel allocation (DCA) algorithm described in Section 4.4.2.3.

2. If there is an rt-VBR RCC but no ABR RCC, the DCA algorithm is activated to see whether a new ABR RCC should be established.

3. If there is neither an ABR RCC nor rt-VBR RCC available, an initial ABR RCC should be established. The bandwidth and access deadline D_{acc} for the initial ABR RCC requirement are fixed. After an ABR RCC is established, the bandwidth of this ABR RCC is dynamically adjusted by the DCA algorithm.

4.4.2.3 Dynamic Channel Allocation (DCA) Algorithm

The bandwidth of an ABR RCC is adjusted dynamically according to the number of the queued ABR PDUs in the LLC entity and the number of free TCHs in a MAC frame. A large number of free TCHs means that the network is lightly loaded so that more TCHs should be allowed to be reserved by ABR RCC even when the number of the queued ABR PDUs is not very large. If the the number of the free TCHs is small, the network is heavily loaded. In this case, a WS decreases the bandwidth of the ABR RCC to share the channel resources fairly among ABR services and to ensure the QoS guarantee for rt-VBR services.

The DCA algorithm is specified as follows:

If $N_{pdu}/N_{tch} \geq N_1$ and $N_{free} \geq N_{free1}$ and $N_{tch} < R$

  then increase the bandwidth of the ABR RCC

If $N_{pdu}/N_{tch} \geq N_2$ and $N_{free} \geq N_{free2}$ and $N_{tch} < R$

  then increase the bandwidth of the ABR RCC

If $N_{pdu}/N_{tch} \geq N_3$ and $N_{free} \geq N_{free3}$

  then increase the bandwidth of the ABR RCC

If $N_{tch} > R$ and $N_{free} \leq N_{free4}$

  then decrease the bandwidth of the ABR RCC

If $N_{pdu}/N_{tch} \leq N_4$ and $N_{free} \leq N_{free4}$ and $N_{tch} > 1$

  then decrease the bandwidth of the ABR RCC

where $N_1 > N_2 > N_3 > N_4$ and $N_{free4} \leq N_{free1} < N_{free2} < N_{free3}$; N_{pdu} is the number of the ABR PDUs queued in the LLC entity; N_1, N_2, N_3 and N_4 are four system variables defining the grade of traffic load in the LLC entity. The parameter set (N_4, N_3, N_2, N_1) is studied in Section 6.2.2.2 in detail. N_{free} is the number of free TCHs in a MAC frame. N_{free1}, N_{free2}, N_{free3} and N_{free4} are four system variables defining the grade of the network load. N_{tch} is the number of TCHs reserved by this LLC entity.

The parameter R is the maximum number of TCHs that a WS can reserve for an ABR RCC in the case that the number of free TCHs is smaller than N_{free3}. In the line 5 of the DCA algorithm we can see that R is not used if the number of free TCHs is larger than or equal to N_{free3}. Usually in the following example analyzes the parameter set (N_{free1}, N_{free2}, N_{free3} and N_{free4}) is set to (1, 2, 3, 1).

DCA aims to use the channel resources more efficiently and fairly. DCA becomes significant in the high bursty traffic case.

4.4.3 Multihop Transport of PDU Trains

Data PDUs queued in the LLC entity are transported via the RCC hop by hop. A delivered PDU is buffered for the possible retransmission. Unacknowledged PDUs are retransmitted by the fast ARQ scheme.

A received rt-VBR PDU is delivered to the W-CHAMB NL immediately. If the rt-VBR PDU should be relayed to another WS, the W-CHAMB NL routes this rt-VBR PDU to the respective relay LLC entity that is responsible for the communication with the next WS. The relay LLC entity requests an RCC following the receipt of the BOT of a rt-VBR train. The rt-VBR RCC should be hold on even there is no rt-VBR to send until the EOT PDU of an rt-VBR PDU train is transported. The LLC entity keeps an rt-VBR RCC by sending an ABR PDU if any or a dummy PDU if there is no ABR PDU to transport. The dummy PDU can be exploited to carry some organization information. The number of consecutive dummy PDUs must be limited. If the number of the consecutive dummy PDUs exceeds a limit, the rt-VBR RCC is released as the EOT PDU of the rt-VBR PDU train may probably have been dropped in the previous hop.

ABR PDU trains, on the other hand, are transported hop by hop as a whole PDU train. In the relay WS ABR PDUs are delivered to the next hop only after the whole PDU train is received. If no more PDUs are available, the ABR RCC is released immediately. To avoid a situation that some WSs occupy TCHs for a very long time and other WSs have no opportunity to send any ABR PDU train, a WS is required to check the channel situation at the end of each ABR train. If the network is highly loaded, the WS should release the ABR RCC even if there are other ABR PDU trains waiting for transport. In this case, the WS backs off for some time before it requests a new ABR RCC again.

4.4.4 The Fast ARQ Scheme

The general concept of automatic repeat request (ARQ) is to detect PDUs with error at the LLC entity and then to request the source LLC entity to repeat the erroneous PDUs. Three basic ARQ schemes are stop-and-wait, go-back-N and selective-repeat, see Section 3.2.2.

The fast ARQ scheme developed for W-CHAMB networks is as efficient as SR-ARQ and as simple as stop-and-wait ARQ. Figure 4.14 illustrates the operation of the fast ARQ scheme. The receipt of a PDU sent on TCH n during MAC frame n is acknowledged by the receiver using an ACK-E-Signal sent on the related ECH n. If the receiver cannot receive the expected PDU on the reserved TCH, the related ACK-E-Signal is not sent. No ACK-E-signal means a NAK to the sender, and the source LLC entity of the sender should retransmit the PDU. The ACK/NAK information is delivered to the LLC entity by the primitive MAC_RCC_STATUS.indication. If a NAK is indicated, the source LLC entity retransmits the PDU that is buffered in the ARQ buffer. If an ACK is received, the LLC entity transmits the next PDU. The fast ARQ is operated like the stop-and-wait ARQ scheme. But there is no additional delay needed for the LLC entity to wait for the ACK/NAK as the ACK/NAK is transported using the ACK-E-signal on the fixed time point. Actually, the source LLC entity never stops transmission due to the delay of an ACK/NAK information. This is why it is called a fast ARQ scheme. Since only failed PDUs are retransmitted, the fast ARQ scheme achieves the same efficiency as the SR-ARQ.

4.5 W-CHAMB Network Layer

The W-CHAMB NL performs the routing function and the network management function that are especially important for multihop connections. It is responsible for the establishment of an end-to-end virtual connection and performs call admission control to ensure QoS guarantee for the established VCs.

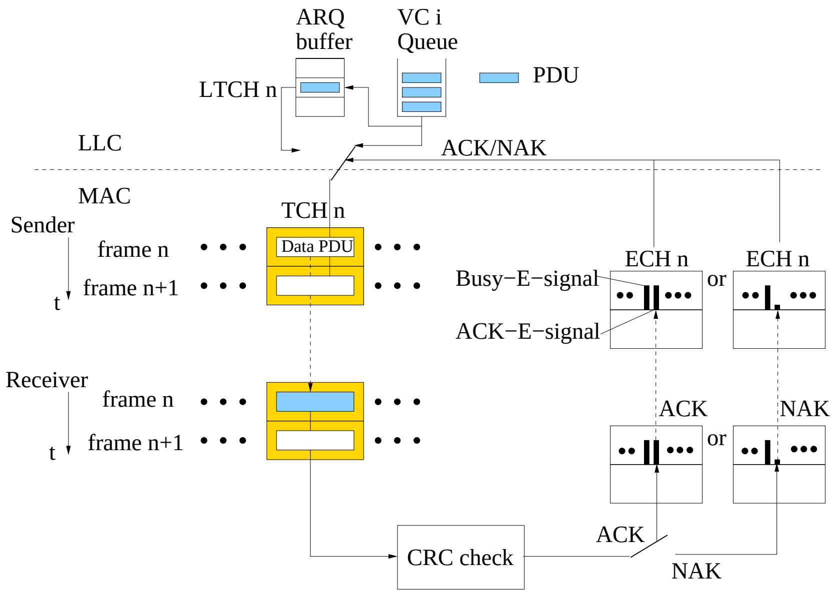


Figure 4.14: The fast ARQ scheme

To make a WS reachable by a fixed network terminal or a WS in another place, a WS in W-CHAMB networks is associated with an AP if an AP connected to a wired network exists within n hops away from this WS. If a WS cannot find an AP within n hops, it cannot be associated with an AP and cannot be reached by other WSs that are more than n hops away from it. An unassociated WS works in ad hoc mode and communicates with other WSs that are within n hops away from it.

In the following studies we assume that a multihop connection in W-CHAMB networks is usually limited to n = 4 hops. A network supporting 4 hop connections can achieve a reasonable communication coverage and still keep the network organization and routing protocol simple. A large scale network can be realized by a number of APs that can be deployed quickly without frequency planning and by using information exchange between the APs with respect to the WSs connected to them.

4.5.1 W-CHAMB Routing Protocol

Before a connection can be established, a route between the source and destination must be found. A design objective of W-CHAMB networks might be that more than 80% of the connections have fewer than or equal to 3 hops. Connections with 4 hops should be less than 20 %. If this assumption is not met, the proposed routing scheme losses efficiency and would have to be replaced. Deploying the network architecture of W-CHAMB networks, an iterative routing protocol (IRP) called W-CHAMB IRP is developed. The W-CHAMB IRP is based on a two-hop neighbor table (NT) which is maintained in each WS proactively. Table 4.3 is an example of the two-hop NT of S in Figure 4.15. F_1, F_2, F_3 and F_4 are one hop neighbors of S. B_1 is a two-hop neighbor of S and an one-hop neighbor of F_1. The maintenance of the two hop NT is proactive, which is realized by the following methods:

1. A WS broadcasts its neighbor table periodically.
2. By listening to the *acc-s-PDU* that includes STA ID of the sender and the receiver, a WS updates its two hop neighbor table. For example, if WS A receives an *acc s-PDU* sent by WS B to WS C, and if WS C is not a neighbor of WS A, WS A can learn from this *acc s-PDU* that WS B and WS C are its one-hop neighbor and two-hop neighbor, respectively.

Table 4.3: Neighbor table of WS S

one-hop neighbor	two-hop neighbor
F1	B1
F2	B2
F3	B3
F4	B4
...	...

Exploiting the information of the two-hop NT the W-CHAMB routing protocol determines the route according to the following procedures:

1. The source station checks whether the destination can be found in the two-hop NT. If so, a route to the destination is known and no further route discovery procedure is needed.

2. If the source station cannot find the destination in the two-hop NT, it broadcasts its route query to its neighbors. The neighbors that receive the route query check their two-hop NT. If the destination is found, a route reply is sent to the source station. Thus, a three hop route is found. The source WS may receive more than one route reply. In this case, the source WS selects the least loaded route that can provide the best QoS. The other ones are used as back-up routes. They can be used if the selected route is broken. This results in a QoS routing. All the WSs identified to be three hops apart and the respective routes are stored at the same WS together with a time stamp of validity.

3. If no route reply is received within a time limit, the destination is most probably out of the three hop coverage of the source station. In this case, the source station selects some of its neighbors to forward the route query so that its two hop neighbors can receive the route query. For example, see Figure 4.15, the source station S wants to communicate with its destination D. As D is not in the two-hop NT of S, S broadcasts a route query to its neighbors. As D is not in the two-hop NT of S's neighbors. S shall not receive any route reply. Thus, S broadcasts a route query forward request message including selected neighbors (F_1, F_2, F_3 and F_4) to propagate the route query to the two-hop neighbors of S, i.e. B_1, B_2, B_3 and B_4. After the receipt of the route query forward request, the selected forwarders, i.e. F_1, F_2, F_3 and F_4 broadcast the route query to its neighbors. WSs that receive the route query check their two-hop NTs. If the destination is found, a route reply is sent back to the forwarder. At this case, B_1 sends a route reply to F_1 and then F_1 sends a route reply to the source station S. A four hop route between S and D is found by this iterative way. All the WSs identified to be 4 hops apart and the respective routes are stored at the same WS together with a time stamp of validity. The 3 and 4 hop routing information is subject to an aging process, depending on the speed of mobility (loss of valid routes) experienced in the past.

In comparison with ZRP (Zone routing protocol), see Section 3.2.3, W-CHAMB IRP generates much less control traffic than ZRP by avoiding global search of the destination throughout the network. Query detection and query termination that must be used in ZRP is not necessary for W-

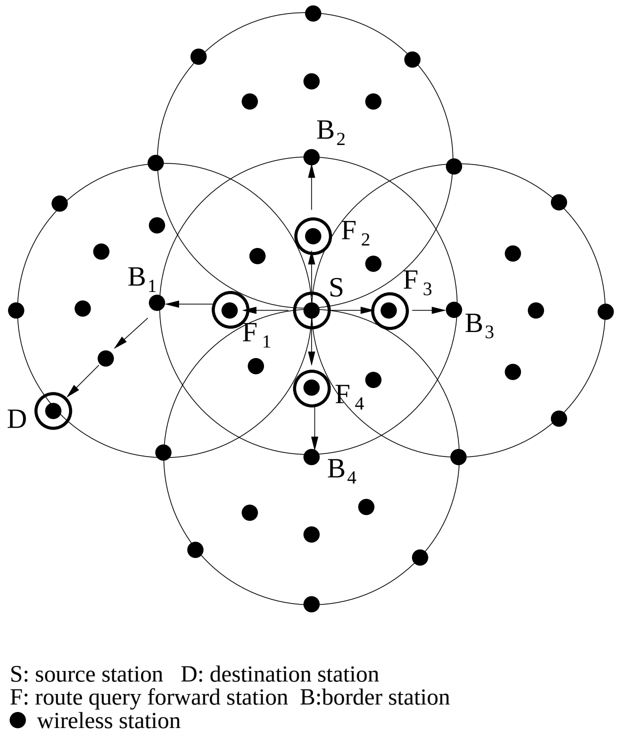


Figure 4.15: An example of the W-CHAMB IRP protocol

CHAMB IRP. The delay resulting from the route discovery can be kept very small as the route query is forwarded at most two times, if $n \leq 4$.

4.5.2 Connection Setup

Before data PDU transfer in W-CHAMB networks between source and destination can start, an end-to-end virtual connection (VC) is established for the duration of a communication session [111, 128].

In Figure 4.16 and Figure 4.17, as examples, we demonstrate the connection setup initiated by WS A to WS D. The connection setup procedure is as follows:

1. After the NL of WS A receives the primitive *Connection Setup Request* with the parameters of DEST ID (destination identifier), QoS requirements of the connection and a communication end point identifier (EP ID) assigned to this connection, the NL of WS A performs the CAC (call admission control) function specified in Section 4.5.3, to check if the required bandwidth is available in the air interface. If the CAC is successful, the NL of WS A searches the two-hop NT, see Table 4.3, to find whether WS D can be reached either directly or through one relay station (RS). If not so, the W-CHAMB IRP algorithm is invoked to find the next WS to reach the destination. In this case, the next WS on the route to WS D is WS B.

2. The NL of WS A assigns a virtual connection identifier (VCI_1) to this connection and sends a SETUP message to the NL of the WS B. VCI_1 uniquely identifies the connection between WS A and WS B.

3. After searching its two-hop NT, WS B finds that WS D can be reached by the relay of WS C. The NL of WS B performs a CAC function to check if there are enough bandwidth resources to relay this connection. If the CAC is successful, WS B assigns VCI_2 to this connection and sends SETUP message to WS C. VCI_2 identifies the connection between WS B and WS C.

4. After searching the NT, WS C finds that WS D is its neighbor. A CAC function is performed in WS C to check if there are enough bandwidth resources to relay this connection. If the CAC is successful, WS C assigns VCI_3 to this connection and sends SETUP message to WS D. VCI_3 identifies the connection between WS C and WS D.

5. WS D accepts the connection after receiving SETUP message from WS C if CAC is successful in its NL.

6. The NL of WS D sends CONNECT message to WS A on the backward route. All intermediate WSs, i.e. WS C and WS B, set the routing table in the W-CHAMB NL entity for this connection.

7. After WS A receives the CONNECT message, a three-hop end-to-end connection between WS A and WS D is established.

At the end of the communication session, the connection is released by either WS A or WS D. To support mobility during the life-time of a VC, handover

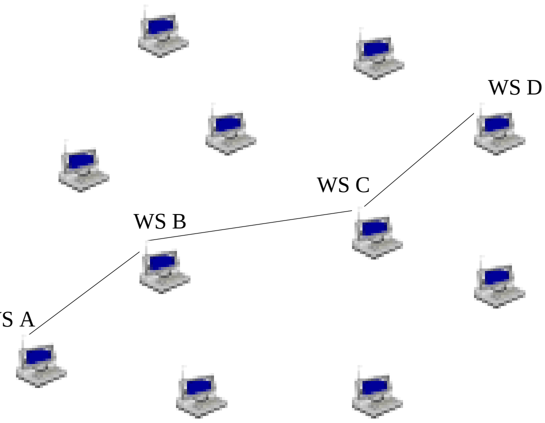


Figure 4.16: Connection establishment

procedures are necessary [3, 88, 128]. In [128], a handover protocol based on the path cut and path extension is developed for the W-CHAMB network.

4.5.3 Call Admission Control

A call admission control (CAC) function is invoked during the connection setup to avoid the QoS degradation of the established connections and to guarantee the QoS requirements of the establishing connection. CAC is especially important for real time traffic. As bandwidth resources of the W-CHAMB network are shared in fully uncoordinated manner, the CAC algorithms that have been studied for fixed networks and centrally organized systems [51, 68, 83] cannot be applied in W-CHAMB networks. The CAC function in W-CHAMB networks can only be implemented distributedly. A WS makes a call admission decision according to the channel occupation condition found in its local COL.

Different CAC algorithms should be used for different service types. CAC is not necessary for ABR *Best Effort* services. All such connections may be accepted. Instead, a flow control mechanism for these connections should be applied to avoid network congestion. When the network is heavily loaded, the higher layer protocol should not generate further ABR PDUs to LLC

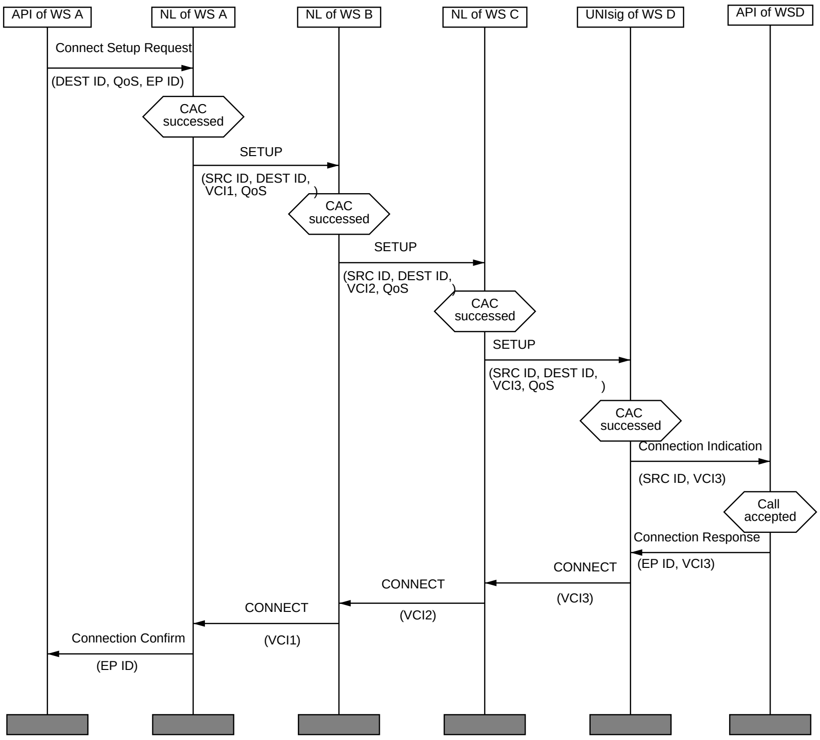


Figure 4.17: Connection setup procedure

entities. For the rt-VBR services, a strict CAC is required to achieve a QoS guarantee for it. The amount of rt-VBR traffic on the radio channel must be limited. An rt-VBR connection can be accepted only if there are enough channel resources available.

The CAC algorithm for rt-VBR services is specified as follows:

rt-VBR_CAC(λ_{peak})

(in source and destination WS)

1 $C_{rt-VBR} = (N_{free} + N_{ABR} - N_{min})/T$

```
2        if   (C_{rt-VBR} > λ_peak)
3            then    accept the call
4        else   reject the call
```

rt-VBR_CAC(λ_{peak})

(in relay WS)

```
1    C_{rt-VBR} = (N_free + N_ABR − N_min)/T
2        if   (C_{rt-VBR} > 2 ∗ λ_peak)
3            then    accept the call
4        else   reject the call
```

where C_{rt-VBR} is the available capacity for rt-VBR services; N_{free} is the number of free TCHs during the measurement duration T counted in frames; N_{ABR} is the number of TCHs reserved for ABR services during the duration T; N_{min} is the minimal number of free TCHs required to meet the QoS requirement of the rt-VBR connection. λ_{peak} (PDU per frame) is the peak rate of the new rt-VBR connection;

In line 1 of rt-VBR_CAC(), TCHs used by ABR traffic are regarded as available for rt-VBR services as the rt-VBR services can obtain these channel resources through the prioritized access and the ABR RCC interruption. The difference of the CAC function in relay WSs from that in source/destination WSs is that double capacity is needed for the connection in a relay WS. The peak rate of an rt-VBR connection is the maximum number of TCHs per MAC frame needed by this connection during its active period. The rt-VBR_CAC() is different from the usual peak rate CAC algorithm used in the fixed ATM network as C_{rt-VBR} is a statistical mean value over the measurement duration T. The channel resources used by rt-VBR connections are not the sum of the peak rate of the rt-VBR connections. The statistical multiplexing gain among rt-VBR connections can be adjusted by the parameter N_{min}.

CHAPTER 5

The Simulation Tool

This chapter describes an integrated stochastic simulation tool for the traffic performance evaluation. The simulation tool is based on the prototypic implementation of the W-CHAMB protocols in SDL. In fact the W-CHAMB network is emulated within the simulator. Using realistic source traffic models and channel models, a self-organizing multihop network with its protocols can be evaluated in detail. In comparison with analytical methods using probability theory, queuing theory and Markov theory, computer simulation is able to evaluate much more complicated systems stochastically.

5.1 Software Architecture

Figure 5.1 shows the software architecture of the developed simulation tool. It has the following major components: the SDL specification of the protocols, the channel model, the location management, the traffic sources, the statistical performance evaluation and the configuration parameters.

The SDL specification is the key component of the simulator. The protocols and algorithms are specified in SDL. Through the C++ code generator SDL2SPTEECL, the SDL specification is translated into SPEETCL C++ class that generates communication objects that are representations of WSs [4, 103]. Each communication object is associated with a traffic source generating communication connections (calls) and data traffic streams. The communication objects communicate with each other through a channel model simulating the realistic radio transport medium. The signal strength and data errors are dependent on the location of the communication object with respect to the transmitter and interference, which is determined by the location management. The simulation result is measured and evaluated by the statistical evaluation components. By using the read-in file that contains the configuration parameters, the simulation program need not to be recompiled to change the simulation configurations.

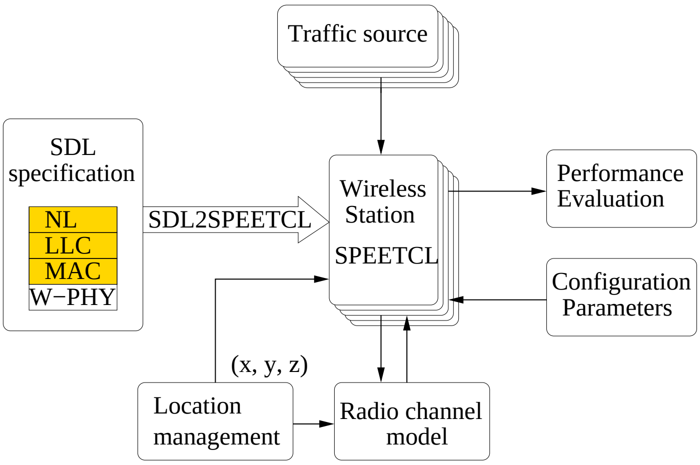


Figure 5.1: Software architecture

Each of the major components of the simulation tool is described in the following Sections.

5.2 Protocol Specification in SDL

To specify protocols in SDL has many advantages [9, 12, 45, 62]. SDL has formal basis which helps to eliminate protocol errors. SDL supports modular design which can be easily reused. SDL is easy to read with a graphical presentation and an easily understood finite state machine basis. Moreover, SDL is well supported by commercial tools, such as Telelogic SDT and SDL2SPEETCL. By using SDT, any syntax or semantics can be automatically found. This can speed up the specification process. The graphical presentation of SDL can be changed to a phase representation by the tool SDT. The exact and formal description of the protocol can be translated into a SPEETCL C++ class using the SDL2SPEETCL compiler.

We use the object-oriented definition for the protocol specification in SDL [9]. This is achieved by the concept of packages in SDL. In a package, system types, block types and process types can be defined. SDL Types can be defined as virtual type and redefined in other place. Type definition enables to generate several entities supporting instantiation, generalization and specialization. The system, block and process instance is then generated from the type definition.

5.2.1 SDL System Type Specification

Figure 5.2 shows a SDL system type called *stCHAMB* that specifies the W-CHAMB protocol stack. It defines a set of blocks that communicate with each other and with the SDL environment via the bidirectional channels, i.e. *cEnvNL*, *cNLLLC*, *cLLCMAC*, *cMACPL* and *cPLEnv*. Signals that are included in the signal lists, *slNLToEnv*, *slEnvToNL*, *slLLCToNL*, *slNLToLLC*, *slLLCToMAC*, *slMACToLLC*, *slMACToPL*, *slPLToMAC*, *slPLToEnv* and *slEnvToPL* are transported via the channels. These Signals may carry additional parameter values or data. Each block *bNL*, *bLLC*, *bMAC* and *bPL* models the protocol layer NL, LLC, MAC and PL, respectively. The SDL system behavior that determines the W-CHAMB protocol performance is defined by the behavior of its blocks that correspond to the protocol layers.

The #MSGDEF Directive and #MSG directive are functions provided by the SPEETCL. The #MSGDEF Directive is necessary before using an #MSG directive to send debug messages from any process in the SDL specification to the environment.

5.2.2 SDL Block Type Specification

Figure 5.3, as an example, shows the W-CHAMB SDL LLC block type specification called *btLLC*. The block type *btLLC* consists of two processes, *pLLCMan* and *pLLCEnt*, that communicate with each other via the signal routes *srLManEnt* through the gate *gLME*. This communication is via the signals included in the signal lists *slLEntToMan* and *slLManToEnt*. The process *pLLCMan* communicates with the block *bLLC* and *bMAC* via the signal routes *srLLCManNL* and *srLLCManMAC* through gate *gLN* and *gML*, respectively. Signals included in the signal lists *slLLCManToNL*, *slNLToLLCMan*, *slMACToLLCMan* and *slLLCManToMAC* are transported via the signal routes. The process *pLLCEnt* communicates with the block *bLLC* and *bMAC* via the signal routes *srLLCEntNL* and *srLLCEntMAC* through gate

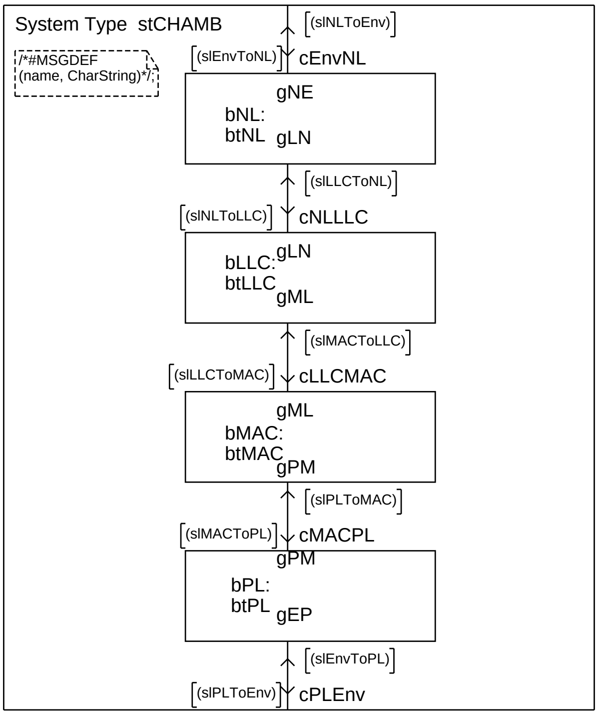


Figure 5.2: W-CHAMB SDL system type

gLN and *gML*, respectively. These signals are included in the signal lists *slLLCEntToNL*, *slNLToLLCEnt*, *slMACToLLCEnt* and *slLLCEntToMAC*.

The process *pLLCMan* is created at the system start, while the process *pLLCEnt* is created dynamically by the process *pLLCMan* at runtime. More than one instance of the process *pLLCEnt* may be created. Each instance has a unique process identifier (PId). This makes it possible to send signals to individual instances of a process. If the first connection with a neighboring communication partner is established, the process *pLLCMan* creates a new instance of the process *pLLCEnt* that takes responsibility of the communication with this partner. The functions of the LLC protocol layer described in Section 4.4 are realized by this way.

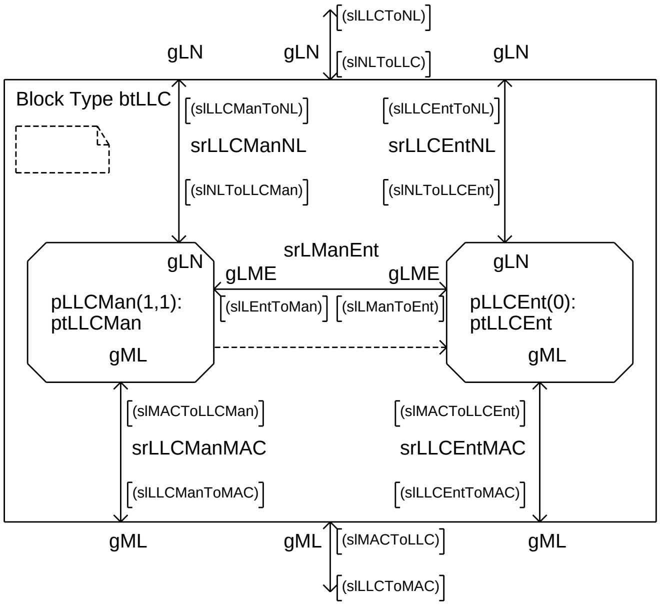


Figure 5.3: W-CHAMB SDL LLC block type

5.2.3 SDL Process Type Specification

The dynamic behavior of an SDL system is described by the SDL processes. The behavior of the blocks and the systems is derived from the behavior of the process instances. The specification of the behavior of the process instances is based on the concept of extended finite-state machines. Figure 5.4 shows a part of specification of the process type *ptLLCMan*. The extended finite-state machine is either in a stable state (e.g. *NotStarted* or *Started*) or in a transition (e.g. the procedure *pdStartUp* or *pdStop*) between the states. The procedure *pdStartUp*/*pdStop* is initiated by the receipt of an input signal *sStart*/*sStop* and ends in a new stable state. Procedures are used in the specification to realize a specific function of the protocol. Figure 5.5 shows the specification of the procedure *pdStartUp* in the process type *ptLLCMan*. The Operators *getInterger*, *getNatural* and *new* are

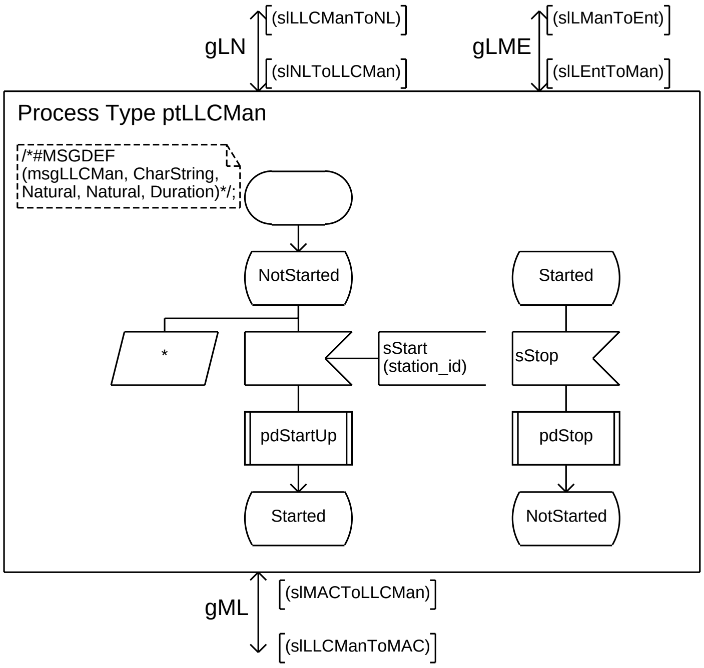


Figure 5.4: W-CHAMB SDL process type

defined in the external ADTs (Abstract Data Types). These operators are implemented by C/C++ functions.

5.3 Channel Model

The channel model simulates the physical transport medium of the wireless networks. Signals transmitted from a sender are processed by the channel model that decides the strength of the signals at the receiver of the other wireless stations of the network [115].

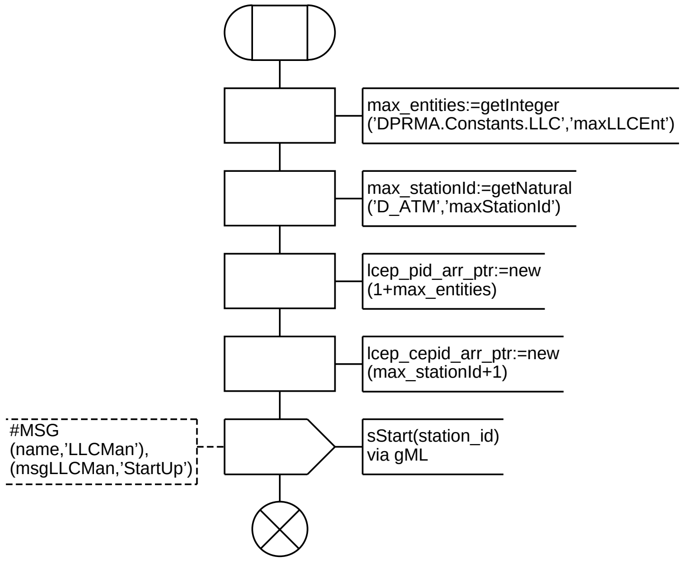


Figure 5.5: W-CHAMB SDL procedure

Assuming a free space propagation, The signal energy arriving at a receiver is

$$P_r = P_t G_t G_r (\lambda/4\pi) 1/d^2 \tag{5.1}$$

where P_t represents the power radiated by the transmitter and P_r the input power of the receiver. G_t and G_r stand for the corresponding absolute antenna gains. λ is the wavelength and d the distance between sender and receiver, that is decided by the location management, see Section 5.4.

In a realistic environment, the signal power decreases much faster than with free-space propagation. With the introduction of the propagation coefficient γ, the signal energy arriving at the receiver becomes

$$P_r = P_t G_t G_r (\lambda/4\pi) 1/d^\gamma \tag{5.2}$$

Realistic value for γ are two in free space, between three and four in radio applications within cities, and perhaps greater than four inside buildings.

In the simulator the signal power at distance d from the transmitter is calculated as

$$P_r = kd^{-\gamma} \tag{5.3}$$

where k is the same constant for all WSs.

The operation of wireless networks like W-CHAMB and IEEE 802.11 requires a WS to detect the on-going transmission to decide the status of the transmission channel. If the sum of all signal powers at the detection time point is larger than a detection level, a transmission signal is detected. The detection level is set by the protocol and usually much higher than the background noise level to distinguish the signals from the noise. The detection level is, however, much lower than decode level. The higher the transmission data rate, the higher the decode level must be. The minimum difference of the decode level is decided by the necessary (Carrier-to-Interference) C/I ratio to decode a data packet correctly. The C/I is the ratio of the received carrier signal C to the received interference I. On the assumption that all WSs transmit independently of one another, the total interference power at the receiver with N interfering WSs can be calculated on the basis of a combination of the individual interference contribution I_k and the noise power N:

$$I = \sum_{all\ k} I_k + N \tag{5.4}$$

The packet error probability is decided by C/I, the modulation scheme and the packet length. Figure 5.6 shows the relationship of the packet error with the OFDM modulation scheme and C/I [69]. A packet with higher C/I has a lower packet error probability. With the same C/I, a packet with higher data rate has a higher packet error probability. The simulation results shown in Figure 5.6 is based on the packet length of 54 bytes. We use the results in Figure 5.6 as a reference to derive the packet error (P_e) of a packet with length L according to Equation (5.5).

$$P_e = 1 - (1 - p)^{\frac{L}{54}} \tag{5.5}$$

where p is the value read from Figure 5.6 with a packet length of 54 bytes.

For the packet-oriented transmission like CSMA/CA with IEEE 802.11, a receiver may experience a different C/I of the same data packet while receiving. Figure 5.7 shows, as an example, another WS sends a short

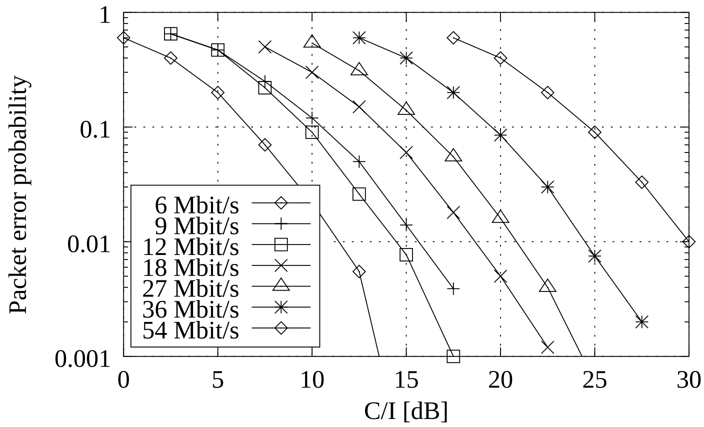


Figure 5.6: Packet error vs. C/I

packet during the transmission of a long packet. At this case, the receiver decodes the data packet at the three different time periods t_1, t_2 and t_3 separately, based on the actual C/I and data length. If one or more of the three parts of the data packet cannot be decoded correctly, the whole packet is discarded.

5.4 Location Management

Section 5.3 indicates that the signal strength reaching the receiver is dependent on the distance d between the sender and the receiver. The distance d is decided by the location management model according to the position of the wireless stations.

$$d = \sqrt{(x_1 - x_2)^2 + (y_1 - y_2)^2 + (z_1 - z_2)^2} \tag{5.6}$$

where (x_1, y_1, z_1), (x_2, y_2, z_2) are positions of the sender and the receiver, respectively.

The position of the all wireless stations are managed by the location management model. The positions of the WSs may change during the simulation

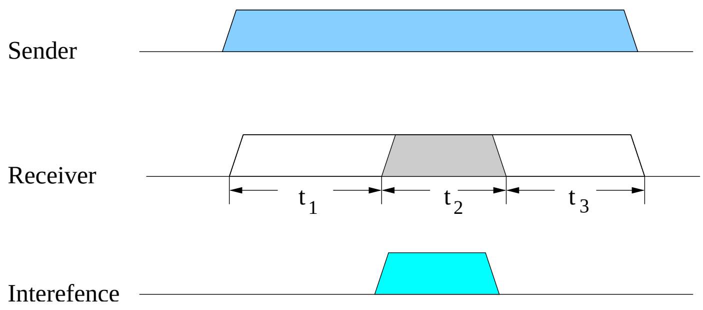
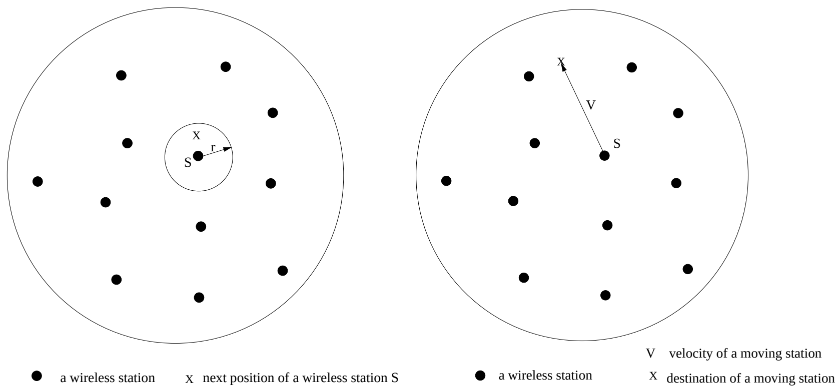


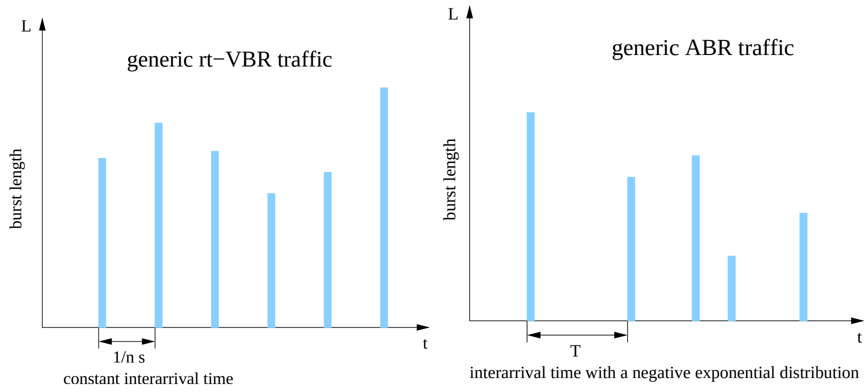
Figure 5.7: C/I at the packet-oriented transmission



(a) Random model **(b)** Target-oriented model

Figure 5.8: Mobility Model

due to the mobility of the WSs. Two kinds of mobility model are implemented in the simulation tool. One is called random mobility. With the random mobility, a mobile WS selects a random position in a small circle with radius r after a time interval $\triangle t$, see Figure 5.8(a). The other one is called target-oriented mobility. In this case, a mobile selects a predetermined target position and moves toward the target position with a speed V, see Figure 5.8(b). Usually the target-oriented mobility is much more realistic than the random mobility.



(a) rt-VBR traffic model

(b) ABR traffic model

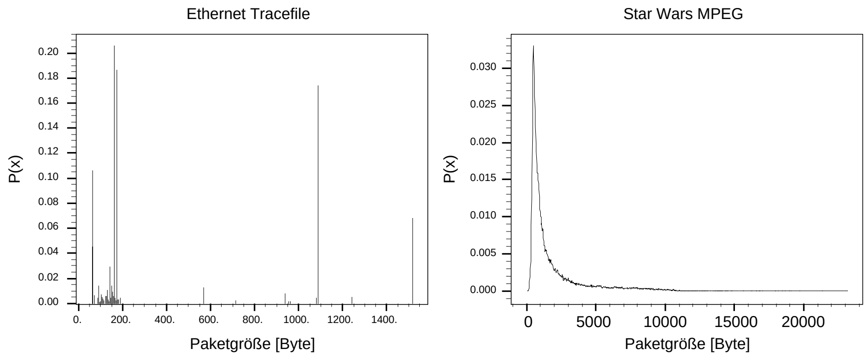
Figure 5.9: The generic traffic source model for rt-VBR and ABR services

5.5 Traffic Sources

To evaluate the traffic performance of the wireless networks, two kinds of traffic sources, stochastic models and trace files, are used in the simulation tool. Stochastic models are used to simulate the generic traffic sources. To compare the traffic performance of W-CHAMB with IEEE 802.11 and HiperLAN/2, trace files are used to make a fare comparison of the traffic performance possible.

5.5.1 Stochastic traffic source models

Generic real time bursty (rt-VBR) traffic is used to model a video codec which produces n pictures per second, see Figure 5.9(a). n is a variable that may vary from 8 to 30 according to the quality of the video pictures. The burst length of rt-VBR traffic is modeled by an autoregressive Markovian process [82] with a mean of m. m is varied to model the different data rates of the video streams. The rt-VBR traffic has two QoS requirements: maximum delay and packet dropping probability.



(a) Ethernet **(b)** MPEG

Figure 5.10: Packet size distribution read from trace files

Delay insensitive (ABR) traffic is used to model the generic data traffic. The burst length of the ABR traffic is geometrically distributed with a mean of γ, see Figure 5.9(b). The interarrival time of ABR bursts is negative-exponentially distributed, with a mean value a. γ and a are changed to model the different ABR traffic characteristics.

5.5.2 Trace files

For the purposes of performance comparison of different wireless networks, the trace files that contain the real measured data are used as the traffic sources. Figure 5.10 shows the packets length distribution of the Ethernet trace and the MPEG video trace files used [13, 46].

The Ethernet traffic is ASCII-format tracing data, consisting of one 20-byte line per Ethernet packet arrival. Each line contains a floating- point time stamp (representing the time in seconds since the start of a trace) and an integer length (representing the Ethernet data length in bytes). The length field does not include the Ethernet preamble, header, or CRC; however, the Ethernet protocol forces all packets to have at least the minimum size of 64 bytes and at most the maximum size of 1518 bytes. 99.5 % of the encapsulated packets carried by the Ethernet PDUs were IP.

The MPEG trace file represents the bandwidth output of a variable bit rate (VBR) video coder which conforms to the MPEG 1 standard. The sequence of MPEG I, P and B frames used is IBBPBBPBBPBB IBB..., so there are 12 frames in a Group of Pictures (GOP). The data file consists of 174,138 integers (in plain ASCII text), representing the number of bits per video frame. The source material used here is the movie "Star Wars", which contains quite a diverse mixture of material ranging from low complexity/motion scenes to those with very high action.

To change the data rates of the traffic sources, the time stamp is modified by a factor to speed up or slow down the traffic stream.

5.6 Configuration Parameters and Performance Evaluation

To configure a simulation, all necessary parameters, such as the number of WSs, transmission power, etc. are stored in a read-in file. The parameters are read by the program during the simulation run. By doing so, the program code need not to be recompiled if some parameters are needed to be changed. This is very important for the protocol design to get the insights of the traffic performance with different parameter sets.

The delay performance is evaluated by the Discrete Limited Relative Error (DLRE) algorithm [49, 100]. The relative error is set to 1%. The throughput performance is obtained directly form the measured data during the simulation.

5.7 Implementation Aspect

The simulation tool is based on the object oriented C++ programming language [64]. Some functions which are necessary for the stochastic simulation are already available in the C++ class library SPEETCL. These are, e.g. event-driven simulation scheduler, random generator, LRE algorithms. The

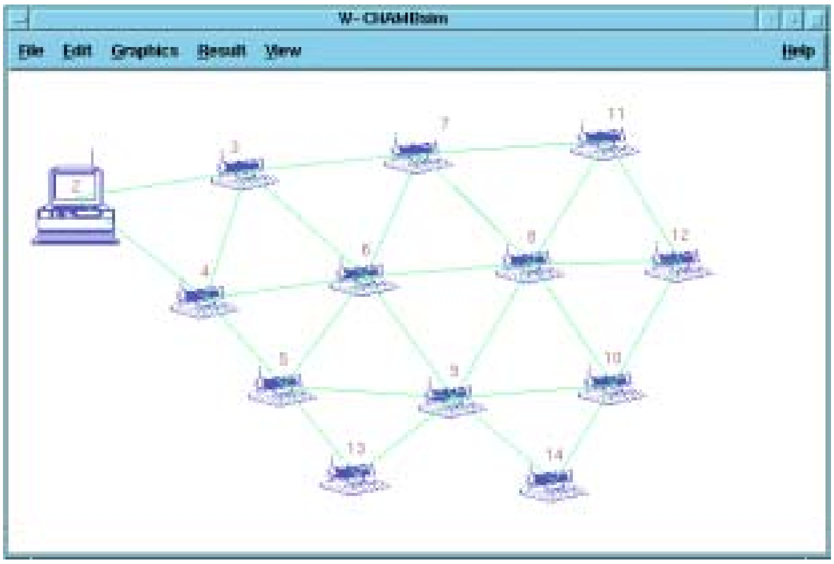


Figure 5.11: GUI of the W-CHAMB simulation tool

SDL system that contains the complete protocol specification is translated into SPEETCL C++ classes with the help of SDL2SPEETCL.

For the project management of the program codes, the commercial available software tool SNiFF+ is used. Using the makefiles automatically generated by the SNiFF++ tool, the software codes can be easily compiled and debugged.

For the easy configuration of the simulation parameters and display of simulation results, a graphical user interface (GUI) is developed using the TCL/TK script language [90], see Figure 5.11.

CHAPTER 6

W-CHAMB Traffic Performance

In this chapter the W-CHAMB traffic performance is studied using analytical methods and the simulation tool. Using probability theory, queuing theory and Markov theory we can obtain insights of the W-CHAMB system performance. The advantage using the analytical method is that an intuitive and elegant explanation of the system behavior can be derived quickly. The limitation of the analytical evaluation is that the system has to be simplified to make the analyzes mathematically tractable. For this reason, the simulation tool described in Chapter 5 is used to study W-CHAMB systems with more complicated scenarios. The impact of the system parameters on the traffic performance is intensively studied and more insights into the W-CHAMB networks are revealed.

6.1 Analytical Evaluation

In this section, analytical methods are used to evaluate the W-CHAMB air interface. We analyze the Access Success Probability (ASP) on the ACH using probability theory. The ASP when a WS randomly chooses an access priority is compared with that when a WS selects it according to a geometric distribution. When a WS selects an access priority according to a geometric distribution, the parameter p of the geometric distribution has a significant impact on the ASP. With the results on the ASP, the performance of the W-CHAMB MAC protocol is analyzed using the two-dimensional Markov chain theory.

6.1.1 The Access Success Probability (ASP)

The W-CHAMB MAC protocol uses a prioritized access scheme to provision QoS and to reduce the collision probability of the random access on the ACH. The WS contending for the access on the ACH selects an access priority according to the algorithms specified in Section 4.3.2.4. The WS with highest priority survives from the contending process and obtains the right to access the ACH. If more than one WS has the highest priority, a collision happens. In this case, the access has failed.

Assume that n WSs are contending for the transmission on the ACH and each WS chooses a priority between 0 and P_{max} independently, the probability, that only one WS has the maximum access probability can be calculated according to the Equation (6.1). $P(s=1|n)$ is the probability that one of the n contending WSs has chosen the access priority larger than the other $n-1$ WSs. This WS survives from the access contention and transmits the access PDU without collision. So $P(s=1|n)$ is the ASP with n contending WSs on the ACH.

$$P(s=1|n)=\begin{cases} n\cdot\sum_{i=1}^{Pmax} P(l=i)\cdot(P(l<i))^{n-1} & n\geq 2 \\ 1 & n=1 \end{cases} \tag{6.1}$$

where $P(l=i)$ is the probability that an access priority i is chosen by a WS. $P(l<i)$ is the probability that a WS chooses an access priority smaller than i. $P(l=i)\cdot(P(l<i))^{n-1}$ is the probability that one WS chooses a priority i and all other n-1 WSs choose a priority smaller than priority i. This is the case that one WS survives from the contention on the ACH. The sum of all such possible cases results in $P(s=1|n)$ that is the ASP with n contending WSs.

The values of $P(l=i)$ and $P(l<i)$ depend on the way how a WS chooses the access priority. If all WSs choose the access priority randomly, the probability that a WS chooses one of the values between 0 and P_{max} is uniformly distributed. In that case, $P(l=i)$ and $P(l<i)$ are calculated according to Equation (6.2) and Equation (6.3), respectively.

$$P(l=i)=\frac{1}{P_{max}+1} \qquad 0\leq i\leq P_{max} \tag{6.2}$$

$$P(l<i)=\frac{i}{P_{max}+1} \qquad 0\leq i\leq P_{max} \tag{6.3}$$

If the probability that a WS selects the priority i is geometrically distributed, $P(l < i)$ is then calculated according to Equation (6.4) and Equation (6.5), respectively. p in Equation (6.4) is the parameter of geometric distribution. A larger p means that the probability to select a larger access priority becomes higher.

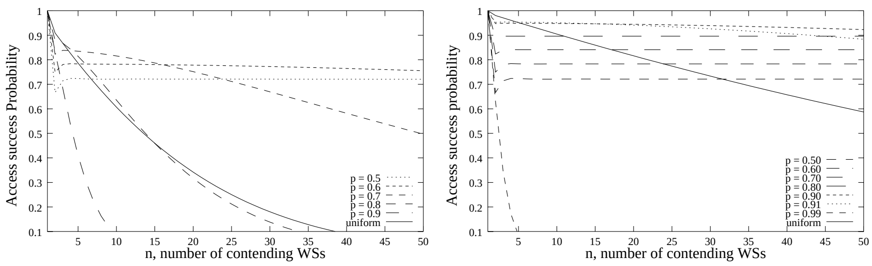
$$P(l = i) = \begin{cases} p^i \cdot (1-p) & 0 \leq i < P_{max} - 1 \\ p^{P_{max}} & i = P_{max} \end{cases} \tag{6.4}$$

$$P(l < i) = \quad 1 - p^i \qquad 0 \leq i \leq P_{max} \tag{6.5}$$

Figure 6.1 and Figure 6.2 show the ASP vs. the number of contending WSs under various parameters. The ASP is calculated according to the Equation (6.1, 6.2, 6.3, 6.4 and 6.5).

Figure 6.1(a) shows that if a WS randomly selects an access priority between 0 and 10, the ASP decreases rapidly with the increase of the number of contending WSs. If an access priority is chosen between 0 and 10 according to a geometric distribution, the parameter p of the geometric distribution has a significant impact on the access success probability. For a small number of contending WSs (fewer than 15), the highest ASP is achieved with p = 0.7. But the ASP decreases significantly if the number of contending WSs increases. When the number of contending WSs is larger than 15, a stable access performance is obtained with p = 0.6 in this case. With $P_{max} = 10$, the ASP degrades quickly with a moderate increase of the number of contending WSs if a WS chooses an access priority randomly. From the results in 6.1(a) we derive the conclusion that an access priority should be chosen according to a geometric distribution if the maximum access priority is small and the number of contending WSs is moderate to large.

Figure 6.1(b) shows the ASP vs. the number of contending WSs with the maximum access priority $P_{max} = 50$. Here we see that the ASP with $P_{max} = 50$ is much higher than the ASP with $P_{max} = 10$ which is displayed in Figure 6.1(a). With $P_{max} = 50$, the best access performance can be achieved with p = 0.90 if a WS chooses a priority between 0 and 50 according to a geometric distribution. With a small number of contending WSs (fewer than 5), the better access performance is achieved when a WS chooses an access priority randomly. In this case, if the number of contending WSs is generally fewer



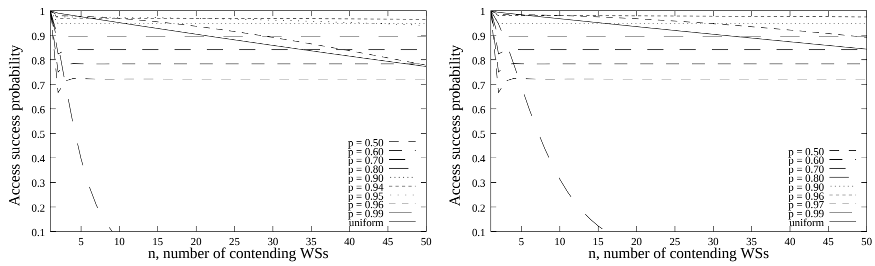
(a) $P_{max} = 10$ **(b)** $P_{max} = 50$

Figure 6.1: Access success probability vs. the number of contending WSs with $P_{max} = 10$ and 50

than 5, a WS should choose an access priority randomly. Otherwise the access priority should be chosen according to the geometric distribution with the parameter p = 0.9.

Figure 6.2 shows the ASP vs. the number of contending WSs with the maximum priority of 100 and 150. The highest ASP is obtained with p = 0.94 and p = 0.96 respectively, when an access priority is chosen according to the geometric distribution. The results indicate that the parameter p has large impact on the ASP and p has different optimized values with different P_{max}. With a small number of contending WSs (fewer than 5), the maximum ASP is almost the same if the access priority is chosen randomly or according to a geometric distribution. If the number of the contending WSs is large, the superiority to choose an access priority according to a geometric distribution is obvious.

Figure 6.3 compares the ASP under various P_{max}. It is natural that the ASP can be improved if the maximum access priority is larger. However, with a maximum access priority more than 100, the access performance will not be improved significantly. This means that $P_{max} = 100$ is enough in most cases. From the results in Figure 6.3, we can also conclude that when the number of contending WSs is larger than 5, a better access performance is achieved by choosing the access priority according to a geometric distribution, whilst the access performance is better by choosing the access



(a) $P_{max} = 100$ **(b)** $P_{max} = 150$

Figure 6.2: Access success probability vs. the number of contending WSs with P_{max} of 100 and 150

priority randomly when the number of contending WSs is fewer than 5. For networks with best effort services that experience usually a large number of contending WSs, the access priority should be chosen according to a geometric distribution because the access performance remains almost independent on the number of contending WSs if the access priority is chosen according to the geometric distribution. The parameter p of the geometric distribution should be optimized to achieve the best access performance. On the contrary, the number of contending WSs with real time traffic services tends to be rarely larger than 5 as the traffic loads of real time services are strongly controlled by the call admission control. Thus, the access priority for the real time services should be selected randomly.

6.1.2 Markov Analysis of the W-CHAMB MAC Protocol

6.1.2.1 Model Formulation

Consider a W-CHAMB network with M WSs, each being able to communicate directly with any of the other WSs. A traffic source in each WS generates PDU trains. The number of PDUs of each PDU train is geometrically distributed with a mean of $\frac{1}{1-p}$. A new PDU train is generated only if the previous PDU train is fully transmitted. This assumption is called a single

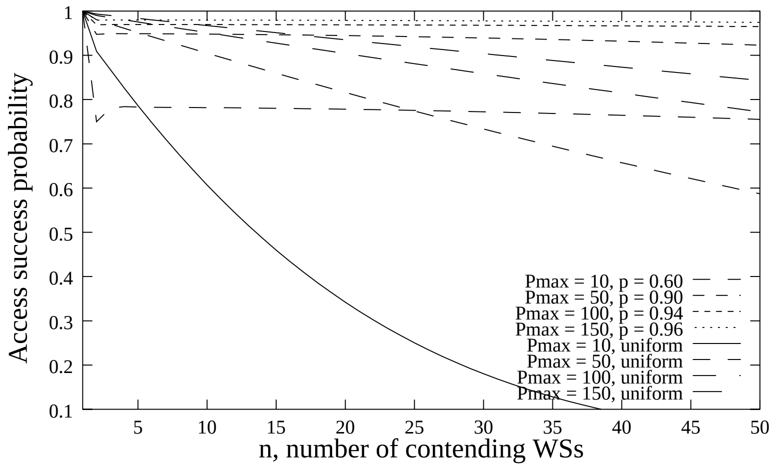


Figure 6.3: The access success probability vs. the number of contending WSs under various maximum access priority

message traffic source [74]. The time interval between the end of previous PDU train and the arrival of a new PDU train is negative-exponentially distributed with a rate of λ.

Upon the arrival of the first PDU of a PDU train, the WS starts to reserve an RCC according to the procedures described in Section 4.3.2.5. For the reason of mathematical tractability, it is assumed that only one TCH (TDMA slot) can be reserved for each PDU train. The channel is assumed error free except for collisions. Each W-CHAMB MAC frame has one ACH and N TCH. If there is any TCH free and one WS successfully transmits its *acc s-PDU* on the ACH, the addressed WS will reply with an ACK PDU on the selected TCH. This TCH is then reserved and the WS begins to transmit its Data PDU on this reserved TCH from the next MAC frame on. At the end of the PDU train, the reserved TCH is released to be available to other WSs again.

The W-CHAMB MAC protocol can be modeled as two queue systems in tandem as shown in Figure 6.4. The first queue system is a distributed queue and it deals with *acc s-PDU*. The ACH is modeled as an ACH-server

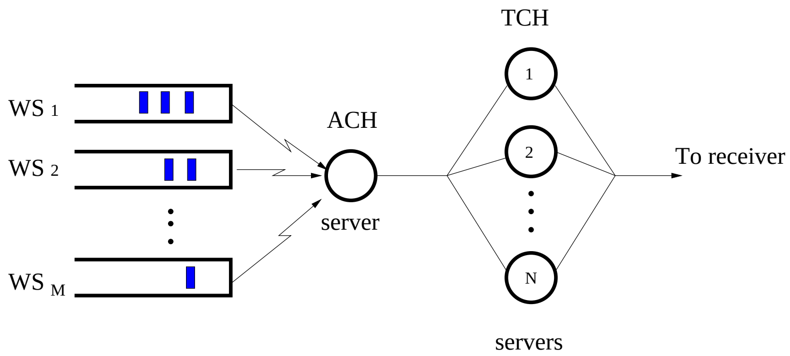


Figure 6.4: W-CHAMB tandem queuing model

to process *acc s-PDU*. The maximum service rate of the ACH-server is one *acc s-PDU* per MAC frame. The service rate of the ACH-server is dependent on the access success probability (ASP) on the ACH. *acc s-PDUs* are queued in each WS that has a PDU train to transport, but has not yet reserved a TCH. The second queuing system is a multi-server system. Each TCH is modeled as a TCH-server to process PDU trains. The service rate of each TCH-server is one PDU per MAC frame. The remaining PDUs of a PDU train that is being transported on a reserved TCH is queued in the WS. The queuing discipline is FIFO (First In First Out) and the capacity of the buffer in each WS is assumed large enough to ensure that no PDU of a PDU train is dropped. There is a blocking mechanism in such a tandem queuing model as the ACH-server stops working whenever all TCH servers are full. This is called communication blocking [124]. This means that the ratio of the number of TCHs in each MAC frame (N) and the mean length of the PDU train ($\frac{1}{1-p}$) should be appropriate. Otherwise, there is a bottleneck effect on the ACH.

The summary of the Model is as follows:

1. A fully connected W-CHAMB network with M WSs is considered. An error-free channel is assumed.

2. Each MAC frame has one ACH and N TCHs.

3. The length of a PDU train (number of PDUs) is geometrically distributed ($P(l) = p^{l-1}(1-p)$) with a mean length $E(L) = \frac{1}{1-p}$. It

is assumed that all PDUs of a PDU train are generated at the same time.

4. Only one TCH can be reserved for a PDU train.
5. The time interval between the end of the previous PDU train and the arrival of a new PDU train is negative-exponentially distributed with a rate of λ.

6.1.2.2 The Analysis

To analyze the W-CHAMB tandem queuing model shown in Figure 6.4, the two dimensional Markov chain analysis method [33, 104, 105] is used. The W-CHAMB MAC system at the start of the nth frame (embedding points) is observed and the system is modeled as a two dimensional discrete time Markov chain process:

$$X = \{X_n = (C_n, R_n) | n >= 0\} \tag{6.6}$$

where X_n is a two-dimensional random sequence. C_n represents the number of WSs that have a PDU train waiting for transmission at the start of the nth frame. These WSs contend for the access on the ACH to reserve a TCH. R_n represents the number of TCHs that are reserved for the transmission of a PDU train at the start of the nth frame.

The steady-state of the system is denoted by the pair (c, r), where c denotes the number of the WSs contending for the transmission on the ACH, and r denotes the number of TCHs reserved for transmission. The values of c and r meet that $0 \leq c \leq M$, $0 \leq r \leq N$ and $0 \leq c + r \leq M$. All possible states are shown in Figure 6.5. The one-step state transition probabilities must be calculated to analyze a Markov chain. It is too complicated to display all the state transitions in Figure 6.5. To have an intuitive image of the state transitions of the W-CHAMB tandem queuing model, we use a much simplified case. The state transition diagram with M=3 and N=2 is displayed in Figure 6.6. For the reason of easy description we define an active WS as a WS that has a PDU train to transmit, whilst an idle WS is a WS that has no PDU to transmit. State (0, 0) means that no WS waits to access the ACH and there is no reservation on TCHs. The possible states to which state (0, 0) can transit are (1, 0), (2, 0) and (3, 0) if one, two or

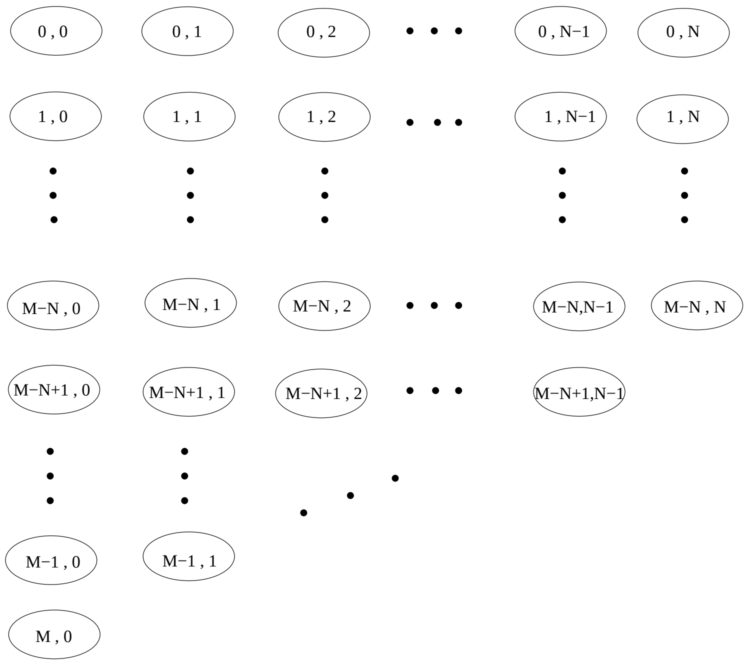


Figure 6.5: State diagram of the W-CHAMB Markov model

three WSs become active during the next MAC frame, respectively. State (0, 0) remains unchanged if no WS becomes active during the next MAC frame. State (0, 1) means one TCH is reserved by a WS and no WS waits for accessing the ACH. State (0, 1) may transit to state (1, 1) or (2, 1) if one or two idle WSs become active and the reserved TCH is not released during the next MAC frame. It is also possible for state (0, 1) to transit to state (1, 0) or (2, 0) if one or two idle WSs become active and the reserved TCH is released during the next MAC frame. State (0, 1) transits to state (0, 0) if the reserved TCH is released and no PDU train arrives in the remaining two WSs. State (0, 1) may also remain unchanged if no idle WS becomes active and the reserved TCH is not released. For the similar reason, state (0, 2) may transit to (0, 0), (0, 1) (1, 0), (1, 1) and (1, 2) , or remain unchanged. State (1, 0) can transit to State (0, 1), (1, 1), (2, 1), but is not possible to remain unchanged as only one WS waits to access the ACH. All other state transitions are shown in the Figure 6.6. From Figure 6.6 we see that all states are positive-recurrent, irreducible and aperiodic so that the two dimensional Markov chain of W-CHAMB system is ergodic.

The one-step transition probability (P_t) of the system

$$P_t = P(c_2, r_2|c_1, r_1) = \lim_{n\to\infty} P(C_{n+1} = c_2, R_{n+1} = r_2|C_n = c_1, R_n = r_1) \tag{6.7}$$

can be calculated according to the Equation (6.8).

$$P_t = \begin{cases} 0 & c_2 < c_1 - 1 \\ 0 & r_2 > r_1 + 1 \\ 0 & c_2 < c_1,\ \ r_2 = 0 \\ [1 - P(s = 1|c_1)] \cdot (1-p)^{r_1} & \\ \quad \cdot P(a = c_2 - c_1 | I = M - c_1 - r_1) & c_2 \geq c_1,\ \ r_2 = 0 \\ P(s = 1|c_1) \cdot p^{r_1} & \\ \quad \cdot P(a = c_2 - c_1 + 1 | I = M - c_1 - r_1) & r_2 = r_1 + 1 \\ P(s = 1|c_1) & \\ \quad \cdot \binom{r_1}{r_1 - r_2 + 1} \cdot p^{r_2 - 1} \cdot (1-p)^{r_1 - r_2 + 1} & \\ \quad \cdot P(a = 0 | I = M - c_1 - r_1) & c_2 = c_1 - 1,\ \ r_2 \leq r_1 \\ P(s = 1|c_1) & \\ \quad \cdot \binom{r_1}{r_1 - r_2 + 1} \cdot p^{r_2 - 1} \cdot (1-p)^{r_1 - r_2 + 1} & \\ \quad \cdot P(a = c_2 - c_1 + 1 / I = M - c_1 - r_1) & \\ \quad + \ [1 - P(s = 1|c_1)] & \\ \quad \cdot \binom{r_1}{r_1 - r_2} \cdot p^{r_2} \cdot (1-p)^{r_1 - r_2} & \\ \quad \cdot P(a = c_2 - c_1 | I = M - c_1 - r_1) & c_2 \geq c_1,\ \ r_2 \leq r_1 \end{cases} \tag{6.8}$$

where $P(s = 1|c_1)$ is the ASP with c_1 contending WSs, calculated according to Equation (6.1). $P(a = x|I = y)$ is the probability that x WSs from y idle WSs become active during the next MAC frame. As the time interval between the end of previous PDU train and the arrival of a new PDU train is negative-exponentially distributed with a rate of λ, $P(a = x|I = y)$ can be calculated according to the Poisson distribution, see Equation (6.9).

$$P(a = x|I = y) = \frac{(y \cdot \lambda \cdot T_f)^x}{x!} e^{-y \cdot \lambda \cdot T_f} \tag{6.9}$$

T_f is the MAC frame duration. $y \cdot \lambda$ is the total PDU train arrival rate of y idle WSs.

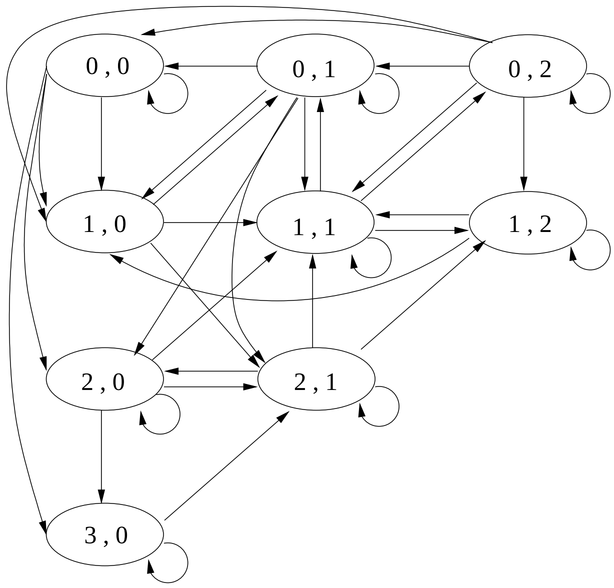


Figure 6.6: State transition diagram (M=3, N=2)

The Equation (6.8) can be explained as follows:

1. $c_2 < c_1 - 1$ is impossible for the state transition as each MAC frame is able to serve at maximum one *acc s-PDU*. The possible state transitions for c_2 are $c_2 = c_1$ (if no *acc s-PDU* is served) or $c_2 = c_1 - 1$ (if one *acc s-PDU* is served).

2. $r_2 > r_1 + 1$ is also impossible for the state transition as the number of reserved TCHs can increase by one at most during one MAC frame because at maximum one *acc s-PDU* can be processed in each MAC frame.

3. if $c_2 < c_1$, then $r_2 = 0$ is impossible as $c_2 < c_1$ means one *acc s-PDU* is served during the MAC frame so that $r_2 \geq 1$ (at least one TCH is reserved).

4. $c_2 \geq c_1$ and $r_2 = 0$ mean that no *acc s-PDU* is served and r_1 TCHs are released during the MAC frame. $[1 - P(s = 1|c_1)]$ is the probability

that no *acc s-PDU* is served with the c_1 contending WSs. $(1-p)^{r_1}$ is the probability that r_1 TCHs are released during one MAC frame. $P(a = c_2 - c_1 | I = M - c_1 - r_1)$ is the probability that a WSs from the I idle WSs becomes active during the MAC frame. The product of these three probability results in the transition probability from state (c_1, r_1) to (c_2, r_2).

5. $r_2 = r_1 + 1$ means that one *acc s-PDU* is served and all the r_1 TCHs remain reserved. $P(s = 1|c_1)$ is the probability that one *acc s-PDU* is served among the c_1 contending WSs. p^{r_1} is the probability that all the r_1 TCHs remain reserved. The transition from c_1 to c_2 means that $c_2 - c_1 + 1$ WSs from the $(M - c_1 - r_1)$ idle WSs should become active during one MAC frame.

6. $c_2 = c_1 - 1 \quad and \quad r_2 \leq r_1$ mean that one *acc s-PDU* is served, the $r_1 - r_2 + 1$ TCHs are released during the MAC frame, and no WSs from the $(M - c_1 - r_1)$ idle WSs become active. The state transition probability in this case is the product of these three probabilities. $\binom{r_1}{r_1 - r_2 + 1} \cdot p^{r_2 - 1} \cdot (1-p)^{r_1 - r_2 + 1}$ is the probability that $r_1 - r_2 + 1$ of the r_1 reserved TCHs are released during one MAC frame.

7. if $c_2 \geq c_1 \quad and \quad r_2 \leq r_1$, then one *acc s-PDU* may or may not be served. The probability that one *acc s-PDU* is served is $P(s = 1|c_1)$, whereas the probability that no *acc s-PDU* is served is $1 - P(s = 1|c_1)$. If one *acc s-PDU* is served, then $r_1 - r_2 + 1$ TCHs of the r_1 reserved TCHs are released and $c_2 - c_1 + 1$ WSs of the $M - c_1 - r_1$ idle WSs become active during the MAC frame so that state (c_1, r_1) transits to (c_2, r_2). In the case that no *acc s-PDU* is served, then $r_1 - r_2$ TCHs of the r_1 reserved TCHs are released and $c_2 - c_1$ WSs of the $M - c_1 - r_1$ idle WSs become active during the MAC frame.

The steady-state probabilities P(c, r) can be obtained by solving a set of well-known liner equations 6.10 [104].

$$P(c,r) = \sum_{all(i,j)} P(i,j) \cdot P(c,r|i,j), \qquad \sum_{all\ \ (c,r)} P(c,r) = 1 \tag{6.10}$$

where (c, r) and (i, j) denote all possible states of the Markov chain, see Figure 6.5.

From the steady-state probability P(c, r) the stationary distribution P(c) and P(r) for the variables $C = \lim_{n\to\infty} C_n$ and $R = \lim_{n\to\infty} R_n$ can be calculated:

$$P(c) = \sum_{all \ \ r} P(c, r) \qquad r = 0, 1...N \tag{6.11}$$

$$P(r) = \sum_{all \ \ c} P(c, r) \qquad c = 0, 1...M \tag{6.12}$$

as well as the expected values E(C) and E(R):

$$E(C) = \sum_{all \ \ c} P(c) \cdot c \qquad c = 0, 1...M \tag{6.13}$$

$$E(R) = \sum_{all \ \ r} P(r) \cdot r \qquad r = 0, 1...N \tag{6.14}$$

From the expected values E(C) and E(R) the W-CHAMB MAC protocol performance, i.e. the system throughput and the mean delay can be derived. The system throughput is defined as the ratio of the average number of transmitted PDUs per frame to the number of TCHs each MAC frame. Assume that the ACH has the same length as TCH, the system throughput η can be calculated according to the Equation (6.15).

$$\eta = \frac{E(R)}{N+1} \tag{6.15}$$

We define the access delay W as the expected value of the time to obtain a reservation, equal to the average interval of time a WS spends in contention. W is the mean queuing time of *acc s-PDU* in the first queuing system in Figure 6.4.

The average access delay (W) can be calculated according to Little's law [71, 72]:

$$W = \frac{E(C)}{(M - E(C) - E(R)) \cdot \lambda} \tag{6.16}$$

where $(M - E(C) - E(R)) \cdot \lambda$ represents the mean arrival rate of *acc s-PDUs*. E(C), the mean number of the contending WSs, corresponds to the mean queue length of the first queuing system.

The mean delay of the PDU train is the average time spent in the system, that is the access delay plus the service time. As one PDU train can reserve at maximum one TCH, the average service time is equal to the average number of PDUs of PDU trains multiplied by T_f, the MAC frame duration. For an individual PDU in a PDU train, the delay of the first PDU is one MAC frame plus the access delay, whereas the last PDU experiences a delay of L MAC frames plus the access delay. L is the PDU train length, that is a random variable with a geometric distribution. The mean PDU delay of a PDU train with l PDUs is $\frac{l+1}{2} \cdot T_f$ plus the access delay. The mean PDU delay T of all PDU trains can be calculated as the ratio of the expected value of the sum of the PDU delay of a PDU train to the mean PDU train length, see Equation (6.17).

$$T = W + \frac{E(\frac{L+1}{2} \cdot T_f \cdot L)}{E(L)} \tag{6.17}$$

As L is geometrically distributed, $P(L = l) = p^{l-1}(1-p)$ we have

$$E(L) = \frac{1}{1-p} \tag{6.18}$$

$$E(L^2) = \frac{1+p}{(1-p)^2} \tag{6.19}$$

Then Equation (6.17) can be expressed as

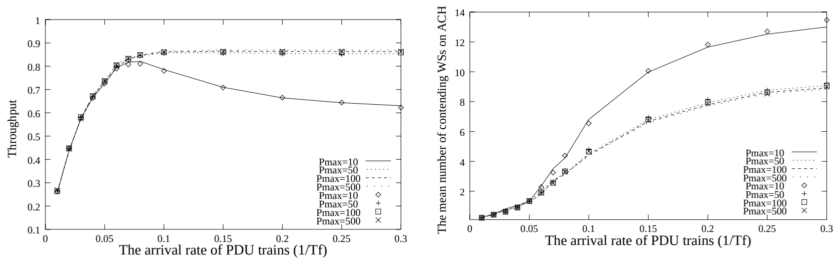
$$\begin{aligned} T &= W + \frac{E(\frac{L^2+1}{2})}{E(L)} \cdot T_f \\ &= W + \frac{E(L^2) + E(L)}{2E(L)} \cdot T_f \\ &= W + (\frac{E(L^2)}{2E(L)} + \frac{1}{2}) \cdot T_f \end{aligned}$$

$$\begin{aligned}
&= W + (\frac{(1+p)}{2(1-p)} + \frac{1}{2}) \cdot T_f \\
&= W + (\frac{1}{1-p}) \cdot T_f \\
&= W + E(L) \cdot T_f
\end{aligned} \tag{6.20}$$

where E(L) is the mean length of the PDU trains. It is interesting to see that the mean PDU delay of all PDU trains (T) is different from the value of $W + \frac{E(L)+1}{2} \cdot T_f$. The reason for this interesting result is that, as indicated in Equation (6.17), longer PDU trains have more PDUs with larger PDU delays than shorter ones. The result is confirmed by the simulation in Figure 6.10(b).

6.1.2.3 Numerical Results

The numerical results of the throughput and mean delay of the W-CHAMB MAC protocol are discussed in this section. If not especially indicated, the parameters M (number of WSs) and N (number of TCHs per MAC frame) are $M = 25$ and $N = 15$, respectively. The access priority is selected randomly by a contending WS. The linear equation system of Equation (6.10) is solved using the BCG (biconjugate gradient) algorithm [104]. By solving Equation (6.10), we obtain all equilibrium state probabilities P(c, r), c = 0...M, r = 0...N, $c + r \leq M$. The distribution of the number of contending WSs and the number of reserved TCHs can be derived from Equation (6.11) and (6.12), respectively. The expected values of the number of contending WSs on ACH and the number of reserved TCHs can be easily calculated according to the Equation (6.13) and (6.14). The throughput and delay performance can finally be derived from Equation (6.15) and (6.20). The accuracy of the analytical results is verified by comparison with simulation results. Simulation results are obtained by means of the stochastic simulation tool described in Chapter 5. Simulation results are depicted with points, whereas the analytical results are plotted by lines.



(a) Throughput **(b)** mean M

Figure 6.7: Impact of access priority

6.1.2.4 Impact of the Maximum Access Priority

Figure 6.7 shows the impact of maximum access priority on the throughput of a W-CHAMB network with 25 WSs. The mean length of PDU trains is set to 20. From Figure 6.7(a) we can see that the maximum access priority $Pmax = 10$ appears not enough in this scenario as the throughput degrades significantly when the network approaches saturation due to the low ASP. On the other hand, very large maximum access priority is also unnecessary as the throughput does not increase much if the maximum access priority is more than 50 as the ASP is high enough with the maximum access priority of 50 in this case. The maximum achievable throughput is about 0.86. Figure 6.7(b) indicates the mean number of the contending WSs vs. the PDU train arrival rates. The increase of the arrival rate increases the number of contending WSs, which decreases the ASP if the maximum access priority is not large enough, see Figure 6.1(a). Figure 6.7(b) shows that the mean number of contending WSs with maximum access priority of 10 is significantly larger than that with a maximum access priority more than 50 due to the lower ASP with $P_{max} = 10$. The analytical results agree with the simulation results very well in Figure 6.7 as the Markov model is in a good equilibrium at this condition.

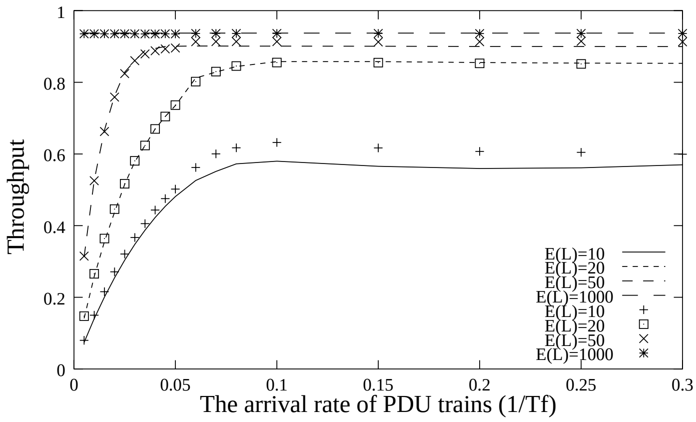


Figure 6.8: Impact of PDU train length on throughput

6.1.2.5 Impact of the Mean Length of PDU Trains

Figure 6.8 shows that the maximum throughput of the system depends on the mean length of PDU trains. If the mean length of the PDU trains (E(L)) is equal to 1000, then the network works like a pure TDMA system. The throughput is close to the maximum achievable throughput of the system, that is $\frac{15}{1+N} = 0.9375$. With a reduced mean length of PDU trains, the maximum throughput decreases. This is due to the bottleneck effect of the ACH. Before a WS can reserve a new TCH, its *acc s-PDU* must be served first by the ACH. As each MAC frame can serve only one *acc s-PDU* at the most, some WSs may not be able to reserve a TCH although there are free TCHs. The bottleneck effect become worse if the mean length of PDU trains decreases. From Figure 6.8 we can see that the bottleneck effect is insignificant for the system with N = 15 if the mean length of the PDU train is larger than 50. But the bottleneck effect becomes very significant with smaller mean length of the PDU trains. With E(L)=10, the maximum achievable throughput is only about 0.6. It seems that for a system to support short PDU trains a short MAC frame is preferred to reduce the bottleneck effect of the ACH. But a short MAC frame results in a large protocol overhead caused by the ACH. The trade-off between the protocol

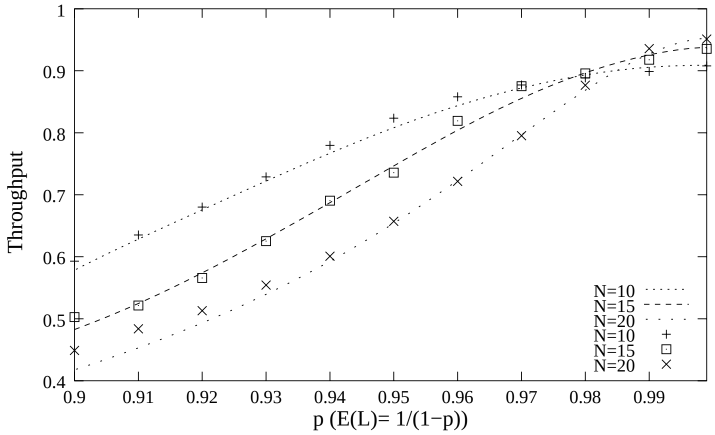
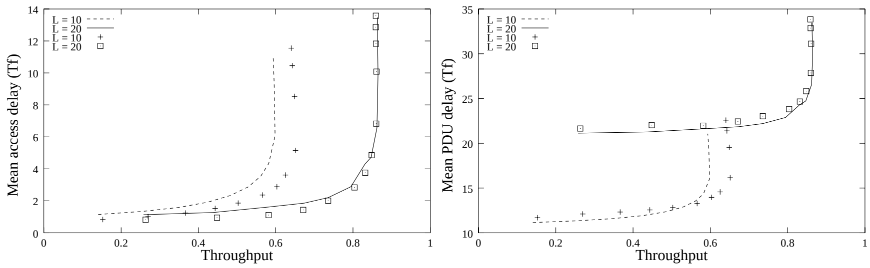


Figure 6.9: Impact of the number of TCHs on the throughput

overhead caused by the ACH and the bottleneck effect with a large N is discussed in Section 6.1.2.6.

6.1.2.6 Impact of the Number of TCHs

Figure 6.9 shows the impact of the number of the TCHs (N) on the throughput. The throughput is compared with the same arrival rate with N = 10, 15 and 20, respectively. The throughput vs. p, the parameter of the PDU train length distribution (E(L)=1/(1-p)) is plotted. The maximum achievable throughput is 0.91, 0.94 and 0.95 with N = 10, 15 and 20, respectively. If PDU trains are long enough (large p), the number of TCHs of one MAC frame (N) should be as large as possible to reduce the protocol overhead caused by the ACH. But from the Figure 6.9 we can see that with smaller p (corresponding to short PDU trains), the throughput is very low if N = 20 is used due to the bottleneck effect on the ACH. For short PDU trains, a smaller N achieves a much higher throughput performance. The throughput with N=10 exceeds the throughput with N=15 and N= 20 if p is smaller than 0.97 and 0.98, respectively. The throughput with N=15 exceeds the throughput with N=20 if p is smaller than 0.99. As the difference of the maximum achievable throughput with N=20 from that with



(a) Mean access delay vs. throughput

(b) Mean PDU delay vs. throughput

Figure 6.10: Mean access delay and mean PDU delay (in time frames)

N=15 is insignificant, N = 15 seems to be the best value if N is not adapted dynamically. The best performance can be achieved if N can be adjusted dynamically according to the mean length of PDU trains.

6.1.2.7 Mean Delay vs. Throughput

Figure 6.10 plots the mean access delay and the mean PDU delay vs. the throughput with a mean length of PDU trains (E(L)) = 10 and 20. Figure 6.10(a) shows that the mean access delay remains low as long as the network is not saturated. After the maximum throughput is reached, the mean access delay increases with the increase of the arrival rates of PDU trains. Here we can see the same effect as indicated in Figure 6.8 that the maximum achievable throughput with E(L) = 20 (0.8) is much higher than that with E(L) = 10 (0.6). The mean access delay will not, however, infinitely increase as one message traffic is assumed. One message traffic means that the next PDU train is generated only after the previous PDU train is fully served. Thus, the number of *acc s-PDUs* waiting for serve by the ACH server is limited.

Figure 6.10(b) shows the mean PDU delay vs. the throughput. The mean PDU delay is proportional to the mean length of PDU trains as indicated in Equation (6.20). The derivation of Equation (6.20) is confirmed in Figure 6.10(b) by simulation results, that are depicted with points.

6.2 Computer Simulation

In this section the traffic performance of W-CHAMB networks is intensively studied using the simulation tool. The impact of the system parameters, i.e. network connectivity, transmission rate, number of APs, traffic characteristics, channel errors, and protocol variables, on the traffic performance is systematically investigated. The performance gain using the developed protocols and algorithms can be seen from the simulation results.

6.2.1 Simulation Scenarios

The k x k grid network shown in Figure 6.11 is used as the basis for the simulation scenarios to study the traffic performance of W-CHAMB networks. The k x k grid network can represent the characteristics of W-CHAMB networks in terms of multihop transmission, frequency spatial reuse, co-channel interference, hidden station problem and network connectivity. All WSs are assumed to have the same radio characteristics, such as transmission power, communication range, receive sensitivity. A desired network connectivity is achieved by adjusting the transmit power of the wireless stations accordingly. The connectivity is defined as the mean number of neighbors, normalized by the number of the maximum possible number of neighbors [15].

$$c = \frac{1}{N(N-1)} \sum_{i=1}^{N} n_i \tag{6.21}$$

where n_i is the number of neighbors to station i, N is the number of stations in the network. A fully connected network has a connectivity of 1.

In the case that no AP is considered, a WS randomly selects another station as its traffic sink. The Minhop routing algorithm is used to establish a multihop connection if the destination cannot be reached directly. To study the traffic performance of access networks, some of the WSs in the network can be replaced by access points (APs). In an access network, most of the connections are established between the access point and WSs. The radio

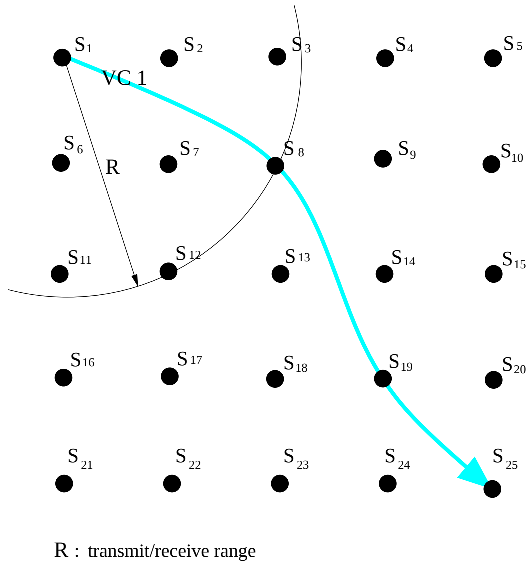


Figure 6.11: Simulation scenario

characteristics of an AP are assumed the same as those of a WS. An AP differs from a WS on the aspect that an AP is connected to the core network.

6.2.2 W-CHAMB Efficiency

To study the protocol efficiency of W-CHAMB networks a 5x5 grid network with a connectivity $c = 1$ is used. Each WS is loaded with a generic ABR traffic source specified in Section 5.5.1 with a mean length of PDU trains of 30, and randomly selects another WS as its traffic sink. The mean inter-arrival time of the ABR PDU trains is varied to model the various traffic loads. No channel error is considered except in Section 6.2.2.4 where the impact of channel errors on the traffic performance is studied.

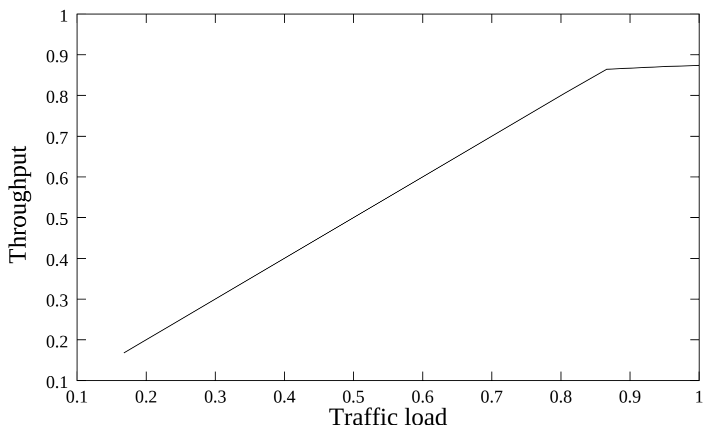
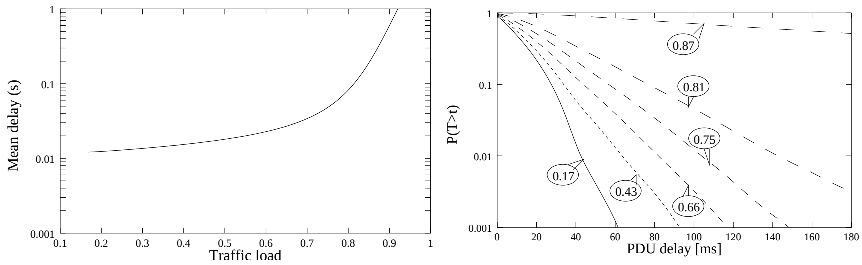


Figure 6.12: Throughput

6.2.2.1 Throughput and Mean Delay

As a wireless channel is always a very scare and expensive resource, the channel utilization of a wireless network with a high load should be as high as possible under the condition that the mean delay is still acceptable for the application. The throughput (channel utilization) of W-CHAMB networks is the ratio of the number of all transmitted data PDU to the number of simulation time slots. As one time slots every MAC frame is used as an ACH, the maximum channel utilization is $15/16 = 93.75\%$ if the MAC frame has 16 time slots, and E-signals are sent on the NECH.

Figure 6.12 shows that the maximum throughput of ABR traffic with the mean length of PDU trains of 30 is about 87% that is about 7 % less than the theoretical maximum throughput. One reason is that a part of TCH slots are used to send signaling PDU (*ack-s-PDU*) to establish an RCC or remain empty at the end of the RCC to implicitly inform other WSs the release of the RCC. Another reason is the bottleneck effect of the ACH. As each MAC frame can serve one *acc-s-PDU* at the most, other WSs have to wait for the next frame even when there are many free TCHs. But 87% is really a very high utilization for a decentralized MAC protocol. Before the network is saturated, the throughput increases linearly with the increase of the traffic load. This means that all PDUs are served till the maximum throughput is



(a) Mean delay **(b)** CDF

Figure 6.13: Delay performance

reached. After that the network is saturated and the throughput can not increase further. Although W-CHAMB is stable at the saturated situation, the network should be avoided to operate under a saturation condition as the delay performance degrades significantly.

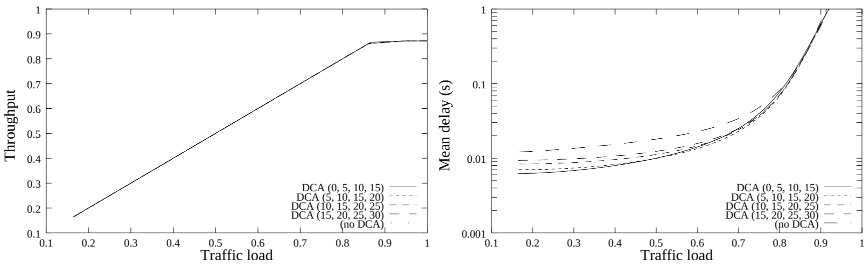
Figure 6.13 shows the delay performance of the W-CHAMB network. The mean PDU delay increases very quickly when the network approaches saturation, see Figure 6.13(a). Before the network is saturated, the increase of the mean delay over load is very small. Even at a very low load situation, the mean delay does not approach zero as a PDU train is transported over many MAC frames in sequence even there are many free TCHs available. The delay performance can be improved if more TCHs are allowed to be reserved for an ABR RCC so that the transmission duration of an ABR train can be reduced. This is achieved by the DCA algorithm described in Section 4.4.2.3. Figure 6.13(b) shows the complementary distribution function (CDF) of PDU delay with various traffic loads. The CDF shows the probability that a PDU experiences a delay larger than a given value. From Figure 6.13(b) we can see that at the traffic load of 0.81 about 0.5% PDUs experience a delay large than 180 ms. With the load of 0.87 about 80% PDUs experience a delay large than 180 ms as the network is saturated with this load.

6.2.2.2 Impact of the DCA Algorithm

In Section 4.4.2.3 the DCA algorithm is designed to use the channel resources more efficiently and fairly. The bandwidth of an ABR RCC is dynamically allocated according to the network load condition. Figure 6.14 shows the traffic performance with different DCA parameter sets (N_4, N_3, N_2, N_1). Figure 6.14(a) shows that DCA has almost no impact on the throughput in this case. This is because with a low traffic load all data PDUs can be transported, whilst with a high traffic load all TCHs are almost always reserved as it is assumed that all WSs have the same traffic characteristics. The throughput gain will become significant in the case that some of the WSs have much higher data rates than others or the mean data rate of the traffic loads varies during the communication. In this case, some WSs may have a lot of PDUs to send and some may not. The channel utilization will be improved to allow the WSs having heavy load to increase the bandwidth of an established RCC if there are free channels.

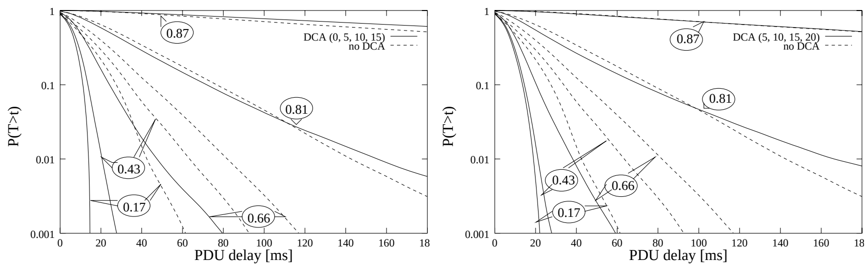
The impact of DCA parameter set on the mean delay performance is shown in Figure 6.14(b). We can see that DCA improves the mean delay performance before the network is saturated. With a smaller DCA parameter set (N4, N3, N2, N1) the delay performance is better with a low traffic load. This is because with a a smaller DCA parameter set a free TCH can be reserved more quickly and the bandwidth of the reserved ABR RCC decreases more slowly. However, with a high traffic load, the delay performance with a larger DCA parameter set is better. This is because at the high load the number of free TCHs is reduced. A smaller DCA parameter set causes more signaling traffic that results in a larger PDU delay. At the network saturation condition, DCA cannot improve the delay performance any more.

The delay performance using the DCA algorithm with various traffic loads can also be seen from the CDF shown in Figure 6.15 and 6.16. With the DCA parameter set (0, 5, 10, 15) a better delay performance is achieved with a traffic load lower than 43%. With the increase of the traffic load to 0.66 a larger parameter set (5, 10, 15 20) results in a better delay performance than with DCA parameter set (0, 5, 10, 15). With a traffic load of 81% where the network approaches the saturation, the DCA parameter set (10, 15, 20, 25) produces the best delay performance. This means that with a



(a) Throughput **(b)** Mean delay

Figure 6.14: Impact of the DCA algorithm

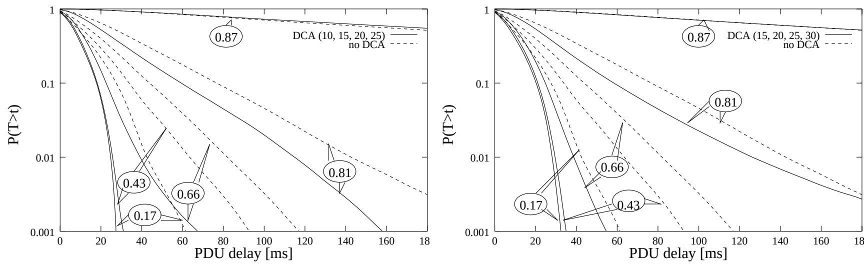


(a) DCA parameter set (0, 5, 10, 15) **(b)** DCA parameter set (5, 10, 15, 20)

Figure 6.15: Impact of DCA parameter set on the delay distribution

high traffic load a larger DCA parameter set is preferred. After the network is saturated, DCA has almost no impact on the delay performance.

The maximum bandwidth that can be reserved by an ABR RCC is limited by the DCA parameter R at the high network load situation to share the channel resource fairly among the WSs. If the network is highly loaded, WSs that reserve more than R TCHs must reduce the number of reserved TCHs to R. This is to avoid that the channel resource is used by a few WSs that have a huge amount of traffic, whilst other WSs have no chance to reserve a TCH. Figure 6.17 shows the impact of the DCA parameter R



(a) DCA parameter set (10, 15, 20, 25)

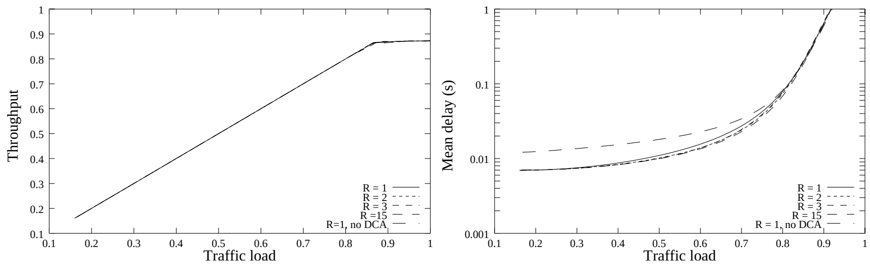
(b) DCA parameter set (15, 20, 25, 30)

Figure 6.16: Impact of DCA parameter set on the delay distribution

on the traffic performance. We see in Figure 6.17(a) that R has almost no effect on the maximum channel utilization. This is because the limitation of the bandwidth of the ABR RCC is valid only at a high load situation. In this case, the TCHs freed by an ABR RCC will always be used by other WSs so that the maximum throughput is not affected. The mean delay is slightly impacted by the DCA parameter R at a moderate to high load situation. With R = 1 the probability that a WS has to release a TCH increases. This results in a moderate increase of the mean delay. But R=2 appears enough for this traffic scenario as the mean delay does not change any more with a further increase of the parameter R. The parameter R has almost no impact on the traffic performance with a low traffic load as there are so many free TCHs that the bandwidth of an RCC is not limited by the parameter R. It is an encouraging feature of DCA that the fairness of the MAC protocol can be achieved without a significant impact on the traffic performance.

6.2.2.3 Impact of the Mean Length of PDU Trains

Figure 6.18(a) shows the impact of the mean length of PDU trains on the traffic performance. Unlike the analytical results in Section 6.1.2.5 the throughput is not impacted by the mean length of PDU trains. No bottleneck effect of the ACH appears in this situation. This is because we have



(a) Throughput **(b)** Mean delay

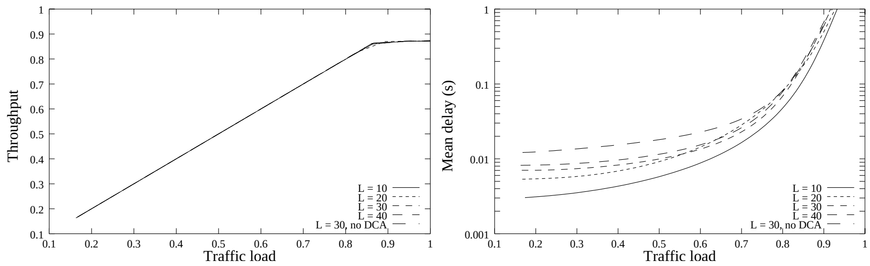
Figure 6.17: Impact of the DCA parameter R

increased the traffic load by reducing the interarrival time of PDU trains. At a high traffic load, the next PDU train at the same source WS may arrive before the previous PDU train is sent. The PDU train can then be served directly without a preceding signaling. The bottleneck of ACH would be visible seriously with a small mean length of PDU trains if the traffic load would have been increased by increasing of the number of active WSs.

The impact of the mean length of PDU trains on the mean PDU delay is shown in Figure 6.18(b). A larger mean length of PDU trains results in a larger mean delay. However, the impact of the mean length of PDU trains on the delay performance is reduced using DCA. We see in Figure 6.18(a) that the mean delay with a mean length of PDU trains of 40 differs from that with a mean length of PDU trains of 10 by about 5 ms. If no DCA is used, the difference should be 30 x 0.72 = 21.6 ms.

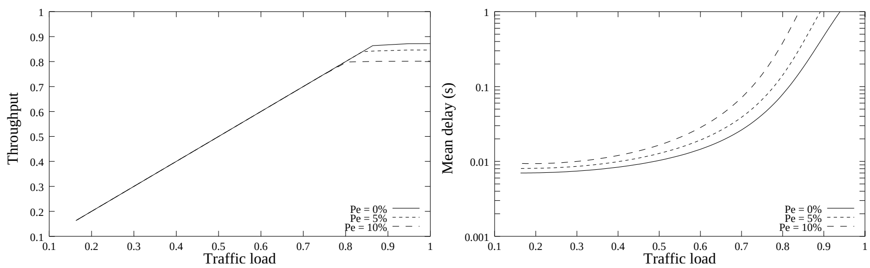
6.2.2.4 Impact of Channel Errors

Figure 6.19 shows the impact of channel errors on the traffic performance. The maximum channel utilization and mean delay performance becomes a little worse with a channel error of 5% and 10%, respectively. This is because some channel resources are used to retransmit the erroneous PDUs. From the traffic performance in Figure 6.19 we can conclude that the error control technology realized by E-Signals is efficient and stable since the



(a) Throughput **(b)** Mean delay

Figure 6.18: Impact of the mean length of PDU trains



(a) Throughput **(b)** Mean delay

Figure 6.19: Impact of channel errors

impact of transmit errors on the traffic performance is very limited even under a channel error rate of 10%.

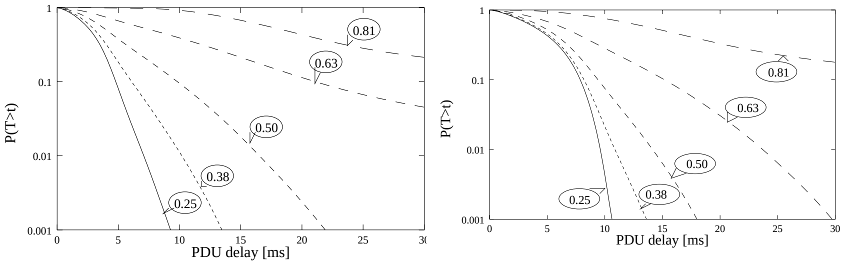
6.2.3 QoS Guarantee for rt-VBR Services

One of the unique features of W-CHAMB networks is its ability to guarantee QoS for rt-VBR services in a self-organizing multihop network. As rt-VBR services are delay sensitive, rt-VBR PDUs must be delivered promptly. An rt-VBR PDU is useless and can be discarded if its delay exceeds a maximum value. QoS guarantee for rt-VBR services means to ensure that the end-to-end rt-VBR PDU delay does not exceed a predefined maximum value. To study the QoS supported by W-CHAMB networks, the 5x5 grid network shown in Figure 6.11 is investigated. The number of active WSs varies to achieve the different traffic loads. All active WSs are loaded with rt-VBR traffic. We use the autoregressive model specified in Section 5.5.1 with a mean length of PDU trains of 30. The interarrival time of rt-VBR PDU trains is 41.67 ms modeling a video codec producing 24 pictures per second.

6.2.3.1 Impact of the rt-VBR RCC Scheduling Algorithm

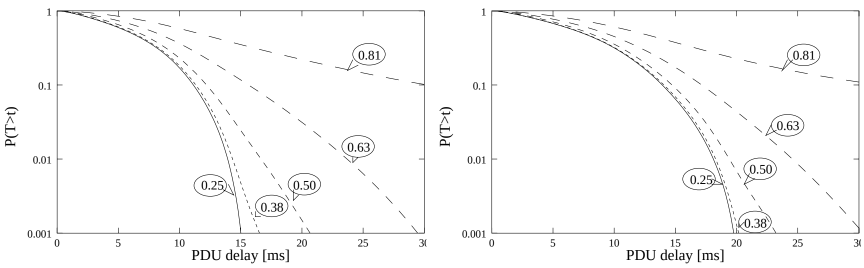
Before an rt-VBR PDU train can be delivered, an rt-VBR RCC must be established. The bandwidth of the rt-VBR RCC is scheduled by the LLC entity, see Section 4.4.2.1. In the scheduling algorithm, the bandwidth of the required rt-VBR RCC is decided according to the parameter called transmission duration T_d together with the PDU train length. If the parameter T_d is set smaller, the PDU train will be delivered in a shorter duration. But a smaller T_d increases the bandwidth requirement for the rt-VBR RCC. If the network is highly loaded, it will be difficult for a WS to establish an RCC with a large bandwidth requirement. Thus, the access delay will increase accordingly. Figures 6.20 and 6.21 show the CDF of the rt-VBR PDU delay with T_d = 5ms, 10ms, 15ms and 20ms. With T_d = 5ms, the best delay performance is achieved at a low load (0.25). With the increase of the traffic load, a much better performance is achieved using T_d = 10ms. With T_d = 10ms, the maximum PDU delay is still controlled to be below 30 ms for 99.9% of all PDUs with 63% traffic load. Figure 6.21 indicates that a further increase of the parameter T_d degrades the traffic performance. $T_d = 10ms$ seems to be the optimized value in this case. The results indicate that the traffic load of rt-VBR traffic must be controlled to guarantee QoS for rt-VBR traffic. Usually, the real time traffic load should not exceed 50% in a fully connected W-CHAMB network. Here, the traffic load refers to the ratio of the number of generated PDUs to the number of channel slots that can deliver one PDU each.

To study the rt-VBR traffic performance in a multihop network a network with connectivity c = 0.34 is investigated. In this case, multihop connections are established for a WS that cannot reach its destination directly. Figures



(a) T_d=5ms, c=1.0 **(b)** T_d=10ms, c=1.0

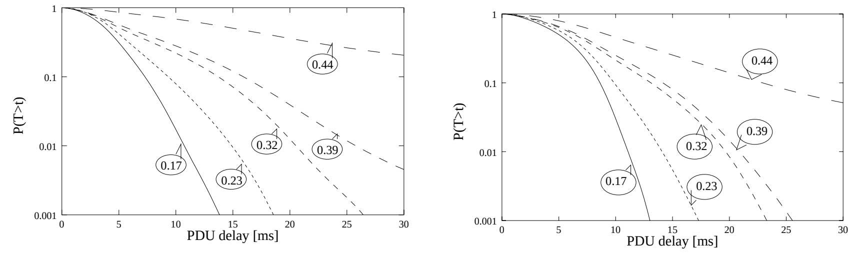
Figure 6.20: CDF of rt-VBR traffic, c=1.0



(a) T_d=15ms, c=1.0 **(b)** T_d=20ms, c=1.0

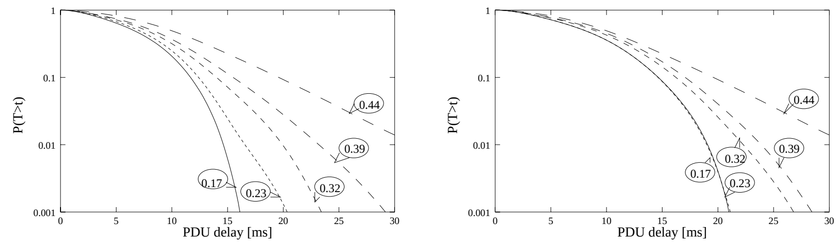
Figure 6.21: CDF of rt-VBR traffic, c= 1.0

6.22 and 6.23 show that the impact of the parameter T_d on the rt-VBR traffic performance in the multihop network is similar to that in the one hop network. The best suited value of the parameter T_d is also 10 ms. Due to the multihop transmission of PDU trains the maximum traffic load must be reduced in a multihop network in comparison with the one-hop case to guarantee the required QoS for the rt-VBR services. For a network with c = 0.34, the traffic load of rt-VBR services should not exceed 40 %.



(a) T_d=5ms, c=0.34 **(b)** T_d=10ms, c=0.34

Figure 6.22: CDF of rt-VBR, c=0.34

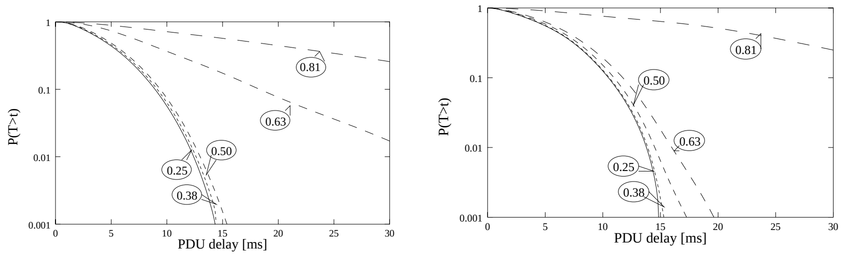


(a) T_d=15ms, c=0.34 **(b)** T_d=20ms, c=0.34

Figure 6.23: CDF of rt-VBR, c=0.34

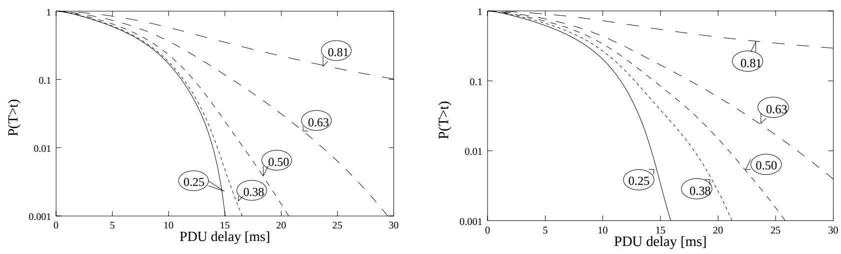
6.2.3.2 Impact of the Mean Length of PDU Trains

The Figures 6.24 and 6.25 show the impact of the mean length of PDU trains (L) on the QoS of the rt-VBR traffic. We can see that as long as the traffic does not exceed 50 %, the QoS of the rt-VBR traffic can be guaranteed since only 0.1% of the PDUs exceed a delay of 15 ms in Figure 6.24(a). With a traffic load lower than 50%, traffic with short PDU trains (L= 10) has the best delay performance as the transmission time of a short



(a) L=10 **(b)** L=20

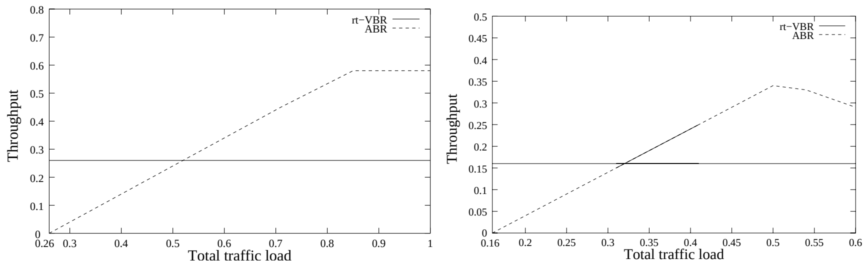
Figure 6.24: CDF of rt-VBR traffic, c=1.0



(a) L=30 **(b)** L=40

Figure 6.25: CDF of rt-VBR traffic, c= 1.0

PDU train decreases. Since the delivery of a longer PDU train needs an RCC with more bandwidth that is difficult to be allocated with the DCA algorithm, the access delay increases accordingly. At the high traffic load (0.63) condition, the traffic performance with L = 20 is better than that with L = 10 due to the bottleneck at the ACH with L = 10. As the maximum delay is well controlled, the traffic performance meets the requirements of real time services.



(a) c=1.0 **(b)** c=0.34

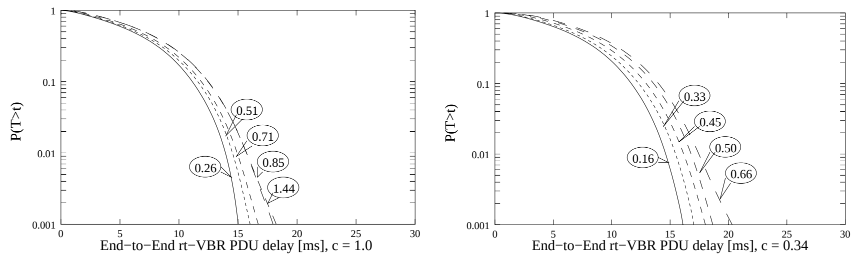
Figure 6.26: Throughput of rt-VBR and ABR traffic

6.2.3.3 Impact of ABR Traffic on the QoS of rt-VBR Services

The rt-traffic performance with a mixed traffic load is studied with the 5x5 grid network with c = 1.0 and 0.34. 8 WSs (5 WSs) generate rt-VBR connections with c = 1.0 (0.34). The other stations are loaded with ABR traffic. The ABR traffic load is increased by adjusting the interarrival time of ABR PDU trains.

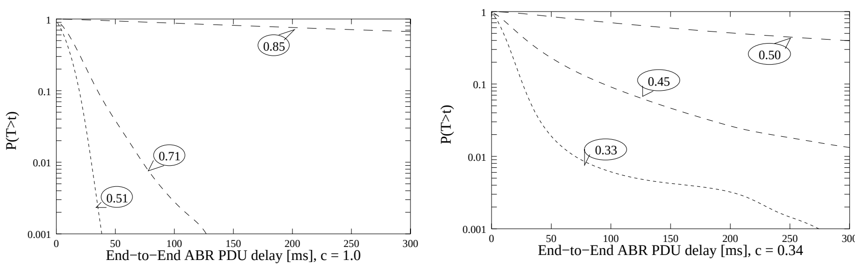
The real time traffic performance under mixed traffic load is displayed in the Figures 6.26, 6.27 and 6.28. The total traffic load is the ratio of the number of generated PDUs including ABR PDUs and rt-VBR PDUs to the number of simulated channel slots including ACHs.

As rt-VBR traffic is given a higher priority than ABR traffic, all rt-VBR PDUs are served without dropping. The increase of ABR traffic does not affect the throughput of the rt-VBR services. Figure 6.26 shows that the throughput of ABR traffic does not increase further with a traffic load higher than 0.85 and 0.50 with connectivity of 1.0 and 0.34, respectively. The maximum throughput of ABR traffic with c = 1.0 is much higher than that with c=0.34 as multihop transmissions are necessary for end-to-end connections with c = 0.34. After the network is saturated, the throughput of ABR traffic reduces a little with a further increase of the traffic load with c= 0.34, whereas the throughput of ABR traffic with c=1.0 keeps constant. The reason is that some PDUs are blocked in the intermediate WSs and are



(a) c=1.0 **(b)** c=0.34

Figure 6.27: CDF of rt-VBR traffic



(a) c=1.0 **(b)** c=0.34

Figure 6.28: CDF of ABR traffic

not able to reach the destination with c= 0.34 after the network is heavy loaded. With c = 1.0 the network is fully connected and no intermediate WSs exist for end-to-end connections.

The CDF of rt-VBR traffic is displayed in Figure 6.27. The delay of the rt-VBR traffic is under control even at the heavy saturation condition. The delay of rt-VBR PDU with c=0.34 is a little higher than that with c = 1.0 due to multihop transmissions of PDU trains. But the QoS of rt-VBR traffic is guaranteed in the both cases. The difference between the delay of rt-VBR PDUs without ABR traffic load and with a high ABR traffic load is not significant. The high ABR load increases the delay of the rt-VBR PDUs about several milliseconds that are necessary for a WS to interrupt an ABR transmission in favor of an rt-VBR PDU train in the case that no free channels are available at the arrival of an rt-VBR PDU train. A comparison of the Figures 6.27(a) and 6.27(b) reveals that with c = 1.0 a WS can more effectively interrupt an ABR transmission than with c=0.34. The delay of rt-VBR PDUs with a load of 0.85 is almost the same as that with a load of 1.44 if c=1.0, whilst the delay of rt-VBR PDUs increases further with the increasing traffic load when c=0.34. The reason is that a WS is not able to interrupt an ABR transmission that is more than one hop away. If a WS can find neither a free TCH for an rt-VBR PDU train nor a TCH reserved for ABR traffic, it must ask its partner WS to release a TCH.

The performance gain of the rt-VBR traffic is on the cost of the delay performance of ABR traffic. The delay performance of ABR traffic is displayed in Figure 6.28. With the increase of the traffic load the delay of ABR PDUs increases significantly with both connectivities, c=1.0 and c=0.34. Fortunately ABR services are delay insensitive. The ABR delay performance is acceptable before the network is saturated. After the network is saturated, the delay of ABR traffic increases greatly. For this reason the network should avoid operation at a heavy saturation. This can be achieved by a flow control mechanism for the ABR traffic as applied with the TCP protocol of the Internet.

6.2.4 E-signal Concept in Comparison with TDD Operation

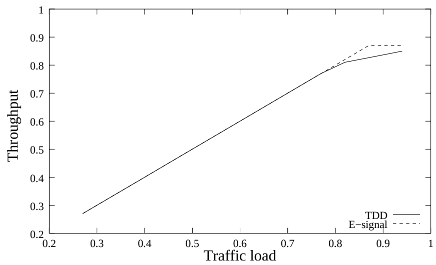
The TDD (Time Division Duplex) operation of a TCH has been proposed for W-CHAMB networks to achieve bidirectional RCCs in [79]. With the TDD operation mode a TCH is used alternatively by the two communicating partners to protect the reserved TCH from the interference of hidden stations and to send an acknowledgment for the previous received PDU. If one of the communicating partners has no PDU to send, the RCC is not released and dummy PDUs must be sent to keep the RCC established. TDD operation suffers from this inefficiency especially with bursty traffic sources. Even with a bidirectional symmetric connection, the traffic performance degrades because the traffic sources of two communicating partners do not have the same active period at the same time. The problem becomes worse for asymmetric traffic sources that are typical with Internet and Multimedia applications.

To solve this problem the E-signal concept has been developed in this thesis instead of the TDD operation to solve the hidden station problem and to achieve a fast acknowledgment. The E-signal concept makes it possible for a WS to reserve a TCH in one direction only. If one of the traffic sources of a bidirectional connection becomes inactive, the respective RCC can be released immediately. This achieves an efficient utilization of the scarce channel resources. Two RCCs may be established independently for a bidirectional connection.

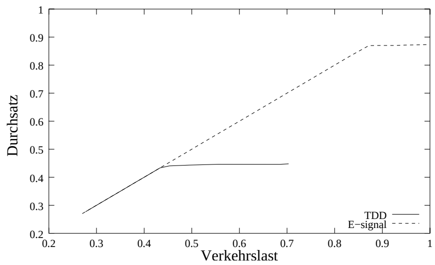
To compare the traffic performance using the E-signal concept and TDD operation mode a network with $c = 1.0$ is investigated. The number of active WSs varies to model the various traffic loads. All WSs have the same traffic load. For the symmetric traffic simulations bidirectional connections are established with symmetric traffic loads. To simulate the network with asymmetric traffic loads, one of the communicating partners being a traffic sink does not generate traffic.

Figures 6.29, 6.30 and 6.31 show the comparison of traffic performance using the E-signal concept and TDD operation mode. With symmetric ABR services, the maximum throughput using E-signal and TDD operation is 0.85 and 0.87, respectively. Although the difference of the maximum throughput is very small, the network using the TDD operation mode approaches saturation much earlier than when using the E-signal concept. The network under TDD operation mode is saturated with a traffic load of 0.80, and a part of PDUs cannot be delivered if the traffic load increases further. The superiority of the traffic performance using the E-Signal concept is much more obvious with an asymmetric traffic load. With the asymmetric traffic, the back channel is not used to deliver a PDU using TDD operation mode. Figure 6.29 shows that the maximum throughput with asymmetric traffic loads is 0.45 and 0.87 using TDD operation mode and E-signal concept, respectively. About 100% performance gain is achieved using E-signal concept in comparison with the TDD operation mode.

Using the E-signal concept, the difference of the traffic performance with asymmetric traffic from that with symmetric traffic is negligible. Using

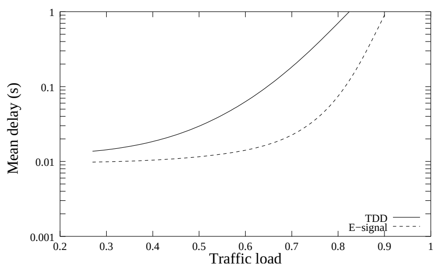


(a) Symmetric load

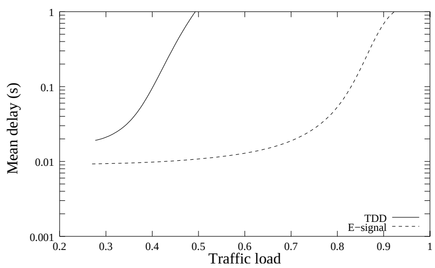


(b) Asymmetric load

Figure 6.29: Throughput



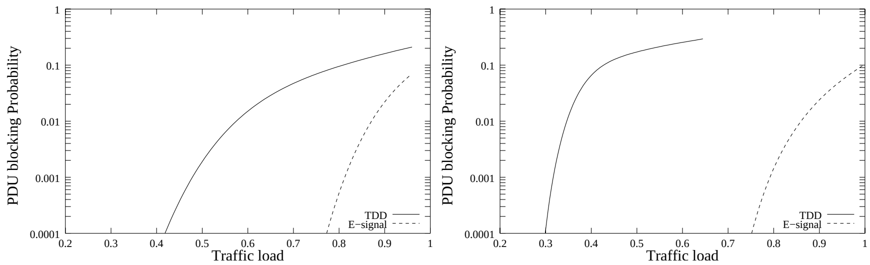
(a) Symmetric load



(b) Asymmetric load

Figure 6.30: Mean delay

the TDD mode, the much higher maximum throughput is achieved with symmetric traffic than with asymmetric traffic. With symmetric traffic, the increase of the traffic load increases the probability the back channel will be used by the communicating partners to deliver a data PDU. This is because that a PDU train has to wait for a long time to be delivered at a high traffic load. The next PDU train may arrive before the previous PDU train is delivered. Two active WSs may have PDUs to deliver at the same time and the back channel can be used to deliver a PDU. But the delay performance degrades with the increase of the traffic load quickly.



(a) Symmetric load **(b)** Asymmetric load

Figure 6.31: rt-VBR PDU blocking probability

Figure 6.30 shows that at a traffic load of 0.70, the mean delay under TDD operation mode is 10 times higher than using the E-signal concept. The maximum throughput improvement when using the TDD operation mode is on the cost of delay performance. rt-VBR PDU trains must be delivered promptly as an rt-VBR PDU must be dropped if its delay exceeds the predefined maximum value. The maximum delay value used in this simulation is 1500 slots (67.5ms). As an rt-VBR PDU cannot be queued for a long time, the blocking probability using the TDD operation mode is very high even with symmetric traffic. Figure 6.31 shows that the blocking probability with symmetric traffic is larger than 0.01 with a traffic load of 0.55 and 0.85 using the TDD mode and the E-signal concept, respectively. With an asymmetric traffic, TDD operation can support rt-VBR traffic only at a very low traffic load. With a small traffic load of 0.35, the rt-VBR PDU blocking probability is already larger than 0.01. The PDU blocking probability using the E-signal concept is almost the same with both symmetric and asymmetric traffic.

The simulation results reveal that the traffic performance using the E-signal concept is much better than that using TDD operation with symmetric and asymmetric traffic. The superiority of the traffic performance using the E-signal concept in comparison with that using TDD Operation becomes especially significant with asymmetric traffic loads.

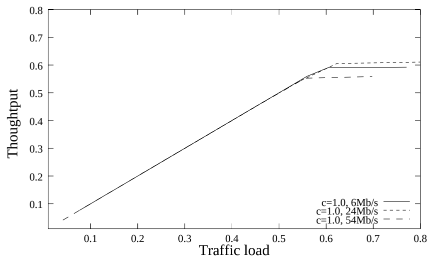
6.2.5 Traffic Performance with Various OFDM Schemes

W-CHAMB networks use various OFDM schemes to realize various transmission rates. In this section the traffic performance with various OFDM schemes is compared for a 5 x 5 grid network using the Ethernet trace file as traffic sources. Each Ethernet packet is segmented into a W-CHAMB PDU train. To compare the traffic performance using various OFDM schemes, the relative throughput that is the ratio of the actually achieved data rate to the channel transmission rate is used.

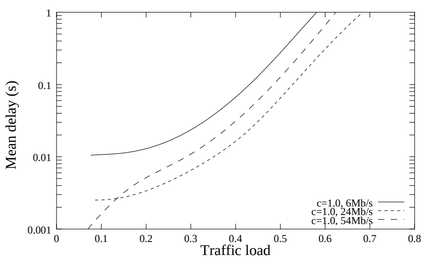
6.2.5.1 Traffic Performance with Various Transmission Rates

In one time slot of the W-CHAMB MAC frame a data PDU with 9 OFDM symbols can be transported. 9 OFDM symbols correspond to a PDU length of 27 byte, 108 byte and 243 byte with the transmission rates of 6 Mbit/s, 24 Mbit/s and 54 Mbit/s, respectively. As one time slot with the transmission rate of 6 Mbit/s can transport a data packet of 27 bytes only, two slots are combined to transport one W-CHAMB PDU of 54 bytes. Considering a PDU header of 6 bytes, the payload of one data PDU is 48 bytes, 102 bytes and 237 bytes with a transmission rate of 6 Mbit/s, 24 Mbit/s and 54 Mbit/s, respectively. The respective theoretical maximum data rate is 4.27 Mbit/s, 18.13 Mbit/s and 42.13 Mbit/s with a transmission rate of 6 Mbit/s 24 Mbit/s and 54 Mbit/s, resulting in the maximum throughput of 0.71, 0.75 and 0.78. It seems that, theoretically, a higher relative throughput can be achieved using a higher transmission rate.

Figure 6.32(a) shows the traffic performance with a network connectivity of 1.0 using a transmission rate of 6 Mbit/s, 24 Mbit/s and 54 Mbit/s. The respective maximum throughput is 0.60, 0.61 and 0.58 with a transmission rate of 6 Mbit/s, 24 Mbit/s and 54 Mbit/s. The simulation results of the maximum throughput are different from the theoretical maximum throughput. The reason is that the mean length of PDU trains using a transmission rate of 54 Mbit/s is much shorter than that using 6 Mbit/s and 24 Mbit/s as a PDU train corresponds to an Ethernet packet. Traffic with short PDU trains causes more contentions on the ACH resulting in a bottleneck there. The bottleneck effect becomes worse with a reduced network connectivity. Figure 6.33(a) and 6.34(a) show that with a smaller connectivity of 0.34



(a) Throughput



(b) Mean delay

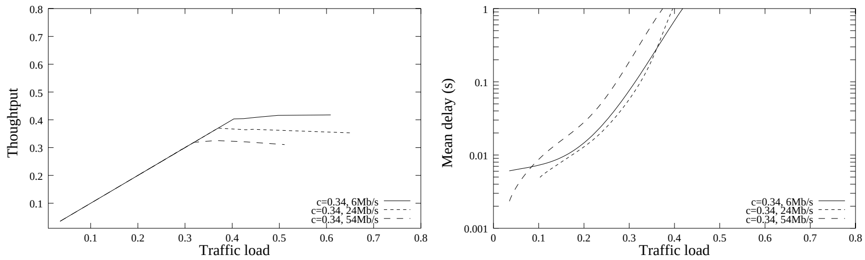
Figure 6.32: Traffic performance at c=1.0

and 0.24, the throughput with a higher transmission rate is much lower than that with a lower transmission rate.

Figure 6.32(b) shows the mean delay performance using various transmission rates. Mean delay using a transmission rate of 6 Mbit/s is larger than that using a transmission rate of 24 Mbit/s and 54 Mbit/s due to the larger mean length of PDU trains at a lower transmission rate. Mean delay using a transmission rate of 54 Mbit/s is higher than that using a transmission rate of 24 Mbit/s with a moderate to high traffic load due to the increase of the access delay with a transmission rate of 54 Mbit/s. The mean delay performance degrades with a reduced network connectivity. Figure 6.34(b) shows that the mean delay using 6 Mbit/s becomes better than using 24 Mbit/s and 54 Mbit/s with a connectivity of 0.24 with a traffic load larger than 0.2. With a very low traffic load, the mean delay using 54 Mbit/s is smaller because the mean length of PDU trains is very short using 54 Mbit/s, whilst the mean delay using 24 Mbit/s and 6 Mbit/s cannot decrease further.

6.2.5.2 Traffic Performance with Various Connectivities

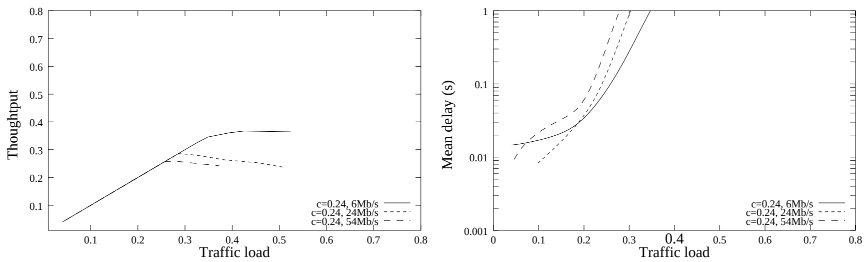
To study the impact of network connectivities on traffic performance, the simulation results with various connectivities using the same transmission rate are plotted. A desired network connectivity is achieved by adjusting the transmission power of the wireless stations accordingly. Using a high



(a) Throughput

(b) Mean delay

Figure 6.33: Traffic performance at c=0.34



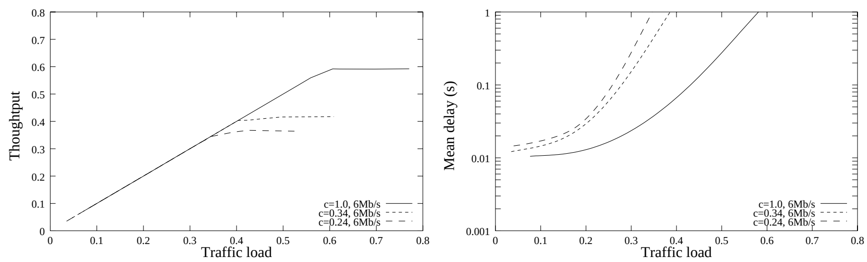
(a) Throughput

(b) Mean delay

Figure 6.34: Traffic performance at c=0.24

transmission rate a higher transmission power is needed to achieve the same network connectivity as using a low transmission rate.

Figure 6.35(a), 6.36(a) and 6.37(a) show the impact of the network connectivity on the throughput using 6 Mbit/s, 24 Mbit/s and 54 Mbit/s. With a connectivity of 1.0 the network has the best traffic performance. There, the throughput is increased linearly with the traffic load until the network becomes saturated at the traffic load of 0.60, 0.61 and 0.58 using 6 Mbit/s, 24 Mbit/s and 54 Mbit/s, respectively. The traffic performance is reduced with a smaller connectivity. With a connectivity of 0.24, the network is satu-



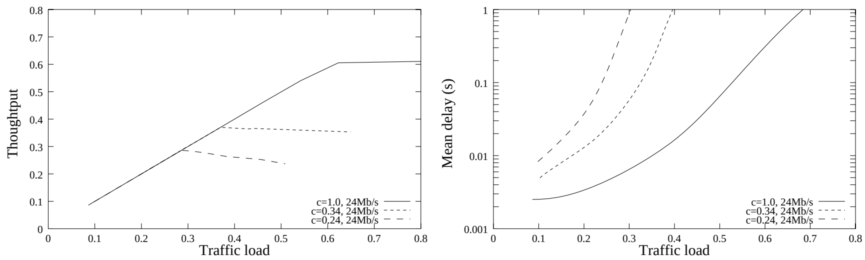
(a) Throughput **(b)** Mean delay

Figure 6.35: Traffic performance using 6Mbit/s

rated with a low traffic load of 0.35, 0.28 and 0.22 using 6 Mbit/s, 24 Mbit/s and 54 Mbit/s, respectively. Figure 6.35(b), 6.36(b) and 6.37(b) show the impact of the network connectivity on the mean delay using 6 Mbit/s, 24 Mbit/s and 54 Mbit/s. It can be seen that if the network is lightly loaded, the mean delay increases slowly with increased traffic load. The mean delay increases significantly if the network approaches saturation. Networks with a connectivity of 1.0 have always the best mean delay performance in comparison with networks with a smaller connectivity. The reason is that with a small connectivity the system capacity is used up rapidly owing to the multihop transmissions needed for multihop connections. The benefits of frequency spatial reuse in multihop networks is adverse to that of cellular networks that have many access points.

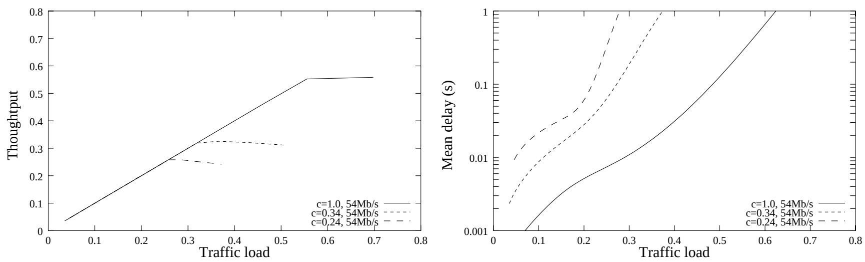
6.2.5.3 Traffic Performance using various OFDM schemes with the same transmission power

The results in Section 6.2.5.1 and 6.2.5.2 show that the relative throughput becomes lower with a higher transmission rate and/or a reduced connectivity. However, the throughput is a relative value that indicates the utilization of the transmission rate. Although the relative throughput is higher using a lower transmission rate, the usable bit rate of the channel that is more important for an application maybe lower with a lower transmission rate.



(a) Throughput **(b)** Mean delay

Figure 6.36: Traffic performance using 24 Mbit/s

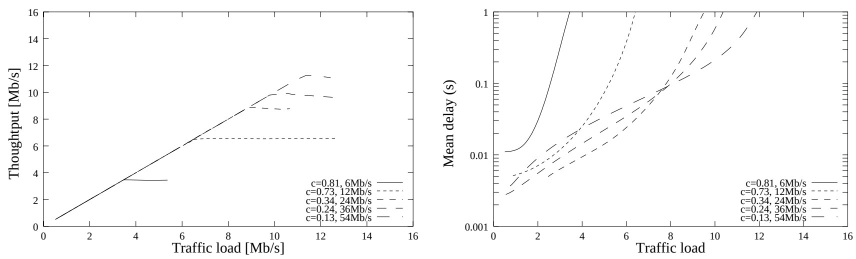


(a) Throughput **(b)** Mean delay

Figure 6.37: Traffic performance using 54 Mbit/s

Assuming that all WSs use the same transmission power, it is interesting to see the absolute throughput in terms of usable bit rate using various OFDM schemes. As a higher C/I ratio is needed to decode a PDU using an OFDM scheme with a higher transmission rate, the communication range of a WS is reduced. This means that the network connectivity is reduced using an OFDM scheme with a higher transmission rate if the transmit power is not increased.

Figure 6.38(a) shows the throughput using various OFDM schemes with the same transmission power. The network connectivity using a transmis-



(a) Throughput **(b)** Mean delay

Figure 6.38: Traffic performance without link adaptation

sion rate of 6 Mbit/s, 12 Mbit/s, 24 Mbit/s, 36 Mbit/s and 54 Mbit/s is 0.81, 0.73, 0.34, 0.24 and 0.13. Although the network connectivity using 54 Mbit/s is only 0.13, the maximum throughput is about 11.5 Mbit/s, whilst the maximum throughput using 6 Mbit/s is only 3.2 Mbit/s with a much higher connectivity of 0.81. The maximum throughput using 12 Mbit/s, 24 Mbit/s and 36 Mbit/s is 6.5 Mbit/s, 9 Mbit/s and 10 Mbit/s, respectively. The results indicate that the highest throughput can be achieved using 54 Mbit/s even though more hops are needed for an end-to-end connection. Figure 6.38(b) shows the mean delay performance. The delay performance using 54 Mbit/s is worse than that using 24 Mbit/s with a traffic load less than 8 Mbit/s. The reason is that using higher transmission rate more hops are necessary for an end-to-end connection. Multihop transmissions cause a large delay. However, OFDM schemes with a very low transmission rate is not recommended owing to its limited capacity. The traffic performance using 6 Mbit/s and 12 Mbit/s is worse than that using other higher transmission rates in almost all cases. OFDM schemes with a transmission rate less than 12 Mbit/s are not suited well for W-CHAMB networks. The OFDM scheme with 24 Mbit/s seems to be best suited for W-CHAMB networks with a low to moderate traffic load when using the modem standardized for HiperLAN/2 and IEEE 802.11a. A higher transmission rate is useful if the network is heavily loaded.

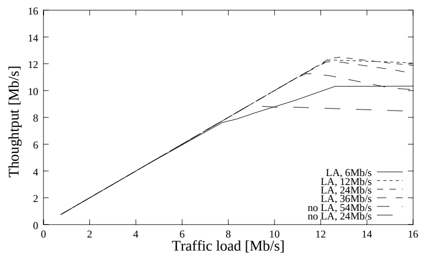
6.2.5.4 Traffic Performance with Link Adaptation

To study whether the traffic performance becomes better if a WS can select an OFDM scheme according to the actual signal strength and the C/I ratio, we use the link adaptation (LA) algorithm specified as follows:

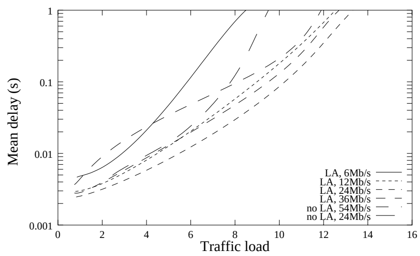
1. If the direct communication partner can be reached with an one-hop connection using a transmission rate larger than or equal to the minimum allowed transmission rate, the WS uses the highest possible transmission rate for the one-hop connection.

2. If the direct communication partner cannot be reached with an one-hop connection using a transmission rate larger than or equal to the minimum transmission rate, a multihop connection with n (n>1) hops is established. The selected transmission rate must be larger than or equal to the minimum transmission rate. n is selected as small as possible. After n is decided, the transmission rate is selected as high as possible based on the actual signal strength.

Figure 6.39 shows the traffic performance using link adaptation. In Figure 6.39(a) we can see that the traffic performance using a minimum of 24 Mbit/s with link adaptation is significantly better than that without link adaptation. The minimum transmission rate should not be very low. The network approaches saturation with a traffic load of 7.5 Mbit/s if the minimum transmission rate is 6 Mbit/s. The throughput can reach as high as 12.5 Mbit/s if the minimum transmission rate is 24 Mbit/s. This means that the traffic performance with a one-hop connection using 6 Mbit/s is worse than that with a multihop connection having a higher transmission rate. Figure 6.39(b) shows the mean delay performance with link adaptation. The mean delay is improved significantly using LA. The best delay performance is achieved with the minimum transmission rate of 24 Mbit/s using LA. If the minimum transmission rate is set to 6 Mbit/s, the mean delay performance is worse than almost all other cases. The only exception is that using 54 Mbit/s without LA at a low traffic load the mean delay is quite high due to the low network connectivity resulting in a multiple hops of the connections.



(a) Throughput



(b) Mean delay

Figure 6.39: Traffic performance with link adaptation

6.2.6 Traffic Performance with Access Points

The difference of an access network with access points from an ad hoc network without access points is that most of the traffic in the access network is to/from the access point. To study the traffic performance with access points, access scenarios with one AP and with 5 APs are studied. Each WS in the network communicates with the nearest AP using up and download ABR traffic connections. The traffic load is read from the Ethernet trace file. Data rates are varied to model the various traffic loads.

6.2.6.1 Traffic Performance with one AP

Figure 6.40 shows the access scenario with one AP. A 9 x 9 grid network is used. The access point is placed in the middle of the network. the communication range is varied from 1.42d to 5.7d to model the different network connectivity. d is the distance between two neighbors. We use the communication range instead of the connectivity defined in Section 6.2.1 as all connections are between the AP and WSs in the access scenario. The communication range is better suited to characterize the network in this case. With a small communication range, multihop connections must be established using the Minhop algorithm if the communication partner is

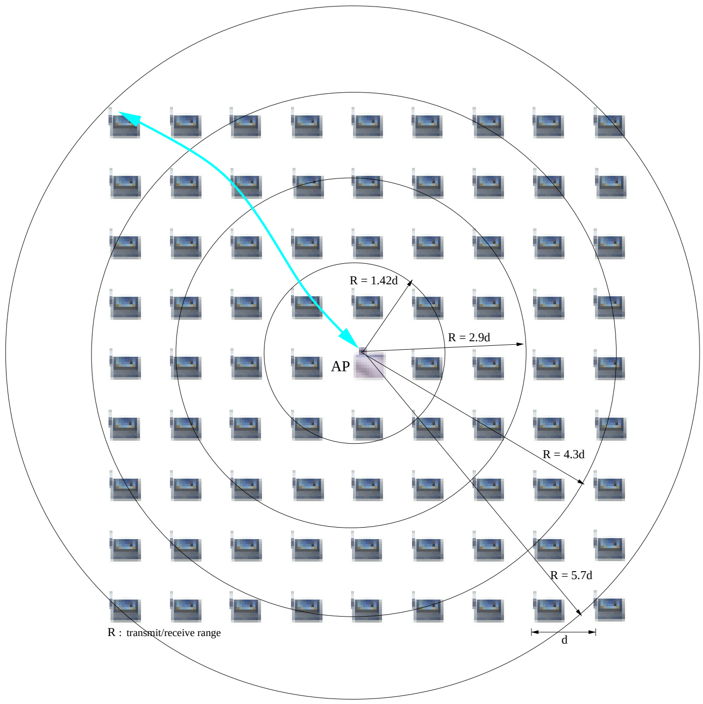
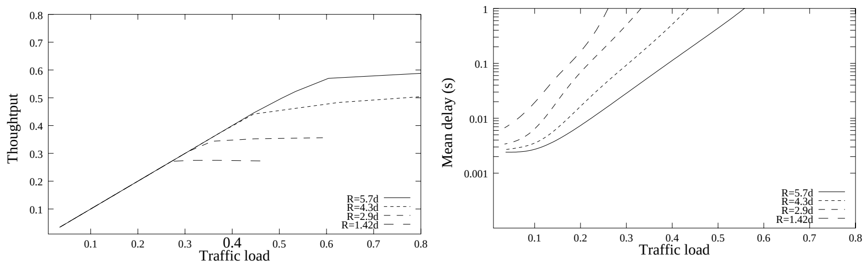


Figure 6.40: Access scenario with one AP

not the direct neighbor. The OFDM scheme 16 QAM 1/2 is used in the simulation giving a transmission rate of 24 Mbit/s.

The simulation results shown in Figure 6.41 indicate that a reduced communication range (transmission power) degrades the traffic performance in terms of throughput and mean delay in the access scenario with one AP. With a communication range of 1.42d, 2.9d, 4.3d, and 5.7d, the maximum throughput is 0.28, 0.35, 0.50 and 0.59. One reason for that is more transmission hops needed for an end-to-end connection with a reduced communication range. The gain from the frequency spatial reuse in the network cannot compensate the increase of traffic load due to the multihop transmission. Another reason is the bottleneck at the AP. With a reduced transmission power, the WSs at the edges of the network have more free channels



(a) Throughput **(b)** Mean delay

Figure 6.41: The Impact of network connectivities on the traffic performance with one AP

than necessary, whilst WSs in the center of the network have not enough free channels to deliver PDUs.

6.2.6.2 Traffic Performance with Multiple APs

The solution to the bottleneck at the AP is to place more APs in the access network, forming a cellular-like multihop ad hoc network. Figure 6.43 shows an access scenario with 5 APs, each connecting to the fixed core network. With a multiple APs scenario, traffic performance can be improved with a reduced transmission range since the traffic load is most balanced now throughout the networks.

Figure 6.43(a) shows that with a communication range of 1.42d, a maximum throughput more than 1.0 can be achieved because of the spatial reuse of the frequency channels. With a transmission range of 2.9d, all WSs can reach the AP with an one-hop connection. The maximum throughput is only 0.65 in this case. The reason is that with a transmission range of 1.42d, the bottleneck at the AP is significantly reduced as the interference of other WSs decreases. The benefit of the frequency spatial reuse can be gained in the access scenario with multiple APs. The mean delay performance with a transmission range of 1.42d is also much better than with a transmission range of 2.9d with a moderate to high traffic load, see Figure 6.43(b). Only

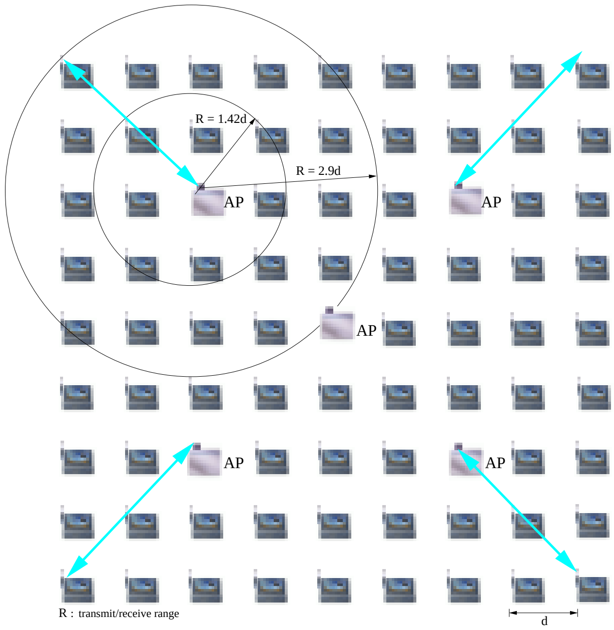
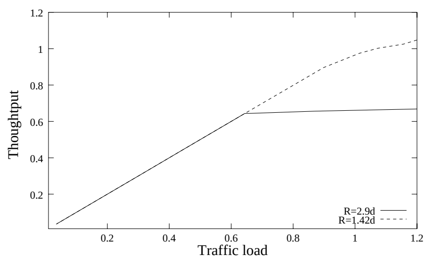
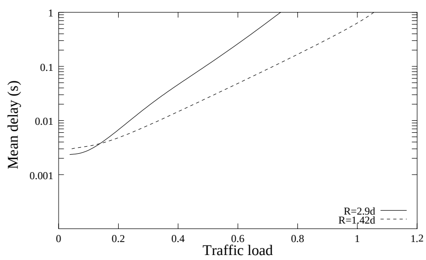


Figure 6.42: Access scenario with multiple APs



(a) Throughput



(b) Mean delay

Figure 6.43: The Impact of network connectivities on the traffic performance with multiple APs

with a very low traffic load, the mean delay with a transmission range of 1.42d is a little worse than that with a transmission range of 2.9d due to multihop transmissions

CHAPTER 7

Comparison: HiperLAN/2, IEEE 802.11a and W-CHAMB

In this Chapter the multihop traffic performance of HiperLAN/2, IEEE 802.11a and W-CHAMB with various scenarios is intensively studied and compared using the stochastic simulation tool where the protocols of HiperLAN/2 and IEEE 802.11a are also implemented based on the SDL specification [57, 120]. Scenarios of one-hop, two-hop and multihop with/without AP are used to compare the system behaviors of W-CHAMB, IEEE 802.11a and HiperLAN/2. Efficiency, QoS and multihop capability are the three main considerations to evaluate the systems. Supporting real time services in a self-organizing multihop ad hoc networking environment is one of the essential features of the next generation wireless communications.

7.1 One-hop Scenario

Figure 7.1 shows a one hop scenario for a fully connected wireless LAN. It consists of an AP serving a video stream download (MPEG) and a duplex voice connection (N-ISDN), and eight terminals exchanging Ethernet packets with their direct neighbor Mobile Terminals (MTs) in a circular way. The data rate of the video traffic has been set to 5 Mb/s whereas the mean data rate of the LAN traffic has been varied to model different loads. Traffic loads of MPEG and Ethernet are read from trace files [57, 120].

In the simulation, the transmission rate of HiperLAN/2 is set to 27 Mb/s. The transmission rate of IEEE 802.11a is 24 Mb/s. The PCF of IEEE 802.11 with Contention Free Repetition Interval of 10 ms is used to serve the N-ISDN and MPEG services that have priority over the Ethernet service. The transmission rate of W-CHAMB is also 24 Mb/s. Figure 7.2(a) shows the relative throughput of the different traffic flows under no bit errors where the load of the Ethernet service has been varied. In all systems

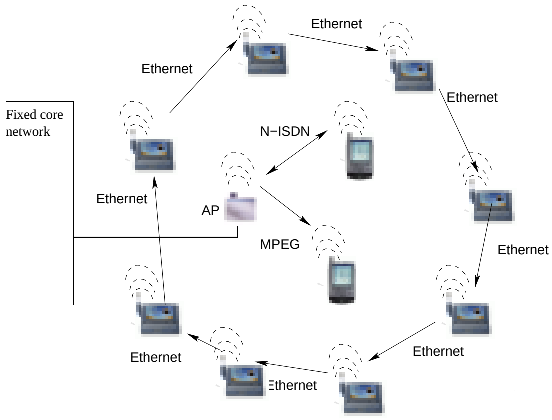


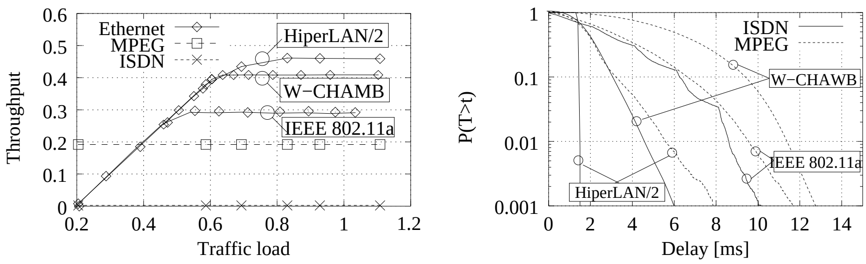
Figure 7.1: One hop scenario

the prioritized services N-ISDN and MPEG are served well under all load conditions, whereas the throughput for Ethernet is different. HiperLAN/2 has the highest throughput for the Ethernet service. The throughput of W-CHAMB for the Ethernet service is a little lower than that of HiperLAN/2, but significantly higher than that of IEEE 802.11a.

The complementary distribution function (CDF) of the packet delay at a high load condition is shown in Figure 7.2(b). Although W-CHAMB has neither a central controller as is used in HiperLAN/2 nor a point coordinator as used in IEEE 802.11a, the packet delay of N-ISDN is bounded to 6 ms and the delay of MPEG is limited to 13 ms with a total traffic load of 0.75, which meets the requirements of high performance video. In this one hop scenario, H/2 has the the best traffic performance for all services.

7.2 Two-hop Scenario

Figure 7.3 shows a two hop scenario with a central access point to a fixed network. Some MTs are placed out of the coverage range of the AP to act as RMTs. The end-to-end user connections (i.e. connections between AP and RMT) are loaded with Poisson traffic. We use this scenario to



(a) Throughput

(b) CDF of MPEG and N-ISDN

Figure 7.2: Traffic performance at one hop scenario

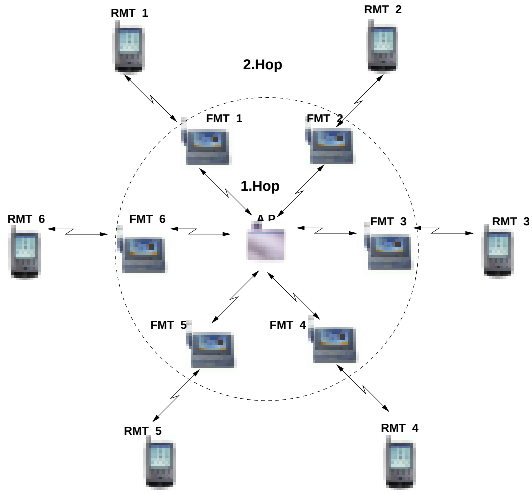


Figure 7.3: Two hop scenario

compare maximum throughput performance of IEEE 802.11a, HiperLAN/2 with FMT and W-CHAMB.

In Figure 7.4 the maximum system end-to-end throughput of the three systems under transmission rates of 18 Mb/s, 36 Mb/s and 54 Mb/s is shown. The packet size with W-CHAMB is 9 OFDM symbols, equivalent

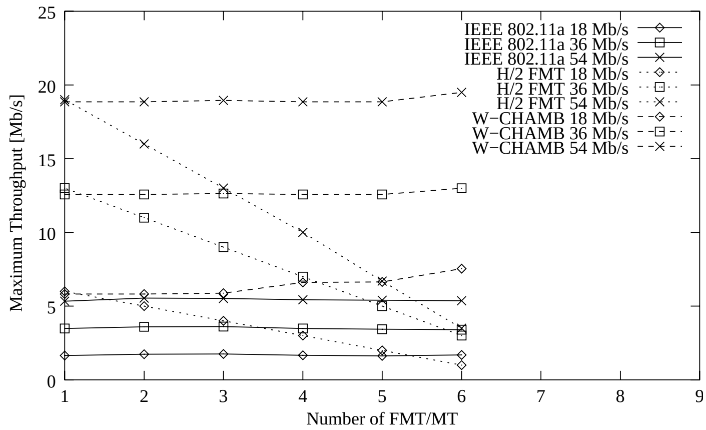
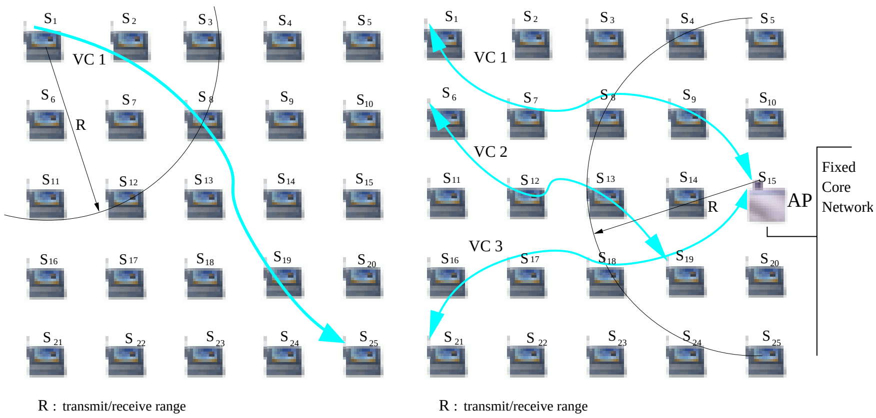


Figure 7.4: Maximum throughput performance

to 81 byte, 162 byte and 243 byte under transmission rates of 18 Mb/s, 36 Mb/s and 54 Mb/s, respectively. The respective packet sizes with IEEE 802.11a are 115 byte, 196 byte and 277 byte, each including a 34 byte packet header. The packet size with HiperLAN/2 is standardized as 54 byte including 6 byte packet header under all transmission rates. From Figure 7.4 we see that the throughput of IEEE 802.11a is very low due to short packet size and two hop transmission. For HiperLAN/2 with FMT, an increase of the number of FMTs that are serving one RMT strongly decreases the capacity available for AP-to-RMT connections as more and more capacity is needed for the overhead introduced by the sub frames of the FMTs. In contrast to HiperLAN/2 with FMT, an increase of the number of FMTs increases the maximum throughput slightly because of the frequency spatial reuse of the decentralized MAC protocol of W-CHAMB.

7.3 Multihop Scenario

To study the traffic performance in a multihop scenario WLAN, a 5x5 square grid network with/without an AP is studied, see Figure 7.5. A desired network connectivity is achieved by adjusting the transmit power of the wireless stations (WSs) accordingly. Different from the access scenario in Figure



(a) Multihop ad hoc scenario without AP

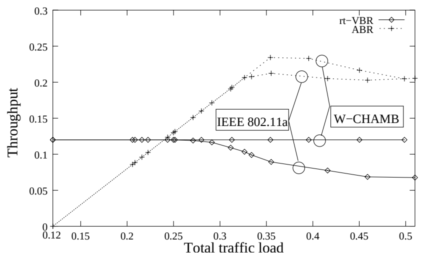
(b) Multihop access scenario with AP

Figure 7.5: Multihop scenario

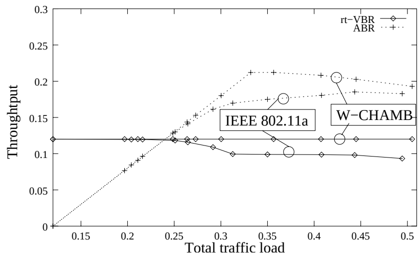
7.5(b), no access to a fixed network is considered at the ad hoc scenario in Figure 7.5(a). The Min-hop routing algorithm is used to establish a multihop connection. Since HiperLAN/2 cannot support multihop connections, it has not been included into the comparison.

7.3.1 QoS Guarantee

The unique feature of W-CHAMB is QoS guarantee for real time traffic services in a multihop network without any central control. This feature makes it best suited to be applied as a self-organizing wireless broadband multihop network. To evaluate the grade of the QoS support for real time traffic services, we study the network shown in Figure 7.5 at a connectivity of 0.34. For the ad hoc scenario in Figure 7.5(a), five stations are loaded with real time (rt)-VBR traffic. All other WS have one ABR connection each. Each WS randomly selects another station as its traffic sink. For the access scenario with AP in Figure 7.5(b), there are 25 end-to-end virtual connections (VCs) in total that are established permanently during the



(a) Throughput performance without AP (see Figure 7.5(a))



(b) Throughput performance with AP (see Figure 7.5(b))

Figure 7.6: Throughput performance at $c = 0.34$

simulation. Five rt-VBR end-to-end connections between the AP and 5 WSs are established to model the download real time video stream. Another 10 WSs receive ABR traffic from the AP, resulting in 10 download ABR end-to-end connections. 5 upload ABR end-to-end connections are established to transport ABR traffic from five WSs to the AP. There are five WSs that randomly select another WS as ABR traffic sink, resulting in 5 direct link end-to-end connections. For both scenarios, Figure 7.5(a) and Figure 7.5(b), the resulting mean of the five rt-VBR end-to-end connections is 1.6 hops, while the resulting mean of all ABR end-to-end connections is 1.9 hops. The packet size of rt-VBR traffic is modeled by an autoregressive Markovian process, with a mean of 3060 bytes and a maximum of 6120 bytes, yielding a burstiness factor of 2. The rt-VBR traffic source produces 24 packets per second. The packet size of ABR traffic is read from an Ethernet trace file (see Figure 5.10(a)). The interarrival time of Ethernet packets read from the trace file has been varied to model different loads.

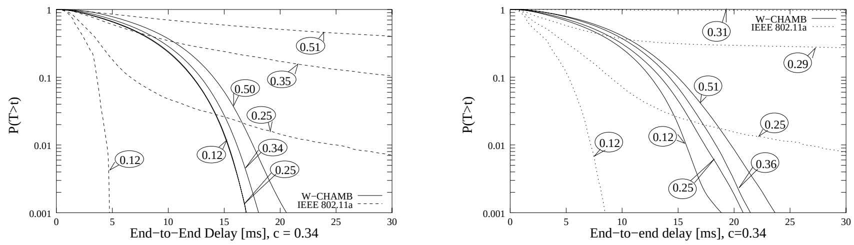
Figure 7.6 shows that for both scenarios in Figure 7.5, the prioritized rt-VBR service is served with the same throughput under all load conditions with W-CHAMB, whereas the throughput of rt-VBR traffic decreases with an increasing traffic load with IEEE 802.11a since a rt-VBR packet has been dropped there if its delay exceeds 30 ms. With the ad hoc scenario in Figure 7.5(a), the maximum throughput of the ABR service is 0.25 and 0.21 with W-CHAMB and IEEE 802.11a, respectively. At a heavy saturation

(0.4), the ABR throughput of W-CHAMB decreases a little due to the interruptions in favor of rt-VBR traffic.

Different from the ad hoc scenario in Figure 7.5(a), there are 5 rt-VBR VCs and 15 ABR VCs in the same AP at the access scenario in Figure 7.5(b) making it the highest loaded station. The maximum throughput of ABR traffic is 0.21 and 0.17 in Figure 7.6(b) instead of 0.25 and 0.21 in Figure 7.6(a) with W-CHAMB and IEEE 802.11a, respectively, due to the bottleneck at the AP. With the access scenario the maximum throughput of the ABR service is 0.21 with W-CHAMB at the total traffic load of 0.33, including rt-VBR traffic (0.12). After that, the network is saturated and the throughput of the ABR service decreases a little since ABR RCCs are frequently interrupted and resumed later to free a channel for rt-VBR traffic at a saturation condition. IEEE 802.11a, however, approaches saturation at a lower traffic load (0.26). After that, the throughput of the ABR service with IEEE 802.11a increases a little as many rt-VBR packets are discarded freeing capacity to carry more ABR packets. The throughput of the rt-VBR service with IEEE 802.11a does not decrease further, as visible from Figure 7.6(a), after the load of 0.31 since a WS or the AP transports rt-VBR packets with preference over ABR packets if both rt-VBR packets and ABR packets exist in the same WS.

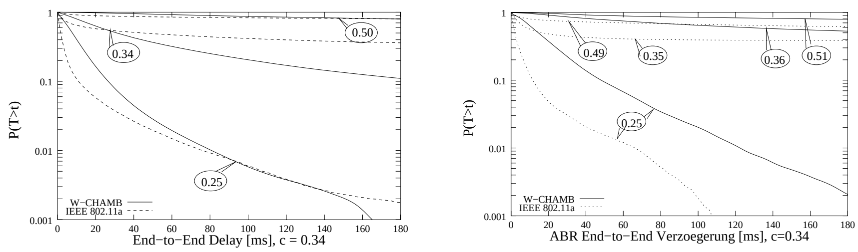
Figure 7.7 shows the CDF of the rt-VBR service with W-CHAMB and IEEE 802.11a under various traffic loads. The delay distribution of the rt-VBR service with W-CHAMB is still under control even under a heavy overloaded condition (0.5) for both scenarios in Figure 7.5. With an increasing traffic load from 0.3 to 0.5, the packet delay of rt-VBR traffic increases several milliseconds because the probability that a station cannot find a free channel for an rt-VBR PDU train at its arrival increases. Several milliseconds are necessary to interrupt the transmission of an ABR PDU train to free a channel for the rt-VBR traffic. IEEE 802.11a instead is not able to differentiate rt-VBR and ABR traffic among WSs in the air interface. At a very low traffic load condition, the rt-VBR service has a good delay performance. But the delay performance degrades rapidly with an increasing traffic load. With a moderate traffic load of 0.25, the delay performance of the rt-VBR service is no longer acceptable.

Figure 7.6 and 7.7 reveal that IEEE 802.11a cannot support an rt-VBR service in a multihop environment, whereas W-CHAMB is able to guarantee



(a) CDF of rt-VBR traffic without AP **(b)** CDF of rt-VBR traffic with AP

Figure 7.7: CDF of rt-VBR traffic at $c = 0.34$



(a) CDF of ABR traffic without AP **(b)** CDF of ABR traffic with AP

Figure 7.8: CDF of ABR traffic at $c = 0.34$

QoS even under a high traffic load for both scenarios, with and without an AP.

Figure 7.8 shows the CDF of the ABR traffic. The delay distribution of the ABR service significantly depends on the traffic load for both, W-CHAMB and IEEE 802.11a. In comparison with the ad hoc scenario in Figure 7.5(a), the access scenario in Figure 7.5(b) has a worse delay performance for both, ABR and rt-VBR services, due to the bottleneck effect at the AP.

7.3.2 Efficiency

7.3.2.1 Comparison of the Traffic Performance with Various Connectivities and Transmission Rates

To compare the traffic performance of W-CHAMB and IEEE 802.11a in a multihop environment with various network connectivities and transmission rates at the ad hoc scenario in Figure 7.5(a), realistic packet sizes read from an Ethernet trace file are used for the traffic load. Figure 7.9, 7.10 and 7.11 show the traffic performance with the transmission rates of 6 Mbit/s, 24 Mbit/s and 54 Mbit/s with a network connectivity of 0.24, 0.34 and 1.0, respectively. For both systems, W-CHAMB and IEEE 802.11a, the smaller the network connectivity is, the lower is the throughput. This is because at a smaller network connectivity more hops are needed to establish an end-to-end connection. The system capacity is used up rapidly owing to the multihop transmissions. The benefit of frequency spatial reuse in multihop networks is adverse to that of cellular networks that only use one hop per connection.

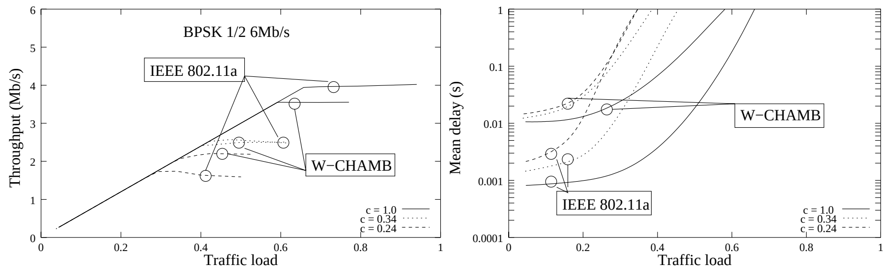
Figure 7.9 shows that with a transmission rate of 6 Mbit/s the traffic performance with IEEE 802.11a is better than that with W-CHAMB with a network connectivity of 1.0 and 0.34. The maximum throughput with IEEE 802.11a and W-CHAMB is 4 Mbit/s and 3.5 Mbit/s with $c = 1.0$. The inefficiency of W-CHAMB in this case is caused by the segmentation of the Ethernet packet to small W-CHAMB PDUs. With a smaller connectivity of 0.34, the difference of the throughput with IEEE 802.11a and W-CHAMB decreases. The respective maximum throughput is 2.6 Mbit/s and 2.5 Mbit/s. With a further decrease of the network connectivity, the efficiency of IEEE 802.11a decreases as it suffers from the hidden station problem. With a connectivity of 0.24, the maximum throughput with W-CHAMB is much better than with IEEE 802.11a. The respective maximum throughput is 2 Mbit/s and 1.5 Mbit/s. From the results in Figure 7.9 we can make a conclusion that IEEE 802.11a is suited with a low transmission rate as well as a high network connectivity. Two reasons can explain the results. One is that the protocol overhead of IEEE 802.11a caused by inter frame space and back off is relatively low with a low transmission rate. The other reason is that CSMA/CA of IEEE 802.11a functions well without the existence of hidden stations. The efficiency of IEEE 802.11a decreases with a high transmission rate as the inter frame space and back off time cause a significant protocol overhead. The conclusion is confirmed by the results shown in Figure 7.10 and 7.11. The superiority of the traffic performance of W-CHAMB is very significant for all connectivities with a transmission rate of 24 Mb/s and 54 Mbit/s. With a transmission rate of 24 Mbit/s, the maximum throughput with W-CHAMB and with IEEE 802.11a is 14.2 Mbit/s and 11 Mbit/s with c = 1.0, 9 Mbit/s and 7 Mbit/s with c=0.34, and 7.2 Mbit/s and 5 Mbit/s with c= 0.24, respectively. With a transmission rate of 54 Mbit/s, the traffic performance with IEEE 802.11a becomes worse. The respective maximum throughput with IEEE 802.11a is only 12.5 Mbit/s, 8Mbit/s and 6 Mbit/s, whilst the maximum throughput with W-CHAMB is 30 Mb/s, 17.5 Mbit/s and 12.5 Mbit/s with a connectivity c = 1.0, 0.34 and 0.24.

Figures 7.9(b), 7.10(b) and 7.11(b) compare the mean delay performance for the ad hoc scenario shown in Figure 7.5(a). The impact of the network connectivity on the mean delay performance can be seen in Figures 7.9(b), 7.10(b) and 7.11(b). With a larger network connectivity, a better delay performance can be achieved. We can see that IEEE 802.11a has a better overall mean delay performance than W-CHAMB as long as it is not saturated. The reason is that IEEE 802.11a can transmit a packet as large as 2304 bytes, whereas W-CHAMB has to segment large packets into 102 byte fragments with the transmission rate of 24 Mbit/s. The fragments are transmitted over a number of W-CHAMB MAC frames as a PDU train. But IEEE 802.11a goes into saturation with a much lower traffic load than W-CHAMB can carry. It is worth mentioning that IEEE 802.11a is not able to differentiate between service classes and the mean delay of all service classes together is a too rough measure to characterize a system.

7.3.2.2 Comparison of Traffic Performance with Various OFDM Schemes using the Same Transmission Power

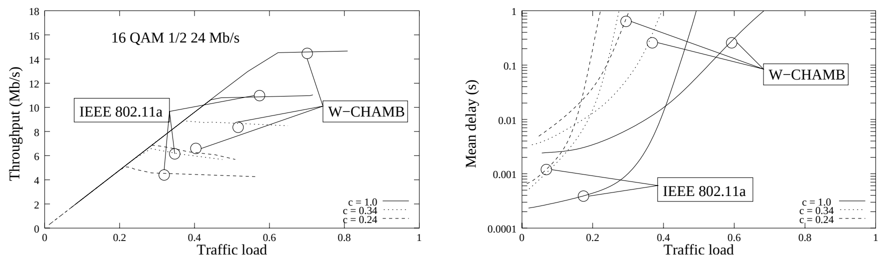
To compare the traffic performance with various OFDM schemes using the same transmission power, the same simulation scenarios used in Section 6.2.5.3 are used for IEEE 802.11a and W-CHAMB.

Figure 7.12 plots the throughput results of both systems. All WSs use the same transmission power and the same transmission rate. As a high C/I



(a) Throughput **(b)** Mean delay

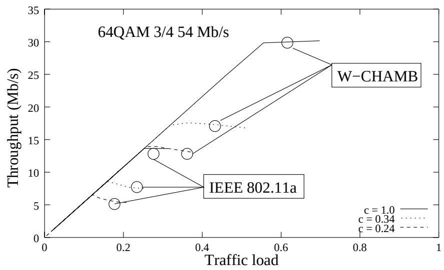
Figure 7.9: Traffic performance with 6 Mbit/s



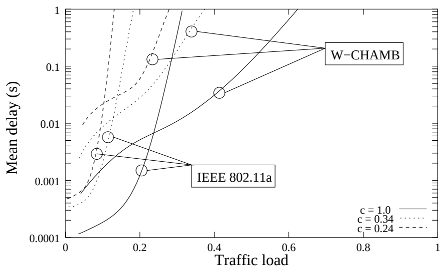
(a) Throughput **(b)** Mean delay

Figure 7.10: Traffic performance with 24 Mbit/s

ratio and a high signal strength are needed to decode a PDU with a high transmission rate, the network connectivity decreases with the increase of the transmission rate. From Figure 7.12 we can see that IEEE 802.11a is extremely inefficient with a high transmission rate due to the small connectivity. On the contrary to W-CHAMB, the maximum throughput with IEEE 802.11a with a transmission rate of 54 Mbit/s is lower than that with a transmission rate of 36 Mbit/s, and the maximum throughput with a transmission rate of 36 Mbit/s is lower than that with a transmission rate of 24 Mbit/s due to the reduced network connectivities. The maximum



(a) Throughput



(b) Mean delay

Figure 7.11: Traffic performance with 54 Mb/s

throughput with IEEE 802.11a at a transmission rate of 24 Mbit/s is the highest (6.5 Mbit/s). From Figure 7.12 we conclude that the maximum throughput (6.5 Mbit/s) with IEEE 802.11a is about 80% less than that with W-CHAMB (11.5 Mbit/s). The mean delay performance is shown in Figure 7.13. IEEE 802.11a has better mean delay performance with a transmission rate of 12 Mbit/s and 36 Mbit/s than that with other transmission rates before the network is saturated. The high network connectivity with a transmission rate of 12 Mb/s is the reason of the low mean delay in this case. The mean delay with 36 Mbit/s is better than that with 24 Mbit/s before the network is saturated as the higher channel capacity is available with a transmission rate of 36 Mbit/s. From the mean delay performance we can also see that IEEE 802.11a with 24 Mbit/s is saturated a little later than with other transmission rates. With a transmission rate of 6 Mbit/s,the traffic performance is worst due to its low channel capacity although the network connectivity is very high in this case. The results indicate that with a low transmission rate the absolute channel capacity is not enough for the broadband application. In comparison with W-CHAMB, IEEE 802.11a has a better delay performance than W-CHAMB at a low traffic load. The reason is just the same as indicated in Section 7.3.2.1. Again it has to be noted that the overall mean delay might be misleading if different service classes with different QoS requirements have to be supported. W-CHAMB has been shown to be substantially superior then to IEEE 802.11a regarding to the QoS provision.

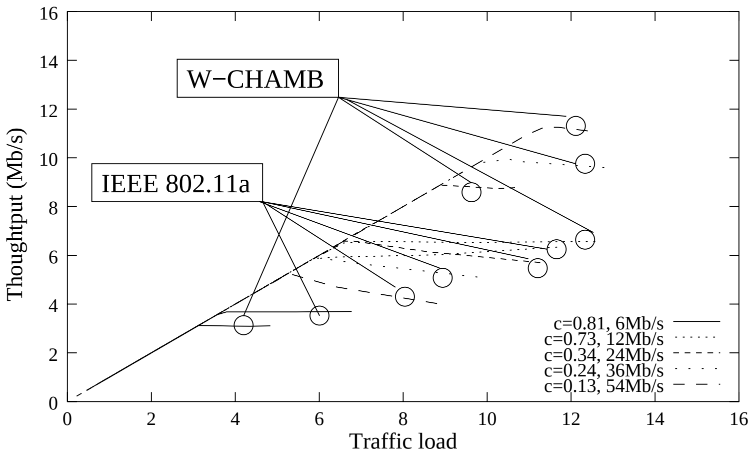


Figure 7.12: Throughput with various OFDM schemes

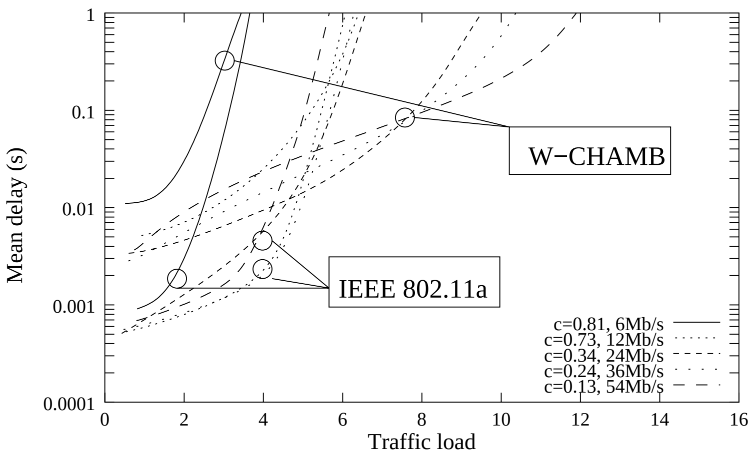


Figure 7.13: Mean delay with various OFDM schemes

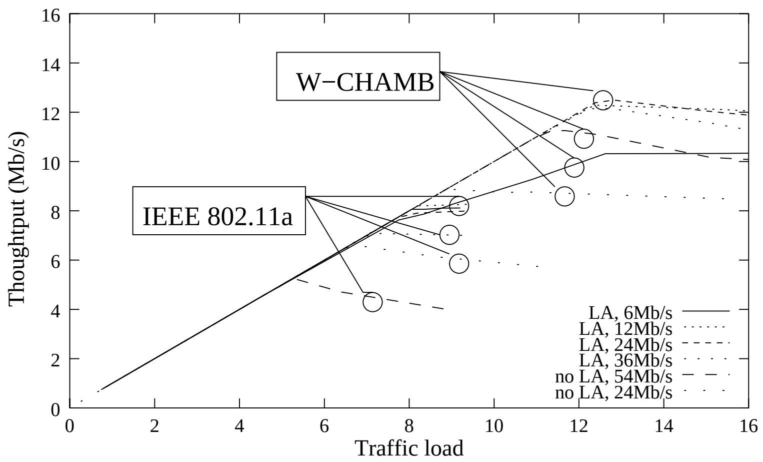


Figure 7.14: Throughput with link adaptation

7.3.2.3 Comparison of the Traffic Performance with Link Adaptation

The traffic performance using the same link adaptation algorithm and scenario as described in Section 6.2.5.4 is shown in Figure 7.14 and 7.15. IEEE 802.11a using link adaptation achieves the best traffic performance with a minimum transmission rate of 12 Mbit/s, whilst W-CHAMB has the best performance with a minimum transmission rate of 24 Mbit/s. The superiority of traffic performance achieved with W-CHAMB over IEEE 802.11a is very significant. The maximum throughput with W-CHAMB is 12.5 Mbit/s, much higher than that with IEEE 802.11a (8 Mbit/s). This means that W-CHAMB can deliver 56 % more traffic than IEEE 802.11a with the same traffic load and network topologies after a link adaptation algorithm is used.

From Figure 7.14 and 7.15 we can see that a performance improvement is achieved using link adaptation for both systems. The maximum throughput with IEEE 802.11a and W-CHAMB can be improved about 24 % and 9 %, respectively. The mean delay performance is also improved accordingly.

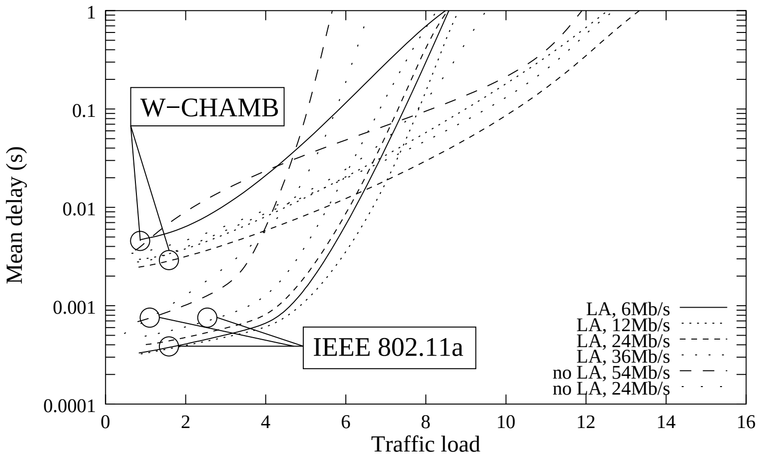
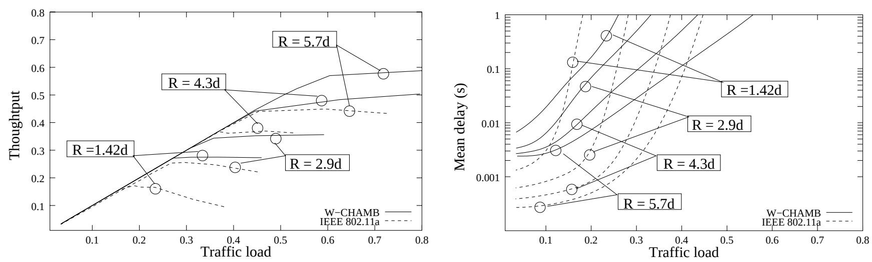


Figure 7.15: Mean delay with link adaptation

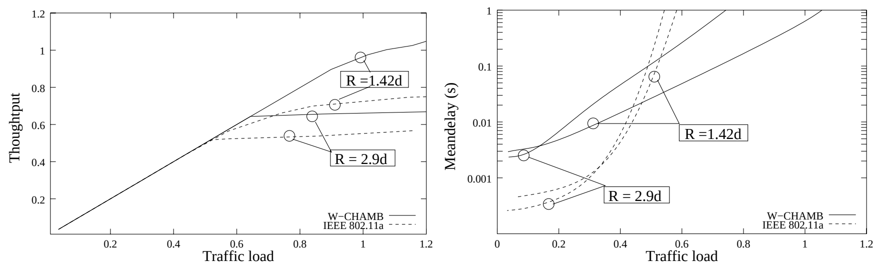
7.4 Access Scenario

The access scenarios described in Section 6.2.6 are studied in this section to compare the traffic performance of W-CHAMB and IEEE 802.11a. The traffic performance with one-AP and multiple APs for both systems, IEEE 802.11a and W-CHAMB, is compared in Figure 7.16 and 7.17. With both systems, the smaller transmit/receive range results in a worse traffic performance in the one-AP case, whilst a reduced transmit/receive range can improve the traffic performance in the multiple APs case due to the more balanced load distribution in the network. In comparison with W-CHAMB, the maximum throughput with IEEE 802.11a is much lower in all cases. In the one-AP scenario with R = 1.42d, 2.9d, 4.3d and 5.7d, the respective maximum throughput with W-CHAMB is 0.28, 0.35, 0.5 and 0.59, much higher than that with IEEE 802.11a (0.18, 0.28, 0.36 and 0.42). In the multiple APs scenario with R = 1.42d, the maximum throughput with W-CHAMB is 1.05, about 70% more than that with IEEE 802.11a (0.65). The mean delay with IEEE 802.11a is lower than that with W-CHAMB at the low traffic load. The reason is the same as explained before.



(a) Throughput **(b)** Mean delay

Figure 7.16: Impact of the network connectivity on the traffic performance with one AP



(a) Throughput **(b)** Mean delay

Figure 7.17: Impact of the network connectivity on the traffic performance with multiple APs

7.5 Summary

The traffic performance with HiperLAN/2, IEEE 802.11a and W-CHAMB has been compared under various network scenarios. HiperLAN/2 with its central control has the best traffic performance in a fully connected small network. But HiperLAN/2 appears not to be suited for a self-organizing multihop network.

DCF of IEEE 802.11a is a decentralized MAC protocol that supports multihop transmission. However, simulation results reveal that DCF is inefficient for packets with short to medium size. Moreover, DCF has no means to guarantee QoS for real-time services in a multihop network where PCF is not applicable.

W-CHAMB with its channel-oriented MAC protocol is able to guarantee QoS for rt-VBR services in a self-organizing multihop network. In addition, with realistic Ethernet packet sizes, W-CHAMB achieves much higher efficiency than IEEE 802.11a. W-CHAMB can deliver 50 % more traffic than IEEE 802.11a in a multihop network. The W-CHAMB MAC protocol appears to be the best suited solution for self-organizing broadband multihop networks.

CHAPTER 8

Conclusions and the future work

The thesis has focused on the protocol design and performance evaluation of the W-CHAMB network air interface for self-organizing wireless broadband multihop networks with QoS guarantee. W-CHAMB featuring broadband, multimedia, multihop, self-organizing and QoS guarantee seems to be one of the most promising solutions for the next generation wireless communication networks.

Self-organizing broadband wireless networks operating at the 5 GHz license-exempt frequency spectrum shall use OFDM schemes as its physical layer to increase the robustness against frequency selective fading or narrow-band interference. W-CHAMB uses the same OFDM schemes as IEEE 802.11a and HiperLAN/2 to achieve a high transmission rates. The radio coverage is extended by multihop transmission without the need to increase the transmission power. In the multiple APs case, multihop transmission improves the traffic performance thanks to the reduction of the interference power.

The responsibilities of organizing and controlling of W-CHAMB networks are fully distributed among the wireless stations themselves. Distributed control algorithms are of importance for the survivability, robustness, and freedom from reliance on any single WS for network control. The channel-oriented MAC protocol that is based on dynamic channel reservation realizes statistical multiplexing of the bursty multimedia traffic with QoS guarantee in a decentralized multihop network. E-signals are used to realize distributed access priorities, to solve the hidden station problem and to achieve a MAC level acknowledgment (ACK) for the fast ARQ scheme. A channel occupation list with RSSI is maintained in each WS for the purpose of dynamic channel reservation. By allowing stations to receive on the TCH with *Hidden* status and to send on the TCH with *Interfered* status, frequency spatial reuse can be increased in a self-organizing network.

The main contributions of this thesis include:

1. Design of new protocols and algorithms for self-organizing broadband multihop networks with QoS guarantee. In comparison with the previous works [15, 79], this thesis emphasizes the following three aspects:

   (a) QoS guarantee for real time traffic services in self-organizing multihop networks. New protocols and algorithms that are used to ensure the QoS of the real time traffic services in self-organizing multihop networks include measurement-based Call Admission Control (CAC) (Section 4.5.3) that is used to limit the real time traffic on the call level, rt-VBR scheduling algorithm (Section 4.4.2.1) that plans the needed bandwidth on the PDU train level, the distributed prioritized access (DPA) algorithm (Section 4.3.2.4) and the ABR interruption procedures (Section 4.3.2.6) that ensure the prompt reservation of rt-VBR RCC with the scheduled bandwidth, and the fast ARQ scheme (Section 4.4.4) that ensures the correct transport of the individual PDU.

   (b) Support of asymmetric multimedia broadband bursty traffic services. Using E-signals to realize the MAC level ACK (Section 4.3.2.3) an RCC can be established for the PDU delivery in one direction. This feature is much suited for the delivery of asymmetric multimedia bursty traffic services (Section 4.3.2.3). About 100% performance gain is achieved using the E-signal concept in comparison with a TDD operation mode. The mean delay using the TDD operation mode is 10 times higher than using the E-signal concept.

   (c) Efficient utilization of the channel resources. Efficient utilization of the scarce and expensive wireless channel resources is always a very important issue in the design of wireless networks. The author proposes the new COL (Section 4.3.2.2) to increase the spatial reuse of the channel resources, the LA algorithm (Section 6.2.5.4) to increase the actual data rate and the Dynamic Channel Allocation (DCA) algorithm (Section 4.4.2.3) to dynamically adjust the bandwidth of ABR RCC to achieve the MAC efficiency as well as fairness. The W-CHAMB IRP (Section 4.5) reduces the routing traffic significantly by avoiding global search of the destination throughout the network.

2. Developing a simulation tool based on the prototypic implementation of protocols in SDL. The simulation tool provides a platform for a detailed investigation of self-organizing multihop networks.

3. Analyzing the W-CHAMB air interface using probability theory and two dimensional Markov Chain method.

   The ASP (Section 6.1.1) is derived using probability theory. The analyzes reveals insights of the relationship among contending WSs, ACC-E-Signals and the way how the access priority is selected. The results indicate that the access priority for best effort services should be chosen according to a geometric distribution. The optimized value of the parameter p of the geometric distribution is 0.94 with a $P_{max} =$ 100 that is large enough in most cases. For real time services, on the other hand, the access priority should be chosen at random.

   A two dimensional Markov Chain is used to model the W-CHAMB MAC protocol. The dynamic behavior of W-CHAMB MAC protocol is described intuitively with the Markov state transition probability. The numerical results provide interesting insights of the W-CHAMB traffic performance. For a W-CHAMB network with 25 WSs a maximum access priority of 10 is not enough due to the low ASP, whilst the maximum access priority of 50 seems to be enough. High efficiency can be achieved for long PDU trains. The throughput performance degrades with short PDU trains due to the bottleneck at the ACH. With a smaller number of TCHs per MAC frame the bottleneck at the ACH can be reduced, whilst the protocol overhead becomes higher if the number of TCHs per MAC frame decreases. The reasonable value of the number of TCHs is 15 if it cannot be adjusted dynamically. The mean access delay and mean PDU delay remain low before the network is saturated. The delay performance degrades significantly when the network approaches saturation. The mean PDU delay is proportional to the mean length of PDU trains.

4. Evaluating the traffic performance of W-CHAMB using computer simulations. The traffic performance of W-CHAMB networks is intensively studied stochastically using the developed simulation tool. The impact of the system parameters, i.e. traffic characteristics, channel errors, protocol variables, network connectivities, transmission rates,

number of APs and etc., on the traffic performance is systematically investigated.

With a small DCA parameter set a better delay performance is achieved with a low traffic load. With the increase of the traffic load a larger parameter set results in a better delay performance. This means that with a high traffic load a larger DCA parameter set is preferred. DCA algorithm achieves the fairness of the MAC protocol without the significant impact on the traffic performance as well.

The bottleneck at the ACH will not appear with short PDU trains if the next PDU train arrives before the previous PDU train is sent and is served immediately after the previous PDU train is delivered. The bottleneck at the ACH will be severe with a small mean length of PDU trains if the traffic load is increased by increasing the number of active WSs.

The simulation results indicate that the error control technology realized by E-Signals is efficient and stable as the degradation of the traffic performance is very limited even under a channel error rate of 10%.

The bandwidth of the required rt-VBR RCC is planned by the rt-VBR scheduling algorithm according to the parameter called transmission duration T_d together with the length of the PDU train. If the parameter T_d is set small, the PDU train will be delivered in a shorter duration. But a small T_d value increases the bandwidth requirement of the rt-VBR RCC. If the network is highly loaded, it will be difficult for a WS to establish an RCC with a large bandwidth requirement. Thus, the access delay will increase accordingly. Simulation results indicate $T_d = 10ms$ seems to be the optimized value. The results also reveal that the traffic load of rt-VBR traffic must be controlled to guarantee QoS for rt-VBR traffic. Usually, the real time traffic load should not exceed 50% in a fully connected W-CHAMB network.

The delay of the rt-VBR traffic under mixed traffic load is under control even at the heavy saturation condition. The delay of rt-VBR PDU with c=0.34 is a little higher than that with c = 1.0 due to multihop transmissions of PDU trains. But the QoS of rt-VBR traffic is guaranteed in the both cases. The difference between the delay of rt-VBR PDUs without ABR traffic load and with a high ABR traffic load is not significant. The high ABR load increases the delay of the rt-VBR PDUs about several milliseconds that are necessary for a WS to interrupt an ABR transmission in favorite of rt-VBR PDU train in the case that no free channels are available at the arrival of an rt-VBR PDU train.

Networks with a connectivity of 1.0 have always the best traffic performance in comparison with networks with a smaller connectivity. With a small connectivity the system capacity is used up rapidly owing to the multihop transmissions needed for multihop connections. The benefits of frequency spatial reuse in multihop networks is adverse to that of cellular networks that have many access points.

Although the relative throughput using an OFDM scheme with a lower transmission rate is higher, OFDM schemes with a very low transmission rate are not recommended for W-CHAMB owing to its limited capacity per frequency carrier. The traffic performance using 6 Mbit/s and 12 Mbit/s is worse than that using other higher transmission rates in almost all cases. OFDM schemes with a transmission rate less than 12 Mbit/s are not suited for W-CHAMB networks. The OFDM scheme with 24 Mbit/s seems to be best suited for W-CHAMB networks with a low to moderate traffic load. A higher transmission rate is useful if the network is heavily loaded. The best traffic performance can be achieved using the link adaptation algorithm where a WS selects an OFDM scheme according to the actual signal strength and the C/I ratio. The very low transmission rate (6 Mbit/s) should not be used as the traffic performance with a one-hop connection using 6 Mbit/s is much worse than that with a multihop connection having a higher transmission rate.

In the access scenario with one AP a reduced communication range (transmission power) degrades the traffic performance in terms of throughput and mean delay due to the bottleneck effect in the AP. The solution to the bottleneck effect in the AP is to place more APs in the access network, forming a cellular-like multihop ad hoc network. With a multi-APs scenario, traffic performance can be significantly improved with a reduced transmission range.

5. Comparing the multihop traffic performance of W-CHAMB with IEEE 802.11a and HiperLAN/2. As HiperLAN/2 adopts a fully centralized scheduler-oriented MAC approach it has the best traffic performance in a fully connected small network. But HiperLAN/2 appears not to be suited for a multihop network. The traffic performance of HiperLAN/2 degrades dramatically with the increase of forwarder terminals due to the large protocol overhead. IEEE 802.11a uses a packet-oriented CSMA/CA MAC solution that is inefficient for short to medium packets, and cannot guarantee QoS in a multihop networking environment where PCF is not applicable. W-CHAMB meets QoS requirements for different classes of service and realizes statistical multiplexing of broadband bursty traffic in a fully distributed and efficient manner. The superiority of the multihop traffic performance of the W-CHAMB can be seen in comparison with that of IEEE 802.11a and HiperLAN/2. With 3 FMT, the maximum throughput with W-CHAMB is about 40 % higher than that with HiperLAN/2. With realistic Ethernet packet sizes, W-CHAMB can deliver 50 % more traffic than IEEE 802.11a in a multihop network.

The results in this thesis are valuable for the development and standardization of the next generation wireless communication systems. W-CHAMB has caused much attention in the communication industry and academic areas. The future work includes prototyping and standardization of the W-CHAMB air interface as well as further design and evaluation of the W-CHAMB network protocol, such as mobility management, interworking and TCP/IP performance.

APPENDIX A

List of Acronyms

ACC	*Access*
ACH	*Access Channel*
ACK	*Acknowledgment*
AP	*Access Point*
ARQ	*Automatic Repeat Request*
ASP	*Access Success Probability*
ATM	*Asynchronous Transfer Mode*
BCH	*Broadband Channel*
BCH	*Broadcast Channel*
BCCH	*Broadcast Control Channel*
BOT	*Begin of the Train*
BRAN	*Broadband Radio Access Network*
BpS	*Byte per Symbol*
BPSK	*Binary Phase Shift Keying*
CAC	*Call Admission Control*
CC	*Central Controller*
CCA	*Clear Channel Assessment*
CF	*Coordination Function*
CFP	*Contention-Free Period*
C/I	*Carrier-to-Interference Ratio*
CL	*Convergence Sublayer*
COL	*Channel Occupation List*
COT	*Continuing of the Train*
CP	*Contention Period*
CPCS	*Common Part Convergence Sublayer*
CRC	*Cyclic Redundancy Check*
CS	*Carrier Sensing*

CSMA/CA	*Carrier Sense Multiple Access with Collision Avoidance*
CSMA/CD	*Carrier Sense Multiple Access with Collision Detection*
CTS	*Clear To Send*
CW	*Contention Window*
D_{acc}	*Access Deadline*
DCA	*Dynamic Channel Allocation*
DCF	*Distributed Coordination Function*
DCR	*Dynamic Channel Reservation*
DECT	*Digital Enhanced Cordless Telecommunications*
DEST	*Destination*
DFS	*Dynamic Frequency Selection*
DIFS	*Distributed (Coordination Function) Interframe Space*
DiL	*Direct Link*
DFT	*Discrete Fourier Transformation*
DL	*Downlink*
DLC	*Data Link Control*
DPR	*Distributed Priority Access*
ECH	*Energy Channel*
EIRP	*Equivalent Isotropic Radiated Power*
EOT	*End of the Train*
EP	*End Point*
E-Signal	*Energy Signal*
ETSI	*European Telecommunications Standards Institute*
EY-NPMA	*Elimination Yield Non Preemptive Priority Multiple Access*
FCC	*Federal Communications Commission*
FCCH	*Frame Control Channel*
FCH	*Frame Channel*
FCS	*Frame Check Sequence*
FDMA	*Frequency Division Multiple Access*
FEC	*Forward Error Correction*
FER	*Frame Error Ratio*

FHSS	*Frequency Hopping Spread Spectrum*
FFT	*Fast Fourier Transformation*
FIFO	*First In First Out*
FMT	*Forwarding Mobile Terminal*
GPRS	*General Packet Radio Service*
GOP	*Group of Pictures*
GUI	*Graphic User Interface*
H/2	*HIPERLAN/2*
HEE	*Home Environment Extension*
HiperLAN	*High Performance Radio Local Area Network*
HiperLAN/2	*High Performance Radio Local Area Network Type 2*
IBSS	*Independent Basic Service Set*
IEEE	*Institute of Electrical and Electronics Engineers*
IFFT	*Inverse Fast Fourier Transformation*
IFS	*Interframe Space*
IMT-2000	*International Mobile Telecommunications at 2000 MHz*
IP	*Internet Protocol*
IRP	*Iterative Routing Protocol*
ISDN	*Integrated Services Digital Network*
ISM	*Industrial, Scientific and Medical*
ISO	*International Standards Organization*
ITU	*International Telecommunication Union*
LA	*Link Adaptation*
LAN	*Local Area Network*
LBT	*Listen Before Talk*
LCCH	*Link Control Channel*
LCH	*Long Channel*
LLC	*Logical Link Control*
LOS	*Line Of Sight*
MAC	*Medium Access Control*
MPDU	*MAC Protocol Data Unit*
MS	*Mobil station*

MSAP	*MAC Service Access Point*
MSDU	*MAC Service Data Unit*
MT	*Mobile Terminal*
NAV	*Net Allocation Vector*
NL	*Network Layer*
NECH	*Narrow-band Energy Channel*
NT	*Neighbour Table*
OFDM	*Orthogonal Frequency Division Multiplexing*
OSI	*Open Systems Interconnection*
PA	*Priority Assertion*
PC	*Power Control*
PC	*Point Coordinator*
PCF	*Point Coordination Function*
PCI	*Protocol Control Information*
PDU	*Protocol Data Unit*
PER	*Packet Error Ratio*
PHY	*Physical Layer*
PIFS	*Point (Coordination Function) Interframe Space*
PPDU	*PHY PDU*
QAM	*Quadrature Amplitude Modulation*
QoS	*Quality of Service*
QPSK	*Quadrature Phase Shift Keying*
RCC	*Real Channel Connection*
RES	*Radio Equipment and Systems*
RFCH	*Random access Feedback Channel*
RG	*Resource Grant*
RG-IE	*Resource Grant Information Element*
RLC	*Radio Link Control*
RMT	*Remote Mobile Terminal*
RR	*Resource Request*
RRC	*Radio Resource Control*
RSS	*Received Signal Strength*
RSSI	*Receive Signal Strength Indicator*
RTS	*Ready To Send*
RTS/CTS	*Ready To Send/Clear to Send*

rt-VBR	*real time Variable Bit Rate*
SAP	*Service Access Point*
SAR	*Segmentation And Reassembly*
SCH	*Short Channel*
SDL	*System Description Language*
SDL-GR	*SDL Graphical Representation*
SDL-PR	*SDL Phrase Representation*
SDU	*Service Data Unit*
SIFS	*Short Interframe Space*
SINR	*Signal to Interferer-Noise Ratio*
SN	*Sequence number*
SOT	*Single PDU of the Train*
s-PDU	*signaling PDU*
SR	*Selective Repeat*
SSCS	*Service Specific Convergence Sublayer*
SSCS-PDU	*Service Specific Convergence Sublayer PDU*
STA	*Station*
TCH	*Traffic Channel*
TDD	*Time Division Duplex*
TDMA	*Time Division Multiple Access*
TTA	*Transceiver Turn-Around*
TSF	*Timing Synchronization Function*
UL	*Uplink*
U-NII	*Unlicensed National Information Infrastructure*
UMTS	*Universal Mobile Telecommunications System*
USA	*United States of America*
VBR	*Variable Bit Rate*
VC	*Virtual Connection*
W-CHAMB	*Wireless CHannel Oriented Ad hoc Multihop*
WLAN	*Wireless Local Area Network*
WS	*Wireless Station*
WT	*Wireless Terminal*

ZRP *Zone Routing Protocol*

[1] ITU-T Rec. I. 371. *Traffic Control and Congestion Control in B-ISDN.* Recommandation, IRU-T, Perth, U.K., Nov. 1995.

[2] N. Abramson. *The Throughput of Packet Broadcasting Channels.* IEEE Transactions on Communications, Vol. Com-25, No. 1, pp. 117–128, January 1977.

[3] A. S. Acampora, M. Naghshineh. *An Architecture and Methodology for Mobile-Executed Handoff in Cellular ATM Networks.* IEEE Journal on Selected Areas in Communications, Vol. 12, No. 8, pp. 1365–1375, October 1994.

[4] AixCom. *SDL Performance Evaluation Environment Toolkit SPEET, SPEET Class Library.* `www.AixCom.de`.

[5] A. Alwan, R. Bagrodia, N. Bambos, M. Gerla, L. Kleinrock, J. Short, J. Villasenor. *Adaptive Mobile Multimedia Networks.* IEEE Personal Communications Magazine, Vol. 3, No. 2, pp. 34–51, April 1996.

[6] M. E. Anagnostou, E. N. Protonotarios. *Performance Analysis of the Selective Repeat ARQ Protocol.* IEEE Transactions on Communications, Vol. 34, No. 2, pp. 127–135, February 1986.

[7] G. Anastasi, L. Lenzini, E. Mingozzi. *HiperLAN/1 MAC Protocol: Stability and Performance Analysis.* IEEE Journal on Selected Areas in Communications, Vol. 18, No. 9, pp. 1787–1798, September 2000.

[8] G. Anastasi, L. Lenzini, E. Mingozzi, A. Hettich, A. Krämling. *MAC Protocols for Wideband Wireless Local Access: Evolution Toward Wireless ATM.* IEEE Personal Communciations, Vol. 5, No. 5, pp. 53–64, October 1998.

[9] Mitschele-Thiel Andreas. *Systems Engineering with SDL*. John Wiley & Sons, Ltd, New York, 2000.

[10] K. Aretz, M. Haardt, W. Konhäuser, W. Mohr. *The Future of Wireless Communications*. Computer Networks, Vol. 37, No. 1, pp. 83–92, September 2001.

[11] J.J. Bae, T. Suda. *Survey of Traffic Control Schemes and Protocols in ATM Networks*. Proceedings of the IEEE, Vol. 79, No. 2, pp. 170–189, February 1991.

[12] F Belina, D. Hogrefe. *The CCITT-Specification and description Language SDL*. Computer Networks and ISDN Systems, Vol. 16, No. 4, pp. 311–341, March 1989.

[13] Bellcore (now Telcordia). *Ethernet Tracefile pAug.TL*. `ftp.telcordia.com/pub/world/wel/lan_traffic/`.

[14] D. Bertsekas, R. Gallager. *Data Networks*. Prentice-Hall, Englewood Cliffs, New Jersey, 2. Edition, 1992.

[15] S. Böhmer. *Entwurf eines ATM-basierten Funknetzes und Software-Entwurfs-methodik zur Implementierung*. PHD Thesis, RWTH Aachen, ISBN 3-86073-384-2, 1996.

[16] G. Bianchi. *Performance Analysis of the IEEE 802.11 Distributed Coordination Function*. IEEE Journal on Selected Areas in Communications, Vol. 18, No. 3, pp. 535–547, March 2000.

[17] J.A. Bingham. *Multicarrier Modulation for Data Transmission: An Idea Whose Time Has Come*. IEEE Communications Magazine, Vol. 28, No. 5, pp. 5–14, May 1990.

[18] Broadband Radio Access Networks (BRAN). *HIPERLAN Type 2; Cell based Convergence Layer; Part1: Common Part*. Standard TS 101 763-1, ETSI, April 2000.

[19] Broadband Radio Access Networks (BRAN). *HIPERLAN Type 2; Cell based Convergence Layer; Part2: UNI Service Specific Convergence Sublayer (SSCS).* Standard TS 101 763-2, ETSI, April 2000.

[20] Broadband Radio Access Networks (BRAN). *HIPERLAN Type 2; Data Link Control (DLC) Layer; Part 1: Basic Transport Functions.* Standard TS 101 761-1, ETSI, February 2001.

[21] Broadband Radio Access Networks (BRAN). *HIPERLAN Type 2; Data Link Control (DLC) Layer; Part 2: Radio Link Control (RLC) Sublayer.* Standard TS 101 761-2, ETSI, September 2000.

[22] Broadband Radio Access Networks (BRAN). *HIPERLAN Type 2; Data Link Control (DLC) Layer; Part 4: Extension for Home Environment.* Standard TS 101 761-4, ETSI, June 2000.

[23] Broadband Radio Access Networks (BRAN). *HIPERLAN Type 2; Packet based Convergence Layer; Part1: Common Part.* Standard TS 101 493-1, ETSI, April 2000.

[24] Broadband Radio Access Networks (BRAN). *HIPERLAN Type 2; Packet based Convergence Layer; Part2: Ethernet Service Specific Convergence Sublayer (SSCS).* Standard TS 101 493-2, ETSI, December 2000.

[25] Broadband Radio Access Networks (BRAN). *HIPERLAN Type 2; Packet based Convergence Layer; Part3: IEEE 1394 Bridge Layer.* Standard TS 101 493-4, ETSI, January 2001.

[26] Broadband Radio Access Networks (BRAN). *HIPERLAN Type 2; Packet based Convergence Layer; Part3: IEEE 1394 Service Specific Convergence Sublayer (SSCS).* Standard TS 101 493-3, ETSI, January 2001.

[27] Broadband Radio Access Networks (BRAN). *HIPERLAN Type 2; Physical (PHY) layer.* Standard TS 101 475, ETSI, February 2001.

[28] Broadband Radio Access Networks (BRAN). *HIPERLAN Type 2; System Overview.* Technical Report TR 101 683, ETSI, February 2000.

[29] F. Cali, M. Conti, E. Gregori. *IEEE 802.11 protocol: Design and performance Evaluation of an Adaptive Backoff Mechanism.* IEEE Journal on Selected Areas in Communications, Vol. 18, No. 9, pp. 1774–1786, September 1998.

[30] C.-S. Chang, K.-C. Chen, M.-Y. You, J.-F. Chang. *Guaranteed Quality-of-Service Wireless Access to ATM Networks.* IEEE Journal on Selected Areas in Communications, Vol. 15, No. 1, pp. 106–118, January 1997.

[31] Y. L. Chang, C. C. Hsu. *Routing in Wireless/Mobile ad hoc networks via dynamic group construction* . Mobile Networks and Applications, Vol. 5, No. 1, pp. 27–37, April 2000.

[32] T. Chen, S. Liu, V. Samalam. *The Available Bit Rate Service for Data in ATM Networks.* IEEE Communications Magazine, Vol. 34, No. 5, pp. 56 – 71, May 1996.

[33] K.L. Chung. *Markov Chains with Stationary Transition Probabilities.* Springer-Verlag, Berlin, 1967.

[34] L.J.JR. Cimini. *Analysis and Simulation of a Digital Mobile Channel using Orthogonal Frequency Division Multiplexing.* IEEE Transactions on Communications, Vol. 33, No. 7, pp. 665–675, July 1985.

[35] J. Deng, Z. Haas. *Dual Busy Tone Multiple Access (DBTMA): A New Medium Access Control for Packet Radio Networks.* pp. 973–978, Florence, Italy, October 1998.

[36] J. Dunlop, J. Irvine, D. Robertson, P Cosimini. *Performance of a Statistically multiplexed Access Mechanism for a TDMA Radio Interface.* IEEE Personal Communications, Vol. 2, No. 3, pp. 56–64, June 1995.

[37] K.Y. Eng, M.J. Karol, M. Veeraraghavan, E. Ayanoglu, C.B. Woodworth, P. Pancha, R.A Valenzuela. *A wireless broadband ad-hoc ATM local-area network.* Wireless Networks, Vol. 1, No. 1, pp. 161 – 174, February 1995.

[38] N. Esseling, H. S. Vandra, B. Walke. *A Forwarding Concept for HiperLAN/2.* Computer Networks, Vol. 37, No. 1, pp. 25–32, September 2001.

[39] ETSI. *Digital European Cordless Telecommunications (DECT)*, 1992. prETS 300 175-1..7.

[40] ETSI. *Radio Equipment and Systems (RES); High PErformance Radio Local Networks (HiperLANs) Type 1 Functional specification.* Standard EN 300 652, ETSI, Sophia Antipolis, France, 1998.

[41] European Radiocommunications Committee. *Decision on the harmonised frequency bands to be designated for the introduction of High Performance Radio Local Area Networks (HIPERLANs).* Technical Report ERC/DEC/(99)23, ERC, November 1999.

[42] R. Fantacci. *Queuing Analysis of the Selective Repeat Automatic Repeat Request Protocol for Wireless Networks.* IEEE Transactions on Vehicular Technology, Vol. 45, No. 2, pp. 256–264, May 1996.

[43] R. Fantacci, M. Scardi. *Performance Evaluation of preemptive Polling Schemes and ARQ Techniques for Indoor Wireless Networks* . IEEE Transactions on Vehicular Technology, Vol. 45, No. 2, pp. 248–257, May 1996.

[44] Federal Communications Commision. *Rules to Provide for Operation of Unlicensed NII Devices in the 5 GHz Frequency Range.* Report and Order 96-102, FCC, 1997.

[45] O. Feegemand, A. Olsen. *Introduction to SDL-92.* Computer Networks and ISDN Systems, Vol. 26, No. 9, pp. 1143–1167, May 1994.

[46] M. W. Garrett. *MPEG Tracefile MPEG.data.* `ftp.telcordia.com/pub/vbr.video.trace/`.

[47] M. Gerla, J. Tzu-Chieh Tsai. *Multicluster, mobile, multimedia radio network.* Wireless Networks, Vol. 1, No. 3, pp. 255–265, October 1995.

[48] D.-J. Goodman, S.-X. Wei. *Efficiency of Packet Reservation Multiple Access.* IEEE Trans. on Vehicular Technology, Vol. 40, No. 1, pp. 170–176, February 1991.

[49] C. Görg. *Stochastische Simulationstechnik.* Lecture Manuscript, RWTH Aachen, 1998.

[50] C. Görg. *Verkehrstheoretische Modelle und stochastische Simulationstechniken zur Leistunsanalyse von Kommunikationsnetzen.* Nummer 13 in Aachener Beiträge zur Mobil- und Telekommunikation. RWTH Aachen, ISBN 3-86073-662-1, 1997.

[51] R. Guérin, H. Ahmadi, N. Naghshineh. *Equivalent capacity and its application to bandwidth allocation in high-speed networks.* IEEE Journal on Selected Areas in Communications, Vol. 9, No. 7, pp. 968–981, September 1991.

[52] J. C. Haartsen. *The Bluetooth Radio System.* IEEE Personal Communications, Vol. 7, No. 1, pp. 28–36, February 2000.

[53] Z.J. Haas, M.R. Pearlman. *Zone Routing Protocol (ZRP) for Ad Hoc Networks.* Technical report, Internet Engineering Tasks Force (IETF), November 1997.

[54] J. Habetha, M. Nadler. *Concept of a Wireless Centralised Multihop Ad hoc Network.* Computer Networks, Vol. 37, No. 1, pp. 63–71, September 2001.

[55] R. Händel, M.-N. Huber. *Integrated Broadband Networks: An Introduction to ATM–Based Networks.* Addison–Wesley, 1991.

[56] R. Händel, M.-N. Huber, S. Schröder. *ATM Networks: Concepts, Protocols, Applications.* Addison–Wesley, 1994.

[57] A. Hettich. *Leistungsbewertung der Standards HIPERLAN/2 und IEEE 802.11 für drahtlose lokale Netze.* PHD Thesis, RWTH Aachen, ISBN 3-86073-824-0, 2001.

[58] J. Y. Hui. *Resource allocation for broadband networks.* IEEE Journal on Selected Areas in Communications, Vol. 6, No. 9, pp. 1598–1608, December 1988.

[59] S. Hulyalkar, C. Ngo, Y. Du, A. Hettich. *Contention-free MAC Protocol for Wireless ATM LAN.* In *ACTS Mobile Communications Summit*, pp. 728–733, Aalborg, Dänemark, October 1997.

[60] IEEE. *High Speed Physical Layer in the 5 GHz band.* Draft Supplement to Standard IEEE 802.11, IEEE, New York, 1999.

[61] IEEE. *Wireless LAN Medium Access Control (MAC) and Physical Layer (PHY) Specifications.* Standard IEEE 802.11, IEEE, New York, Nov. 1997.

[62] ITU-T. *Recommendation Z-100, Specification and Description Language SDL.* ITU-T, 1994.

[63] R. Jain, S. A. Routhier. *Packet Trains — Measurements and a New Model for Computer Network Traffic.* IEEE Journal on Selected Areas in Communications, Vol. 4, No. 6, pp. 986–995, September 1986.

[64] M. Junius, C. Görg. *Ein Programmpaket zur ereignisgesteuerten Simulation in C++.* In *Messung, Modellierung und Bewertung von Rechen– und Kommunikationssystemen, KWV*, pp. 204–215. Communication Networks, University Aachen, March 1993.

[65] M.J. Karol, L. Zhao, K.Y. Eng. *An efficient demand-assignment multiple access protocol for wireless packet (ATM) networks.* Wireless Networks, Vol. 1, No. 2, pp. 267 – 279, July 1995.

[66] I. Katzela, M. Naghshineh. *Channel Assignment Schemes for Cellular mobile Telecommunication Systems: A Comprehensive survey.* IEEE Personal Communications, Vol. 3, No. 3, pp. 10–31, June 1996.

[67] H. Kerner. *Rechnernetze nach OSI.* Addison-Wesley, Bonn, New York, 2. Edition, 1993.

[68] P. Key. *Modelling, measurement and connection admission control in ATM networks.* In *Proc. of 9th ITC Specialist Seminar Teletraffic Modeling and Measurement in Broadband and Mobile Communication*, pp. 273–291, KPN Research, November 1995.

[69] J. Khun-Jush, P. Schramm, F. Wachsmann, U. Wenger. *Structure and Performance of the HiperLAN/2 Physical Layer.* In *Vehicular Technology Conference 1999-Fall*, pp. 2667–2671, Amsterdam, Niederlande, September 1999.

[70] L. Kleinrock, F. Tobagi. *Packet Switching in Radio Channels: Part I- Carrier sense multiple-Access Modes and Their Throughput-Delay Characteristics.* IEEE Transactions on Communications, Vol. COM-23, No. 12, pp. 1400–1415, December 1975.

[71] Leonard Kleinrock. *Queuing Systems — Computer Applications*, Vol. 2. Wiley & Sons Ltd., New York, 1976.

[72] Leonard Kleinrock. *Queuing Systems — Theory*, Vol. 1. Wiley & Sons Ltd., New York, 1975.

[73] G. K. Konheim. *A Queuing Analysis of Two ARQ Protocols.* IEEE Transactions on Communications, Vol. 28, No. 7, pp. 1004–1014, July 1980.

[74] S. S. LAM. *Packet Broadcast Networks-A Performance Analysis of the R-ALOHA Protocol.* IEEE Transactions on Computers, Vol. C-29, No. 7, pp. 596–603, July 1980.

[75] S.S. Lam, G. G. Xie. *Burst Scheduling Networks.* Performance Evaluation, Vol. 31, No. 1-2, pp. 133–157, November 1997.

[76] M Leiner, D. L. Nielson, F. A. (Guest Editor) Tobagi. *Special Issue on Packet Radio networks.* Proceedings of the IEEE, Vol. 75, No. 1, January 1987.

[77] L. Lenzini, B. Meini, E. Mingozzi. *An efficient numerical Method for performance Analysis of Contention MAC Protocols : A Case Study (PRMA++).* IEEE Journal on Selected Areas in Communications, Vol. 16, No. 5, pp. 653–667, June 1998.

[78] C. R. Lin, J. S. Liu. *QoS Routing in Ad hoc wireless Networks.* IEEE Journal on Selected Areas in Communications, Vol. 17, No. 8, pp. 1426–1438, August 1997.

[79] M. Lott. *Entwurf eines drahtlosen multihop Ad-hoc-Funknetzes mit Dienstgüteunterstützung.* PHD Thesis, RWTH Aachen, ISBN 3-89653-853-5, 2001.

[80] M. Lott. *The Indoor Radio Channel at 5.2 GHz: Prediction by means of Ray Tracing and Measurements.* In *50th IEEE Vehicular Technology Conference (VTC'99 fall)*, pp. 2293–2297, Amsterdam, Niederlande, September 1999.

[81] M. Lott, B. Walke. *Performance Analysis of a Wireless Ad hoc Network with QoS Support.* Telecommunication Systems, Vol. 16, No. 1, 2, pp. 115–134, 2001.

[82] B. Maglaris, D. Anastrassiou, P. Sen, G. Karlsson, J. D. Robbins. *Performance Models of Statistical Multiplexing in Packet Video Communications.* IEEE Trans. on Comm., Vol. 36, No. 7, pp. 834–844, 1988.

[83] J. M. Mapone, I. Stavrakakis. *Determining the Call Admission Region for Real-Time Heterogeneous Applications in Wireless TDMA Networks.* IEEE Network, Vol. 12, No. 2, pp. 38–47, April 1998.

[84] L. Merakos, E. Zervas. *Packet Dropping Analysis for Reservation Voice TDMA Systems.* IEEE Transactions on Vehicular Technology, Vol. 48, No. 4, pp. 1270–1276, July 1999.

[85] N. Nakajima, Y. Yamao. *Development for 4th generation mobile communication.* Wireless Communications and Mobile Computing, Vol. 12, No. 8, pp. 1365–1375, October 1994.

[86] S. Nanda. *Stability Evaluation and Design of the PRMA Joint Voice Data System.* IEEE Transactions on Communications, Vol. 42, No. 5, pp. 2092–2104, May 1994.

[87] S. Nanda, D. J. Goodman, Timor U. *Performance of PRMA: A Packet Voice Protocol for Cellular Systems.* IEEE Transactions on Vehicular Technology, Vol. 40, No. 3, pp. 584–598, August 1991.

[88] J. Naylon, D. Gilmurray, J. Porter, A. Hopper. *Low-Latency Handover in a Wireless ATM LAN.* IEEE Journal on Selected Areas in Communications, Vol. 16, No. 6, pp. 909–921, August 1998.

[89] Y. Ofek. *Generating a Fault-Tolerant Global Clock Using High-Speed Control Signals for the MetaNet Architecture.* IEEE Transactions on Communications, Vol. 42, No. 5, pp. 2179–2188, May 1994.

[90] J. K. Ousterhout. *Tcl und Tk.* Addison-Weslew Publishing Company, Bonn, 1995.

[91] J. Peetz. *A Concept for Interconnection HiperLAN/2 Ad hoc Subnets on Different Frequency Channels.* In 4^{th} *European Personal Mobile Communication Conference (EPMCC)*, Wien, Österreich, February 2001. Session 21.

[92] C. Perkins. *Ad hoc on demand distance vector (AODV) routing.* Technical report, Internet Engineering Tasks Force (IETF), November 1997.

[93] D. Petras. *Entwicklung und Leistungsbewertung einer ATM-Funkschnittstelle.* PHD Thesis, RWTH Aachen, ISBN 3-86073-627-2, 1999.

[94] D. Petras, A. Hettich, A. Krämling. *Design Principles for a MAC Protocol of an ATM Air Interface.* In *ACTS Mobile Summit 1996*, pp. 639–646, Granada, Spanien, November 1996.

[95] F. L. Presti, V. Grassi. *Markov analysis of the PRMA protocol for local wireless networks.* Wireless Networks, Vol. 4, No. 4, pp. 297–306, December 1998.

[96] J. B. Punt, D. Sparreboom, F Brouwer, R. Prasad. *Mathematical Analysis of Dynamic Channel Selection in Indoor Mobile Wireless Communication Systems.* IEEE Transactions on Vehicular Technology, Vol. 47, No. 4, pp. 1302–1313, November 1998.

[97] M. Reiser. *Performance Evaluation of Data Communication Systems.* Proc. of the IEEE, Vol. 70, No. 2, pp. 171–196, February 1982.

[98] E.M. Royer, C-K Toh. *A Review of Current Routing Protocols for Ad Hoc Mobile Wireless Networks.* IEEE Personal Communications, Vol. 6, No. 2, pp. 46–55, April 1999.

[99] J. Sanchez, R. Martinez, M. Marcellin. *A Survey of MAC Protocols Proposed for Wireless ATM.* IEEE Network, Vol. 11, No. 6, pp. 52 – 62, November 1997.

[100] F. Schreiber. *Effective control of simulation runs by a new evaluation algorithm for correlated random sequences.* In *Proc. 12th Int. Teletraffic Congr. (ITC)*, pp. 4.3B.1.1–9, Torino, 1988.

[101] J. L. Sobrinho, A. S. Krishnakumar. *Quality-of-Service in Ad Hoc Carrier Sense Multiple Access Wireless.* IEEE Journal on Selected Areas in Communications, Vol. 17, No. 8, pp. 1353 – 1368, August 1999.

[102] E. Sourous. *Time Slot Assignment Techniques for TDMA Digital cellular Systems.* IEEE Transactions on Vehicular Technology, Vol. 43, No. 1, pp. 121–127, February 1994.

[103] Steppler, M. *Performance Analysis of Communication Systems Formally Specified in SDL*. In *Proc. First Int. Workshop on Software and Performance WOSP 98*, pp. 49–62, Santa Fe, USA, Oct. 1998.

[104] J. Stewart. *Introduction to the Numerical Solution of Markov Chains.* Princeton University Press, Princeton, 1994.

[105] W. Szpankowski. *Analysis and Stability Considerations in a Reservation multiaccess System.* IEEE Transactions on Communications, Vol. COM-31, No. 5, pp. 684–692, May 1983.

[106] S. Tasaka. *Stability and Performance of the R-ALOHA Packet Broadcast System.* IEEE Transactions on Computers, Vol. C-32, No. 8, pp. 717–726, August 1983.

[107] F. Tobagi, L. Kleinrock. *Packet Switching in Radio Channels: Part II- The Hidden Terminal Problem in Carrier-Sense Multiple Access and the Busy-Tone Solution.* IEEE Transactions on Communications, Vol. COM-23, No. 12, pp. 1417–1433, December 1975.

[108] C.-K. Toh. *Wireless ATM and ad-hoc networks.* Kluwer Academic Publishers, 1997.

[109] A. Urie, M. Streeton, C. Mourot. *An Adavanced TDMA Mobile Access System for UMTS.* IEEE Personal Communications, Vol. 2, No. 1, pp. 38–47, February 1995.

[110] R. van Nee. *OFDM for wireless multimedia communications.* Artech House, Boston, 2000.

[111] M. Veeraraghavan, M. Karol, K.Y. Eng. *Mobility and connection management in a wireless ATM LAN.* IEEE Journal on Selected Areas in Communications, Vol. 15, No. 1, pp. 50–68, January 1997.

[112] R. Verdone. *Multihop R-ALOHA for Intervehicle Communications at Millimeter Waves.* IEEE Transactions on Vehicular Technology, Vol. 46, No. 4, pp. 992–1005, November 1997.

[113] B. Walke. *Kommunikationsnetze und Verkehrstheorie I.* Lecture Manuscript, RWTH Aachen, Aachen, 1993.

[114] B. Walke. *Kommunikationsnetze und Verkehrstheorie II.* Lecture Manuscript, RWTH Aachen, Aachen, 1993.

[115] B. Walke. *Mobile Radio Networks - Networking, Protocols and Traffic Performance*, chapter 18. Wiley & Sons, Chichister, New York, 2. Edition, 2001.

[116] B. Walke. *Mobilfunknetze und ihre Protokolle - Band 1.* Teubner, Stuttgart, 3. Edition, 2001.

[117] B. Walke. *Mobilfunknetze und ihre Protokolle - Band 2.* Teubner, Stuttgart, 2000.

[118] B. Walke, S. Böhmer, M. Lott. *Protocols for a Wireless-ATM Multihop Network.* In *International Zurich Seminar*, pp. 75–82, February 1998.

[119] B. Walke, R. Briechle. *A Local Cellular Radio Network(LCRN) for digital voice and data transmission at 60 GHz.* Cellular & Mobile Comm., Online, pp. 215–225, November 1985.

[120] B. Walke, N. Esseling, J. Habetha, A. Hettich, A. Kadelka, S. Mangold, J. Peetz, U. Vornefeld. *IP over Wireless Mobile ATM - Guaranteed Wireless QoS by HIPERLAN/2.* Proceedings of the IEEE, Vol. 89, No. 1, pp. 21–40, January 2001.

[121] B. Walke, D. Petras, D. Plassmann. *Wireless ATM: Air Interface and Network Protocols of the Mobile Broadband System.* IEEE Personal Communications Magazine, Vol. 3, No. 4, pp. 50–56, August 1996.

[122] B. Walke, B. Xu. *Performance Evaluation of MAC Protocols for Self-organizing Broadband Multihop Multimedia Wireless Networks.* In *Proc. of IEEE VTC´01*, pp. Section T34.3, Atlantic City, U.S.A., October 2001.

[123] B. Walke, B. Xu. *Self-Organizing W-ATM Multihop Networks with QoS Guarantee (invited).* In *Proc. of 3GIS*, pp. 1–7, Athens, Greece, Juli 2001.

[124] J. Walrand. *An Introduction to Queuing Networks.* Prentice-Hall International, Inc, New Jersey, 1988.

[125] E. J. Weldon. *An Improved Selective-Repeat ARQ Strategy.* IEEE Transactions on Communications, Vol. 30, No. 3, pp. 480–486, March 1982.

[126] B. Xu, B. Walke. *A Multiple Access Protocol for Ad-hoc Wireless ATM Multihop Networks.* In *Proc. of IEEE VTC´99*, pp. 1141–1145, Houston, U.S.A., May 1999.

[127] B. Xu, B. Walke. *A New Air Interface Concept for Wireless Multimedia Communications beyond the 3rd Generation.* In *Proc. of IEEE WPMC´01*, pp. 473–478, Aalborg, Denmark, September 2001.

[128] B. Xu, B. Walke. *Connection Control and Mobility Management in an Ad-hoc Wireless ATM Multihop LAN.* In *Proc. of the PIMRC 99*, pp. 275–279, Osaka, Japan, IEEE, Sep. 1999.

[129] B. Xu, B. Walke. *Design Issues of Self-Organizing Broadband Wireless Networks.* Computer Networks, Vol. 37, No. 1, pp. 73–81, September 2001.

[130] B. Xu, B. Walke. *Multihop Traffic Performance of IEEE 802.11a, HiperLAN/2 and W-CHAMB.* In *Proc. of 10th Aachen Symposium on Signal Theory -Algorithms and Software for Mobile Communications*, pp. 147–154, Aachen, Germany, September 2001.

[131] B. Xu, B. Walke. *Protocols and Algorithms Supporting QoS in an Ad hoc Wireless ATM Multihop Network.* In *Proceedings of the 3rd European Personal Mobile Communications Conference (EPMCC'99)*, pp. 79–84, Paris, France, March 1999.

[132] B. Xu, B. Walke. *W-CHAMB: A Wireless Channel-oriented Ad-hoc Multihop Broadband Network.* In *Proc. European Wireless'99*, pp. 409–414, Munich, Germany, October 1999.

[133] Wei Zhu. *Synchronisation mobiler Stationen in Nahbereichsfunknetzen.* PHD Thesis, RWTH Aachen, ISBN 3-18-330210-1, 1994.

ACKNOWLEDGMENTS

The presented work has been done during my activity as research assistant at the chair of communication networks of Aachen University of Technology.

I am particularly grateful to my advisor, Prof. Dr.-Ing. Bernhard Walke. His kind support, detailed guidance, thought-provoking discussion, and broad knowledge in the field of mobile communications made this work possible. Special thanks also to Prof. Dr. rer. nat. habil. Carmelita Görg for her kind offer as reviewer and many valuable suggestions.

I would like to thank all members at the chair of communication networks. They have directly or indirectly provided stimulation and support to my work.

Finally, I would like to thank my wife Liyuan and my daughter Jieer for their love and support to my work.

Aachen, im September 2002 Bangnan Xu

CURRICULUM VITAE

24. October 1966	Born in Zhejiang, PR China
1972–1977	Primary school
1977–1982	High School
1982–1989	Study at the Dalian Maritime University
1989–1996	Lecturer at the Dalian Maritime University
1996–2001	Research assistant at the chair of communication networks of Aachen University of Technology