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Optimized MAC algorithms and performance report

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Abstract

This OMEGA deliverable presents a description of medium access control (MAC) layer mechanisms and cross-layer resource allocation algorithms for the band-extended PLC in multi user context.

Keyword list

Power Line Communication, OFDM, LP-OFDM, FMT, MAC, cross layer design, multi user communications

Executive Summary

This deliverable presents a description of the MAC layer mechanisms and the cross-layer resource allocation algorithms that are proposed within WP3 to develop a wide band (up to 100 MHz) transmission interface that allows the coexistence and compatibility with the existing HomePlug AV system. It reviews the MAC mechanisms in PLC context. The cross layer resource allocation (bits, carriers, codes, powers, etc.) is analyzed in multi user context with filtered multitone and linear precoded OFDM systems. These techniques are optimized for application to in-home PLC systems. The channel model presented in Deliverable 3.2 is used to obtain performance results.

This deliverable is organized as follows. After the introduction in Section 1, Section 2 gives an overview of different MAC mechanisms in PLC context. The HPAV, the UPA DHS and the IEEE P1901 MAC mechanisms are briefly described.

The MAC and the new inter-MAC interfaces are described in Section 3. A description of available information that will be transmitted from the PLC MAC to the inter-MAC and from the inter-MAC to the PLC MAC are given.

The contributions on cross-layer resource allocation in multi user context are given in Section 4. The downlink multi user communication scenario is studied with filter bank and LP-OFDMA systems. The multicast is also developed for OFDM and LP-OFDM systems.

The conclusion follows in Section 5.

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List of Acronyms

Acronym	Meaning
AWGN	Additive White Gaussian Noise
BPLC	Broadband Power Line Communication
CEPCA	Consumers Electronics Powerline Communication Alliance
CIR	Channel Impulse Response
CP	Cyclic Prefix
CSI	Channel State Information
CSMA	Carrier Sense Multiple Access
CTF	Channel Transfer Function
DWMT	Discrete Wavelet MultiTone
FB	Filter Bank
FMT	Filtered MultiTone
HPAV	HomePlug AV
ICI	Inter Carrier Interference
IEEE	Institute of Electrical and Electronic Engineers
ISI	Inter Symbol Interference
ITU	International Telecommunications Union
LDPC	Low Density Parity Check
LP-OFDM	Linear Precoded OFDM
MAC	Medium Access Control
MC	Multi Carrier
OFDM	Orthogonal Frequency Division Multiplexing
OMEGA	Home Gigabit Access
O-QAM	Offset-QAM
PAM	Pulse Amplitude Modulation
PHY	Physical
PLC	Power Line Communication
PSK	Phase Shift Keying
QAM	Quadrature Amplitude Modulation
QoS	Quality of Service
RS	Reed-Solomon
SINR	Signal over Interference plus Noise Ratio
SNR	Signal to Noise Ratio
UPA	Universal Powerline Alliance

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1 Introduction

The objective envisioned by the OMEGA project is to design wireless and wired communication technologies capable of reaching a physical (PHY) rate of 1 Gbit/s in in-home networks. Within the project, WP3 focuses on power line communications. This deliverable aims at describing the MAC layer mechanisms and the cross-layer resource allocation algorithms that have been developed within WP3. It presents the adaptation of the HPAV MAC layer to the OMEGA PHY enhancement. Here, the multi user resource allocation schemes are analyzed in downlink and multicast contexts.

This deliverable is organized as follows. After the introduction in Section 1, Section 2 gives an overview of different MAC mechanisms in PLC context. The MAC and the new inter-MAC interfaces are described in Section 3. The contributions on cross-layer resource allocation in multi user context are given in Section 4. The conclusion follows in Section 5.

2 Overview of MAC mechanisms for PLC

The basic task of a MAC layer is to control access of multiple subscribers connected to a communications network using a same, so-called “shared transmission medium”, and organization of information flow from different users applying various telecommunications services. Generally, functions of a MAC layer applied to any telecommunications network can be divided into the following three groups:

- Multiple access.
- Resource sharing
- Traffic control functions.

A multiple access scheme establishes a method of dividing the transmission resources into accessible sections [6], which can be used by network stations for transmission of various information types. The choice for a multiple access scheme depends on the applied transmission system within the physical layer and its features. Following the definition of a multiple access scheme, there is a need for specification of the strategy for the resource-sharing MAC protocol. The task of a MAC protocol is the access organization of multiple subscribers using the same shared network resources, which is ensured by management of the accessible sections provided by the multiple access schemes. Duplex mode is one of the functions of the MAC layer controlling the traffic between downlink and uplink transmission directions. Additional traffic control functions, such as traffic scheduling, admission control, and so on, can be implemented in higher network layers, but also completely or partly within the MAC layer. In any case, to fulfill QoS requirements of various telecommunications services, MAC layer and its protocols have to be able to support realization of different procedures for traffic scheduling, as well as to support implementation of a Connection Admission Control (CAC) mechanism [8].

2.1 PLC MAC requirements

A minimum of 20~Mbps average data rate as data payload on application layer is required and the systems shall provide the desired bit rate for 98\% of all connections [12], [13]. For connection in multi level houses additional repeaters are acceptable. For high data rate a minimum of 70~Mbps is required for the scalability of the data rate.

The specific MAC layer requirements are common with de IEEE 802 features. The PLC MAC takes into account the links quality and the traffic loads to determine the best routes within inhome network [14]. The broadcast and the repetition are supported. The MAC coordinator is used without user intervention and provides collision free mechanism.

2.2 Multiple access schemes

In order to efficiently share the available resources between different users in a communication system, several multiple access techniques can be used. The main techniques commonly implemented are frequency-division multiple access (FDMA), time-division multiple access (TDMA) and code-division multiple access (CDMA).

FDMA consists in dividing the spectrum into individual channels that will be allocated to the different users. FDMA technique can be easily implemented since users can be separated at the receiver using a simple filter. However, one disadvantage is the maximum number of users having to share a given band. In fact, increasing the number of users leads to reducing the bandwidth of the individual bands allocated to each user, which should be kept large enough in order to avoid strong signal attenuations.

TDMA allows several users to share the same frequency channel by dividing the signal into different short time slots. The users transmit in rapid succession, one after the other, each using its own time slot. TDMA is generally more difficult to implement than FDMA since it requires perfect synchronization between all transmitters and receivers. Systems adopting TDMA include the 2nd generation cellular systems (GSM) and the digital enhanced cordless telecommunications (DECT) systems.

With the CDMA technique, several users are able to transmit data simultaneously over the same frequency band. The users' signals are distinguished by different PN codes that have to be known at the receiver. The combination of direct-sequence principle and CDMA technique is referred to as DS-CDMA. A judicious selection of PN codes with good cross and autocorrelation is necessary for DS-CDMA systems. In the case of synchronous communications, optimal performance can be obtained using orthogonal codes, such as orthogonal variable spreading factor (OVSF) codes, Walsh-Hadamard codes [16], and the complementary series of Golay [17]. For asynchronous communications, non-orthogonal codes offering good cross and autocorrelation properties, such as Gold [18], Kasami [19], and Zadoff-Chu codes [20], can be used.

2.3 MAC protocols

2.3.1 Fixed access strategies

Fixed access assigns each user a predetermined or fixed channel capacity irrespective of whether the user needs to transmit data at that time. Fixed access strategies are suitable for continuous traffic, but not for bursty traffic which is typical for data transfer provided in the PLC access networks [7].

2.3.2 Dynamic access strategies

Dynamic access schemes are adequate for data transmission and in some cases it is possible to ensure a satisfactory transmission quality for delay-critical traffic [4], [5].

2.3.2.1 Contention based protocols [8]

2.3.2.1.1. ALOHA

Aloha, also called the *pure Aloha method*, refers to a simple communications scheme in which each source (transmitter) in a network sends data whenever there is a frame to send. If the frame successfully reaches the destination (receiver), the next frame is sent. If the frame fails to be received at the destination, it is sent again after a randomly calculated waiting time. Thus, the packets generated by two different network stations A and B collide if they are transmitted at the same time (PG – packet generation, Figure 1).

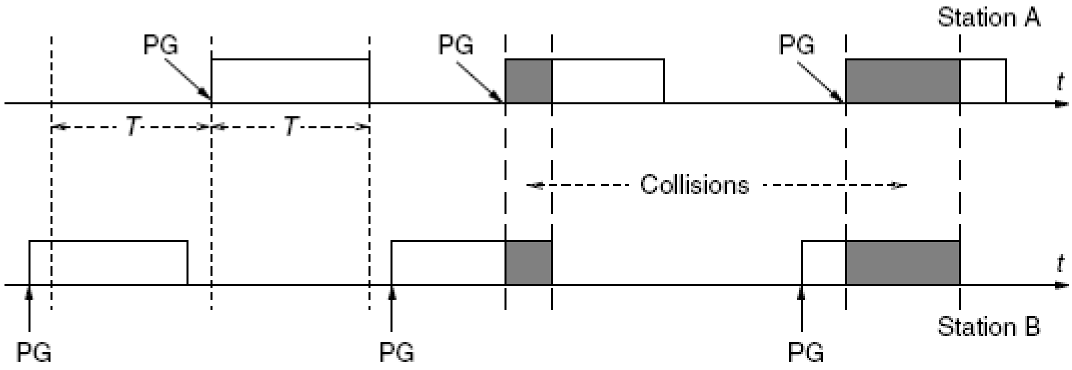


Figure 1: Timing diagram for pure ALOHA protocol

According to network throughput and date rate, the random nature of the ALOHA protocol causes a very low network utilization (maximum 18%). Additionally, ALOHA protocols are characterized by an instable behavior with a resulting performance collapse (network utilization is almost zero) if the network is highly loaded, which makes the realization of QoS guarantees difficult. For these reasons, it can be concluded that the pure ALOHA protocol is not suitable for application in PLC networks.

Performance of pure ALOHA protocol can be improved by application of so-called "slotted ALOHA protocol", where the transmission channel is divided into time slots, whose size equals the duration of a packet transmission *T*. The network stations can start transmission of a packet only at the beginning of a time slot (PT – packet transmission, Figure 2). Thus, after generation of a packet (PG) station A has to wait for beginning of next time slot to transmit the packet. Therefore, there is no collision between second packets of stations A and B, as was the case in pure ALOHA protocol (Figure 1). In slotted ALOHA protocol, a collision occurs only if two or more network stations transmit a packet in the same time slot (Figure 2), as is the case with third packets of stations A and B.

The slotted ALOHA achieves much better network utilization (36%) than the pure ALOHA protocol. However, the same instable performance behavior still remains and two basic requirements on the MAC protocol are not fulfilled by the slotted ALOHA as well (good network utilization, QoS guarantees for various services).

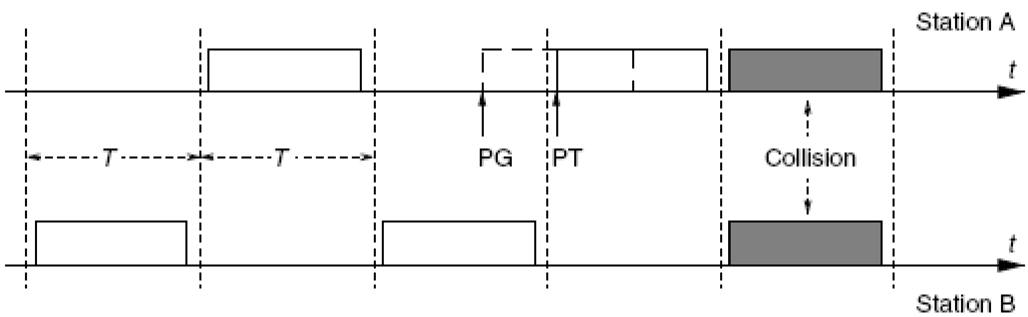


Figure 2: Timing diagram for slotted ALOHA protocol

2.3.2.1.2. CSMA

Carrier Sense Multiple Access (CSMA) is a probabilistic MAC protocol in which a node verifies the absence of other traffic before transmitting on a shared transmission medium, such as an electrical bus, or a band of the electromagnetic spectrum.

"Carrier Sense" describes the fact that a transmitter listens for a carrier wave before trying to send. That is, it tries to detect the presence of an encoded signal from another station before attempting to transmit. If a carrier is sensed, the station waits for the transmission in progress to finish before initiating its own transmission.

"Multiple Access" describes the fact that multiple stations send and receive on the medium. Transmissions by one node are generally received by all other stations using the medium.

2.3.2.1.2.1. Collision detection (CSMA/CD)

It senses the channel for a collision after transmitting. When it senses a collision, it waits a random amount of time before retransmitting again. But on power lines the wide variation of the received signal and noise levels make collision detection difficult and unreliable.

2.3.2.1.2.2.Collision avoidance (CSMA/CA)

As in the CSMA/CD method, each device listens to the signal level to determine when the channel is idle. Unlike CSMA/CD, it then waits for a random amount of time before trying to send a packet. Packet size is kept small due to the PLC's hostile channel characteristics. Though this means more overhead, overall data rate is improved since it means less retransmission. CSMA/CA is used in HomePlug Standards and we will talk about it in more detail in the next sections.

Dynamic protocols with contention can not ensure any guarantees of QoS for time-critical services and also full network utilization can not be reached.

2.3.2.2 Collision-free protocols

Collision-free dynamic protocols can be realized using Token Passing, Polling or reservation methods.

2.3.2.2.1. Token passing

In a network applying a token-passing protocol, the network stations exchange so-called "token-messages" (tokens) in a particular order to specify access right to the medium for every station in the network. A station that just received a token has right to access the medium and transmit its data. After the transmission is completed, the token is sent to another station in the network, which can carry out its transmission. In this way, each network station has an extra time period, determined by the token message, to send its data and collisions between multiple network stations are not possible, which leads to a collision-free network operation. To avoid a situation where a network station transmits its data for a longer time period and with it obstructs other stations to transmit their data, a limit for an individual transmission can be defined. This can be done by limitation of the transmission time, or by specification of a maximum amount of data to be transmitted within one token turn.

The most well-known token-passing protocol is Token-Ring, developed for the application in LANs with a ring topology.

2.3.2.2.2. polling methods

As opposed to the token-passing principle, polling is a centralized access method providing a main station to control the multiple access to the shared medium [10]. The base station (e.g. base station of a PLC access network) sends a so-called “polling message” to each network station in accordance with the round-robin procedure or any other cyclic order. If a station receives a polling message, it can transmit the data for a predefined time period. In the case that a polled station does not have data to transmit, it sends a kind of acknowledgment to the base station to inform it that there is no data to send. Afterward, the base station polls the next station in the cycle. The network station transmits also an acknowledgment after the end of a packet transmission, also informing the base station that the transmission is completed (e.g. before a limit is reached) and that the next station in the cycle can be polled. The polling access procedure can be applied to any network topology. Independent of the physical network structure (e.g. three, bus, ring, or a star network that is typical for wireless communications systems), the polling cycle is carried out in accordance with a logical order of the network stations.

Token passing and polling make possible realization of some QoS guarantees in the network. However, with an increasing number of network stations the time between two sending rights for a stations (round-trip time of tokens or polling messages) becomes longer, making both protocols not suitable for time-critical services [4] , [5].

2.3.2.2.3. Reservation methods

In the case of reservation MAC protocols, a kind of reservation of the transmission capacity is done for a particular user or a service. For this purpose, a part of the transmission resource is reserved for realization of the reservation procedure, so-called “signaling”. Thus, in a general case, a number of accessible sections of the transmission resources, provided by a multiple access scheme, is allocated for signaling that includes transmission of the user requests (demands) to a central network unit (e.g. PLC base station) and acknowledgments/transmission rights from the base stations. After the reservation procedure is finished, the base station has already allocated necessary network resources for the requested transmission, ensuring a contention-free data transmission. For realization of reservation MAC protocol in a TDMA system, the time frames are divided into two intervals Figure 3; one provided for the reservation procedure (R) and another for the collision-free data transmission (T). A reservation completed within the request phase of a time frame (e.g. time frame T – 1) can affect the same or the next time frame, where the transmission is carried out within time frame T – 2, or the transmission can be carried out in any of the next time frames *(*T – *i).* Thus, the base station has an opportunity to schedule multiple requests received from different users for various services in accordance with the required QoS, priorities, and so on. The lengths of the reservation and transmission periods within a time frame can be fixed or they can be dynamically changed, depending on the current load situation in a network.

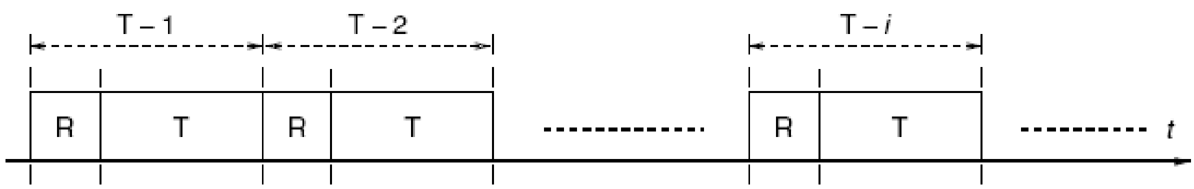


Figure 3: Principle of reservation MAC protocols

2.4 Traffic control

Network traffic control is the process of managing, prioritising, controlling or reducing the network traffic, particularly Internet bandwidth, used by network administrators, to reduce congestion, latency and packet loss. The traffic control mechanisms can be divided in the following three groups:

• Duplex mode, as a part of the MAC layers, that optimization can improve network utilization,

• Traffic scheduling, representing additional mechanisms to be implemented in the MAC layer to improve QoS in the network, and

• Connection Admission Control mechanism, operating above the MAC layer to secure QoS level in the network.

2.4.1 Duplex mode

The duplex mode defines the organization of traffic in downlink and uplink transmission directions, that is, transmission of data from a base station to network stations and in the opposite direction from the network stations to the base station, respectively. For this purpose, the accessible sections of the transmission resources, provided by a multiple access scheme (Sec. 5.2), are divided into two groups; one of them serving the transmission in the downlink direction and the other for the uplink transmission. The division of the network resources between the downlink and the uplink can be made in two ways:

• FDD – Frequency Division Duplex, and

• TDD – Time Division Duplex.

In the first case, a frequency range is used for the uplink transmission and another range for the downlink. Thus, if we consider an FDMA system, a number of the frequency bands (transmission channels) are allocated for the downlink, and the remaining bands are allocated for the uplink, building an FDMA/FDD transmission system.

On the other hand, TDD provides different time frames where the transmission is carried out by turns in the downlink, or in the uplink. So, in a TDMA system, there are two types of the time frames, downlink and uplink frames, usually containing a number of the time slots, building a TDMA/TDD transmission system.

Besides two combinations of the multiple access schemes and the duplex modes presented above, an FDMA system can be realized with TDD, as well as a TDMA system can be combined with FDD. In the first case, the transmission channels provided by the FDMA scheme are used for both transmissions in the uplink and downlink directions. However, there is a division in the time domain, ensuring extra time periods that are used for the uplink and for the downlink transmission. On the other hand, in case of a TDMA/FDD system, the frequency spectrum is divided in the uplink and in the downlink parts. Thus, the corresponding uplink and downlink time slots are accessed in these two frequency ranges.

2.4.2 Traffic scheduling

Mechanisms for the traffic scheduling in communications systems are responsible for management of different data flows transmitted through a network with respect to the fulfilment of the required QoS guarantees for particular telecommunications services. There are a large number of mechanisms for the traffic scheduling investigated for implementation in various telecommunications technologies (ATM, modern IP networks, etc.). This traffic scheduling can take into account the priority realization; QoS control mechanisms and fairness provision in the network. For more details, see [8].

2.4.3 CAC mechanism

Since every telecommunication system provides a finite transmission capacity (a maximum available data rate), a network can carry only a limited number of connections simultaneously. Additionally, if the services with higher data rate and QoS requirements are transferred, the transmission limits can be quickly achieved, particularly in networks with limited data rates, such as recent PLC access networks. Therefore, communications networks apply very often call/connection admission control mechanisms (CAC), which limits the number of connections to be admitted in the network in accordance with current QoS level and data rates that can be ensured for individual connections, applying various telecommunications services. The limitation of the number of admitted connections in a network is specified by so-called “admission policy”. Additionally, in networks operating under unfavorable noise conditions, such as PLC, the influence of disturbances on the change of the available data rate in the network has to be particularly considered in an applied CAC mechanism as well [8].

2.5 HomePlug MAC mechanisms

2.5.1 HomePlug 1.0 MAC [7]

The HomePlug 1.0 Medium Access (MAC) protocol is a modified CSMA/CA (Carrier Sense Multiple Access / Collision Avoidance) protocol with priority signalling. HomePlug 1.0 devices operate in an ad hoc mode in the sense that devices communicate with each other freely, without any centralized coordination. The frame structure and protocol of HomePlug 1.0 is depicted in Figure 1.

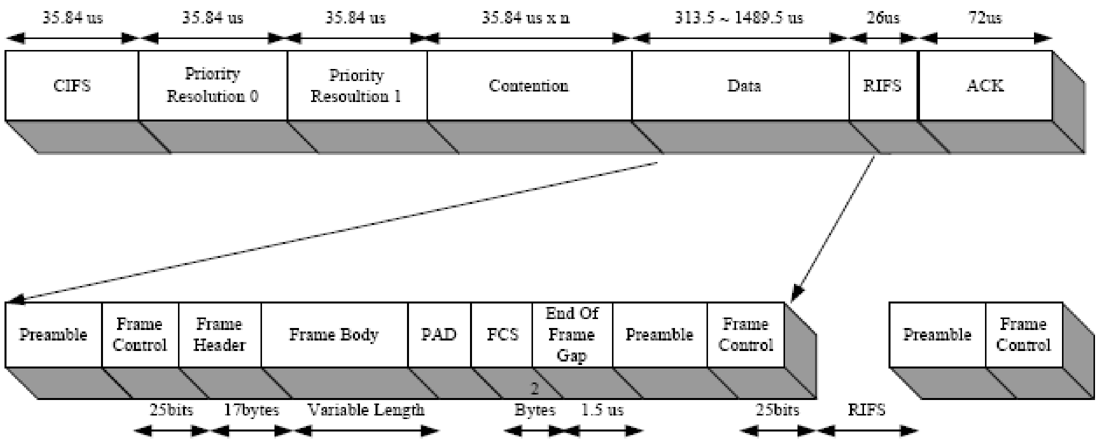


Figure 4: HomePlug 1.0 frame structure and protocol

The RIFS shown in the figure is “Response Inter-Frame Spacing.” A frame control bit is used to indicate the desire of a station to continue to send data, allowing pre-emption only by higher priority traffic. The spacing between the last frame and the incoming frame is CIFS (Contention Window Inter-Frame Spacing).

HomePlug 1.0 provides four priority classes - CA3, CA2, CA1 and CA0 from highest to lowest. Priority resolution is done by asserting signal of the priority level in the PR0 and PR1 slots. For example, to send a CA2 packet, the PLC device should assert a 1 in PR0, causing any node with CA1 traffic to defer, and not assert 1 in PR1 as it would do otherwise. Nodes with CA3 data assert a 1 in both priority slots and CA0 in neither. This effectively resolves contention between different priority classes. Contention within the same priority class is resolved during the contention period. The contention period is a contention period. The contention period is a form of CSMA/CA with a priority dependent back off window size schedule. For the lower two priority classes, it is 8-16-32-64 slots, while it is 8-16-16-32 slots for the two higher priority classes. On collision, the range of contention slots over which a transmission is started is increased according to the schedule. Aside from starting with a smaller range (8 slots compared to 32 slots), when a HomePlug 1.0 node defers (detects another node's transmission in an earlier slot), it uses this information to back off, but less aggressively than in the case of a collision. This technique serves to reduce costly collisions further.

2.5.2 HomePlug AV MAC [21]

HPAV provides connection-oriented Contention Free (CF) service to support the QoS requirements (guaranteed bandwidth, latency and jitter requirements) of demanding AV and IP applications. This Contention Free service is based on periodic Time Division Multiple Access (TDMA) allocations of adequate duration to support the QoS requirements of a connection.

HPAV also provides a connectionless, prioritized Contention based service to support both best-effort applications and applications that rely on prioritized QoS. This service is based on Collision Sense Multiple Access/Collision Avoidance (CSMA/CA) technology which is applied to only traffic at the highest pending priority level after the pending traffic with lower priority levels has been eliminated during a brief Priority Resolution phase at the beginning of the contention window.

To efficiently provide both kinds of communication service, HPAV implements a flexible, centrally-managed architecture. The central manager is called a Central Coordinator (CCo). The CCo establishes a Beacon Period and a schedule which accommodates both the Contention Free allocations and the time allotted for Contention-based traffic. As shown in Figure 5, the Beacon Period is divided into 3 regions:

- Beacon Region
- CSMA Region
- Contention-Free Region

The CCo broadcasts a beacon at the beginning of each Beacon Period; it uses the beacon to communicate the scheduling within the beacon period. The beacons are extremely robust and reliable. The schedules advertised in the Beacon are persistent i.e., the CCo promises not to change the schedule for a number of Beacon Periods—and the persistence is also advertised in the beacon so that the transmitting station for a connection can confidently transmit during its persistent allocation(s) even if it has missed several beacons within the advertised persistence of the schedule. This provides additional continuity even if a few beacons are missed. The CSMA periods are also persistent so that stations wishing to send CSMA traffic can do so even if they miss a few beacons.

The MAC layer provides both Contention (CSMA) and Contention Free (CF) services through the respective regions in the Beacon Period. The CCo-managed Persistent Contention Free (PCF) Region enables HPAV to provide a strict guarantee on Higher Layer Entity (HLE) QoS requirements. An HLE uses the Connection Specification (CSPEC) to specify its QoS requirements. The Connection Manager (CM) in the station evaluates the CSPEC and, if appropriate, communicates the pertinent requirements to the CCo and asks the CCo for a suitable Contention Free allocation. QoS features specified in the CSPEC include:

- Guaranteed bandwidth
- Quasi-Error free service
- Fixed Latency
- Jitter control

If the CCo is able to accommodate the connection request, it will ask the stations to “sound” the channel. This allows the stations to perform the initial channel estimation (i.e., establish a Tone Map specifying the optimal modulation on each OFDM tone). The Tone Map is communicated from the receiver to the transmitter; the channel estimation is also communicated in abbreviated form to the CCo to help it determine how much time should be allocated to the connection. Based on the CSPEC and the channel sounding results, the CCo provides one or more persistent time allocations (Transmit Opportunities (TXOPs)) for the connection within the PCF Region.

The PCF Region also contains time for non-persistent allocations good only in the current beacon period. These non-persistent allocations are used to provide additional short term bandwidth to connections that require it (e.g., because of transient errors or changing channel conditions) to meet their QoS requirements, providing that the transmitting station hears the beacon at the beginning of the Beacon Period. When this time is not used for non-persistent CF allocations, in may be used for CSMA traffic. Again, stations must hear the beacon in order to know whether the time is available for CSMA traffic.

Messaging in HPAV is direct from station to station; however, the CCo monitors the messages. The header of each message contains information about how much data is pending for transmission on the connection; if this amount becomes large on a given connection, the CCo may allocate additional non-persistent time to the connection in the PCF Region.

The Persistent CSMA Region provides prioritized contention-based communication. It is used where there is no CSPEC and/or the traffic is of short duration. When operating in 1.0 Coexistence mode, or “Hybrid Mode”, AV coordinates with HomePlug 1.0 devices and permits them to communicate during the CSMA period.

As shown in Figure 5, the Beacon Period is synchronized to the AC line cycle. By synchronizing to the line cycle, HPAV provides stability of the periodic allocations relative to the line cycle. This, in turn, provides better channel adaptation to the synchronous (to the line cycle) interference, resulting in improved throughput. The beacon provides announcements of where the beacon will occur over the next few beacon periods (i.e., beacon persistence) to enable continued communications by stations that miss an occasional beacon.

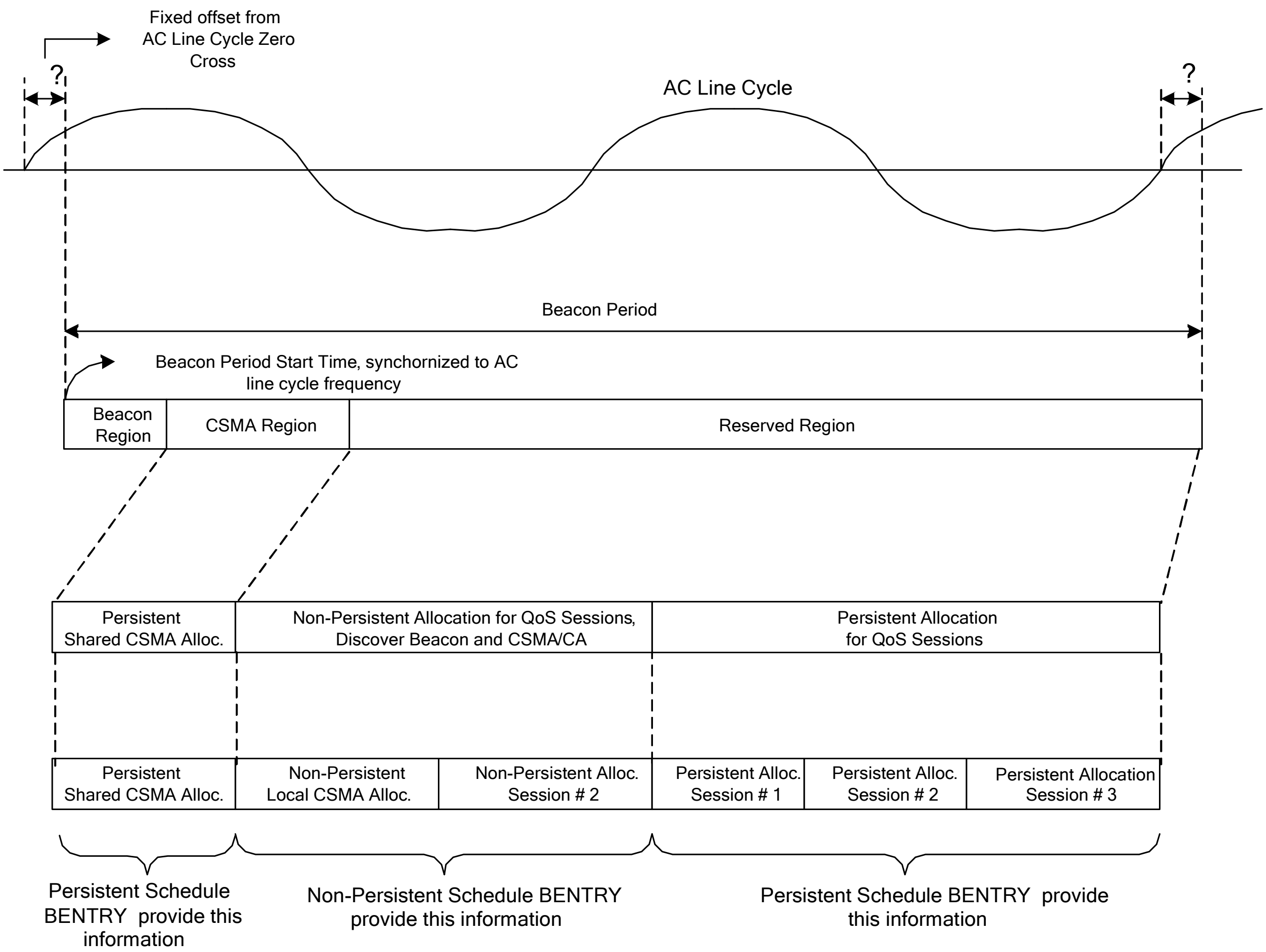


Figure 5: Example of Beacon Period Structure

2.5.2.1 MAC Control Plane

The Medium Access Control (MAC) Layer contains an integrated Connection Manager (CM). HLEs provide a Connection Specification (CSPEC) that details QoS requirements for application data. For bridged traffic, CSPECs may be generated dynamically by the Auto Connection Service (ACS) or by a higher layer QoS Manager that coordinates QoS over multiple network segments; otherwise the traffic is transmitted as prioritized CSMA traffic.

The Control Plane provides a seamless interface to the application layer. Application requirements are received at the H1 Control SAP in the CSPEC and are interpreted by the CM. The CM is responsible for evaluating the CSPEC and setting up the appropriate connection in conjunction with the CM in the station at the other end of the connection and with the CCo. It is the Connection Manager’s responsibility to ensure that the appropriate AV mechanisms are engaged in order to provide the application with the bandwidth it requires. It must also monitor the level of service that the connection is receiving and take remedial action if the guaranteed QoS is not being provided.

The MAC also maintains a clock that is tightly synchronized to the CCo’s clock (the CCo includes a timestamp in the beacon). This means that the entire HPAV network shares a common network clock for use by HLEs that have tight timing constraints (e.g., to synchronize surround sound speakers).

2.5.2.1 *MAC Data Plane*

In the Data Plane, the MAC accepts MSDUs (e.g., Ethernet packets) arriving from the Convergence Layer and encapsulates them with a header, optional Arrival Time Stamp (ATS) and Check Sum to create a MAC Frame. The MAC Frames are then enqueued into the appropriate MAC Frame Stream. It is the MAC's responsibility to ensure that the MSDUs related to a given connection are delivered to the PHY in a timely fashion for transmission during the time allocated for the connection. For this purpose, it maintains individual queues for each connection's data, for each priority level of CSMA traffic and for each priority level of Control Messages.

Each MAC frame stream is divided into 512 octet segments each of which is encrypted and encapsulated into a serialized PHY Block (PB). As shown in Figure 6, the PBs are packed into an MPDU which is delivered to the PHY. The PHY transmitter applies forward error correction and places the resulting PPDU onto the powerline as described in the PHY section above.

As the receiver reconstructs the MSDUs, it selectively acknowledges the PBs; those that are not positively acknowledged are retransmitted during the next TXOP. The Selective Acknowledge (SACK) is an integral part of the TDMA allocation. When all the PBs composing an MSDU have been received correctly, the segments are decrypted and the resulting MSDU is passed to the Convergence Layer for delivery to the appropriate HLE.

Control messages are processed in an analogous fashion.

Since FEC and Selective Acknowledgment (SACK) are performed on relatively small blocks of data, the FEC is more robust and retransmissions are minimized. These two features contribute to HPAV's ability to operate at near channel capacity.

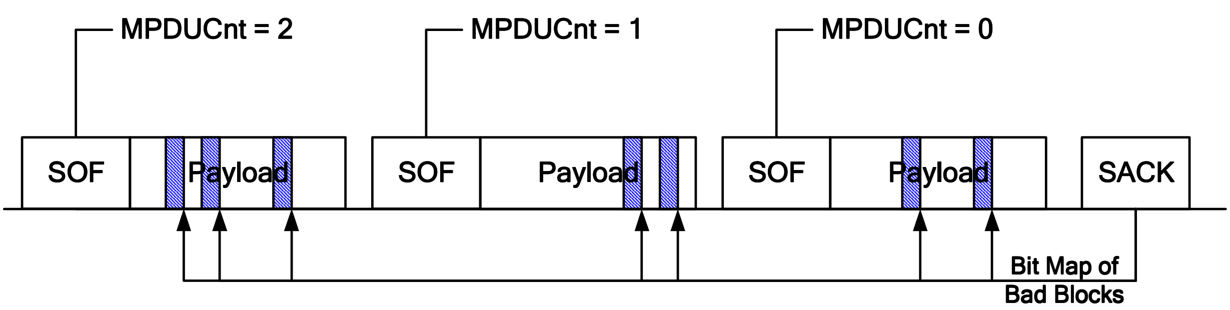


Figure 6: MAC Segmentation and MPDU Generation

2.6 UPA DHS MAC mechanisms

This section resumes the functioning of the medium access method used by UPA DHS technology. For detailed information see [23].

UPA DHS technology uses an advanced dynamic time division MAC (ADTDM) mechanism. The ADTDM is a contention free channel access method. It allows all the nodes belonging to the network to occupy the channel according to different QoS requirements.

The nodes of the network are classified in three groups:

Access Point: the Access Point (AP) is the master of the network. It controls the channel access and it manages the QoS required by each connection. More precisely, based on the QoS required by each node request, on the number of the network nodes, etc, the AP computes the time to allocate at each connection. The information regarding the channel assignment are distributed by the AP, using control packets called channel "token", to the rest of devices in the network. Each "token" specifies the node that can occupy the network and the duration for which it can be hold. When a node ends the time assigned for his data exchange, it gives back the "token" to the AP.

Repeater: the Repeater is a device that solves the problem of the hidden nodes. The repeater relays packets addressed to a node of the network that is hidden.

End-Points: an End-Point is a device that is not an AP neither a Repeater. In an UPA DHS network there is always only one AP. It's worth noting that two UPA DHS network can communicate only if they have the same network identifier. Devices with different network identifier will only coexist.

Figure 7 shows an example of an ADTDM functioning. As we can see, the node *A* is the manager and the nodes *B* and *C* are two end-points.

The communication can be summarized by the following steps:

Node *A* starts the communication by sending a control message announcing that the following packets are sent from *A*. In this message are also sent information regarding the PHY layer (i.e. gain to be used at the receiver and bit-loading information). After this phase, *A* sends a burst of data addressed to *B* and another one addresses to *C*. Finally, *A* sends a control message that releases the channel to *B*. This message contains also the information regarding the time allocated to *B* (channel "token") in which it can occupy the channel.

After receiving the channel "token", *B* sends a message saying that the following packets will be sent from itself. Then, *B* sends data to *A* and *C*. When *B* has finished the transmission, it gives back the "token" to *A* (network manager).

Now, *A* is the channel owner and then it transmits the message indicating that the following packets will be sent from *A*. Then, it sends packets to *C* and finally it gives the channel "token" to *C*.

When *C* becomes the channel owner, it sends the message indicating that it is the channel owner. Nevertheless, *C* doesn't have data to transmit, so it gives back the "token" to *A*.

Now node *A* is again the channel owner and the cycle can start over again.

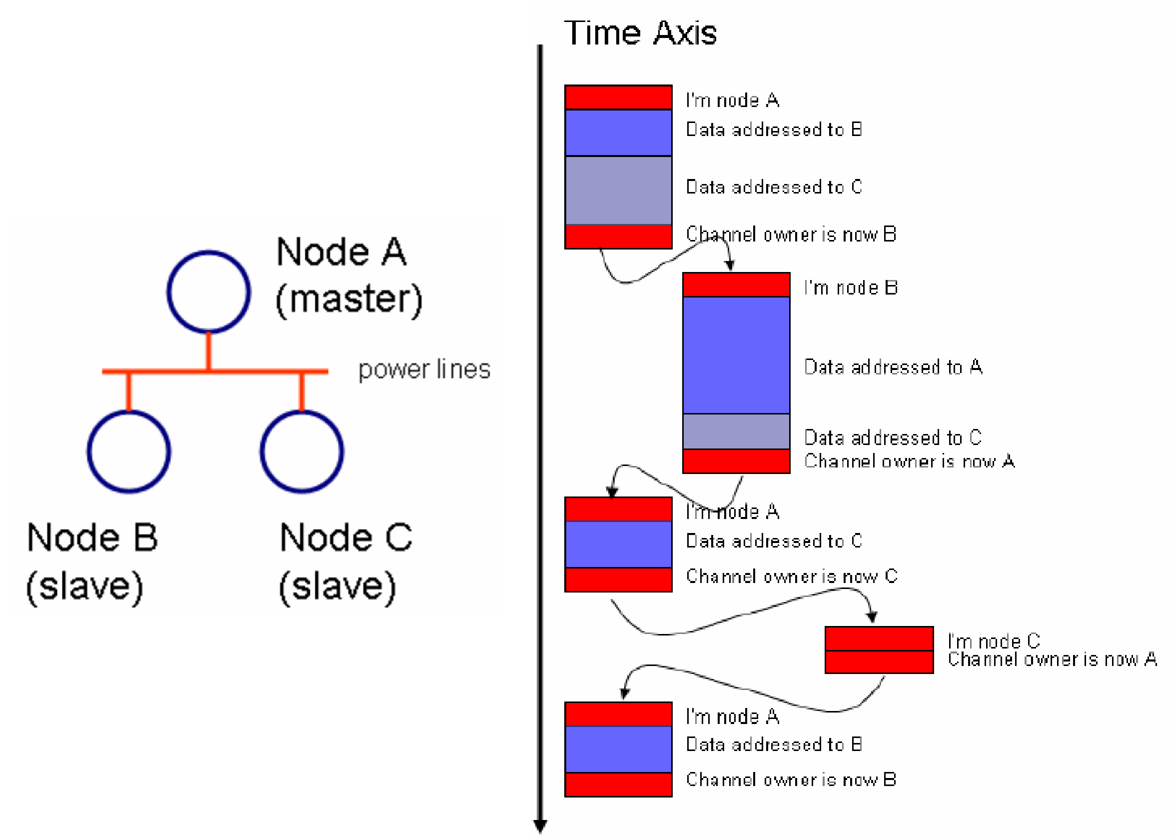


Figure 7: Sample network (left side) and its ADTDM MAC functioning (right side) (figure taken from [23])

UPA DHS uses three main types of MAC frames:

Data frames: data frames contain essentially payload.

Channel estimation frames: channel estimation frames are sent periodically by each node of the network. Therefore, each node can estimate the channel between itself and any other node of the network.

Access frame: The Access frames are sent by the AP and Repeaters to invite new nodes to enjoy the network. When a new node receives an Access frame, it can contend the channel by using a back-off algorithm. After having won the contention, the new node starts a connection with the AP or the Repeater. In this connection the QoS parameters and the PHY layer specifics are negotiated.

We want to underline that the network topology is learned by the devices by using an optimized version of the spanning tree protocol (IEEE 802.1d) [24].

2.7 IEEE P1901 MAC mechanisms

The IEEE P1901 MAC is based on a TDMA/TDD MAC with hybrid resource sharing mechanisms built on top of an OFDM PHY layer. Data are encapsulated inside OFDM symbols [14], [15]. The control and data symbols transmitted consecutively by a single node constitute a transmission frame.

The channel access is done through the use of a special MAC element called a token. Master nodes decide the type of frame that their slave nodes are going to transmit next, based on the type of token included in the current frame, or the kind of channel access the slave nodes can perform. Tokens have several intended uses, depending on the type of token, and the actions to be undertaken upon its reception also depend on the type of token.

The master node manages the type of channel access, contention free access to the channel with a predetermined decision about which of its slaves will be allowed to transmit data, in what order, and for how long, or a contention based access of the slaves to the channel. These decisions will be published by the use of different types of tokens. In both schemes the final result is a node holding a token. The holder of the token has the right to access the medium for a specific time and after that time it has to assign the token to the next destination node. By the control of the frequency of the token occupancy and the length of the time the node is allowed to hold the token, it is possible to guarantee the requested QoS.

3 MAC and inter-MAC interfaces

The Inter-MAC layer [22] is a new layer, technology-independent, that would use the information received from the underlying technologies to select the most appropriate one fitting to services requirements. The Inter-MAC is located in each OMEGA device between the network protocol layer and the medium access control layer. The Inter-MAC layer establishes and manages the OMEGA network, assuring QoS to the end-user applications acting as an intelligent bridge between PLC, Radio and HWO.

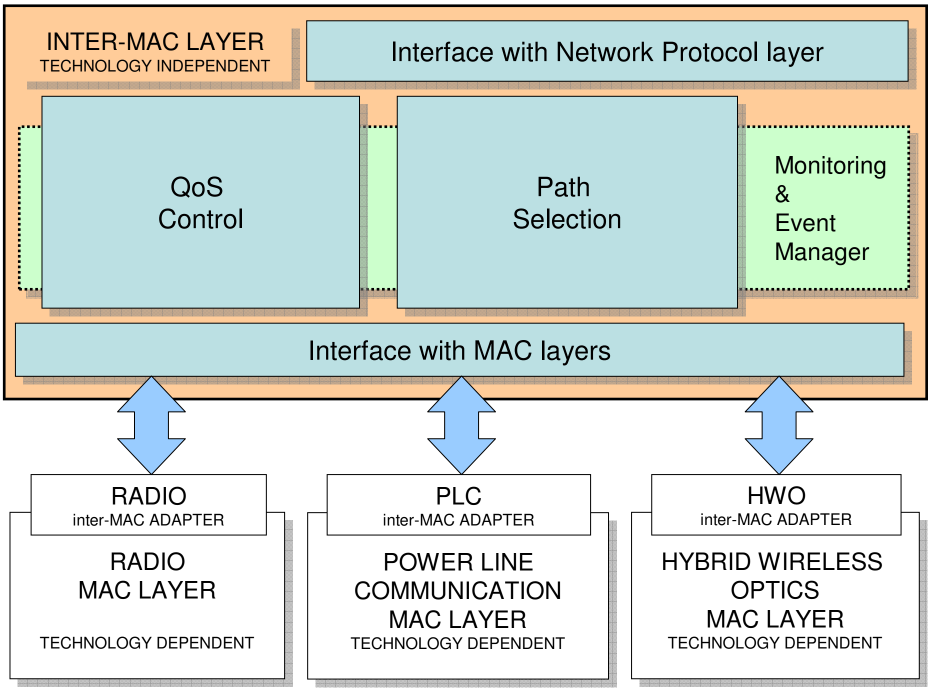


Figure 8: Inter-MAC overview

The goal of this section is to describe the interface between PLC MAC and inter-MAC. This description will be divided into three parts:

- Reporting from PLC-MAC to I-MAC: what kind of information will be sent from PLC MAC to I-MAC
- From I-MAC to PLC-MAC: I-MAC resource reservation is required to PLC Mac layer

- Exchange format: what type of frames will be used with what type of physical interfaces

3.1 From PLC-MAC to I-MAC

In this section, a description of available information that will be transmitted from the PLC MAC to the Inter-MAC will be made, with the goal to give to the Inter-MAC a report about the PLC link. Main information used will be:

- Bit Loading Estimate (BLE): this quantity represents the number of data bits that can be carried over the channel per microsecond. It takes into account FEC and Guard interval redundancy but does not take into account delimiter or protocol overhead. It can also take different values over a beacon period (40ms @50Hz). BLE represent an interesting metric because
    - It aggregates most of channel characteristics to compute capacity.
    - Abstract physical layer characteristics => can be used for any technology
- Detection of legacy devices
- Beacon detection reliability
- Number of MSDU / octets / segments / PHY Blocks transmitted or received
- drops / retries / failures
- Number of PHY frames / Number of bursts
- Latency statistics

3.2 From I-MAC to PLC-MAC

For resources reservation, I-MAC must ask to the PLC MAC what the needs are. The idea is to use the Connection SPECification (CSPEC) (c.f. §2.5.2) for the I-MAC to ask for its specific requirements. Main points available are:

Many parameters can be specified to ask for new link or to negotiate QoS:

- Delay/Jitter bound
- Average / Maximum MSDU size
- Min / Max / Average data rate (not including MAC and PHY overhead)
- Smallest tolerable average data rate
- Min / Max time between 2 Tx opportunities
- MSDU error rate
- Exception policy / max inactivity time

Also include Connection Description:

- IP source/destination addresses and Protocol

Upon reception of a CSPEC, the bandwidth manager is responsible for resource allocation:

- If requested CSPEC cannot be met, an alternate CSPEC can be proposed.
- Link quality is monitored and application should be warned if CSPEC is violated. In that case a new CSPEC may be proposed by the CCO to reconfigure the connection.

The connection specifications provide a format for resource reservation that supports many scenarios for link creation, negotiation, maintenance.

3.3 Exchange format and physical interface

For the data plane, the retained proposal is to use Gigabit Ethernet as physical interface between PLC and I-MAC platform.

For the control plane, the idea is to use Management Messages (MM) over Ethernet to exchange information from I-MAC to PLC and from PLC to I-MAC. Main advantages of using such a format are:

- MM are Ethernet frame using a specific Ethertype
- Format proposition is available, can be detailed when needed
- No need for a second hardware link
- simplify prototyping and development,
- MM can be bridged easily

The full description of frame format will be detailed later

4 Cross layer resource allocation schemes in multi user context

4.1 Multi user scenarios

Multiple user resource allocations (bits, carriers, spreading codes, etc.) in a multi-user environment must be done according to the expected scenarios:

- point to multipoint (multicast HDTV or SDTV streams);
- Multipoint to point (VoIP or video games to reduce latency).

These scenarios can be achieved in a centralized network manner. Advantages of centralized organization are a relative simple realization of QoS guarantees in the network. Whenever, a failure of main station causes a break down of the whole network. The centralized systems may produce more transmission overhead (transmission of signaling information), too [1].

4.2 Multi user Filter bank

In this section we address the problem of resource allocation in the multiuser context when deploying a multi carrier (MC) filter bank (FB) modulation scheme.

The use of MC FB modulation schemes allow realizing the so called frequency division multiplexing access (FDMA) that is characterized by the partitioning of the sub-channels among the network users.

In multiuser context the resource allocation optimization problem becomes the problem of allocating bit, power, and sub-channels to the network users. Furthermore, the optimization of the PHY layer parameters has to be accomplished taking into account the employed MAC.

In sub-section 4.2.1 we show a procedure that allows optimizing the resource allocation and the PHY layer parameters in the multiuser context when deploying orthogonal frequency division multiplexing (OFDM) and filtered multitone (FMT) modulations jointly with FDMA. Then, in sub-section 4.2.2 we introduce the problem of the optimal time slots design when time division multiple access (TDMA) is used over periodically time variant PLC channels.

4.2.1 Multicarrier FDMA

We assume a network where a central coordinator (CCo) allocates resources by collecting information regarding the network state, i.e., number of users, channel conditions of each user, quality of service required from each user request, etc. Once the CCo has collected all the information needed, it dynamically allocates the resources among the users. We focus on the downlink channel from the CCo to the N_U users of the network. Multiplexing is accomplished by partitioning the sub-channels across the users realizing orthogonal frequency division multiplexing access (OFDMA) when OFDM is used and FMT-FDMA when FMT is used.

When deploying FMT or OFDM, the achievable rate on the $k-th$ sub-channel of user u can be computed as [26]

$$C^{(u,k)}(\mu)=\frac{1}{(M+\mu)T}\log_2\left(1+\frac{SINR^{(u,k)}(\mu)}{\Gamma}\right) \quad [bit/s], \tag{1}$$

where
$$SINR^{(u,k)}(\mu)=\frac{P_U^{(u,k)}(\mu)}{P_W^{(u,k)}+P_I^{(u,k)}(\mu)}, \tag{2}$$

and $P_U^{(u,k)}(\mu)$, $P_W^{(u,k)}(\mu)$ and $P_I^{(u,k)}(\mu)$ are respectively the useful, the noise and the interference power terms experienced by the u-th user in the k-th sub-channel. In (1), the term μ represents the overhead (OH) factor for OFDM and FMT non-critically sampled [25]. When deploying OFDM μ corresponds to the cyclic prefix (CP) duration (in samples). Furthermore, T represents the sampling time and M is the number of sub-channels.

In order to allocate resources to the users, the central manager can solve the following joint optimization problem [26]:

$$\begin{aligned}
&\max_{\mu,\alpha} && \sum_{u=1}^{N_U}\sum_{k\in K_{ON}}\alpha^{(u,k)}C^{(u,k)}(\mu) && \\
&s.t. && \sum_{u=1}^{N_{users}}\alpha^{(u,k)}=1 && k\in K_{ON} \\
& && \sum_{k\in K_{ON}}\alpha^{(u,k)}C^{(u,k)}(\mu)\geq\frac{p^{(u)}}{100}\sum_{k\in K_{ON}}C^{(u,k)}(\mu) && u=1,\dots,N_U
\end{aligned} \tag{3}$$

where $\alpha^{(u,k)}$ denotes the binary sub-channel coefficient that is equal to 1 if sub-channel k is allocated to user u, and zero otherwise, while $p^{(u)}$ denotes the percentage of the bit-rate that the $u-th$ user has to achieve with respect to the one that it would achieve in a single user scenario. Problem (3) can be solved using integer programming [27] once μ is fixed. To simplify the complexity we instead use linear programming. That is, for each value of μ, the coefficients that give the sub-channels allocation, are returned via linear programming followed by rounding the $\alpha^{(u,k)}$ coefficients to nearest zero or one integer. The exhaustive search of μ and $\alpha^{(u,k)}$ yields the solution.

In practical implementations the number of bits is rounded to an integer number to take into account that a finite set of M-QAM constellations are used. Therefore, to take into account that only a finite set of constellations can be employed, the CCo has to solve a problem equal to (3) except for the fact that now the capacity (1) has to be replaced with the rate achievable after having loaded the bits associated to the nearest available constellation similarly to the bit-loading algorithm for single user OFDM presented in [29].

Finally, we want to emphasize that the procedure above exposed can be applied to both FMT and OFDM.

In the following we show numerical results for the OFDMA case only.

Numerical Results

To report a numerical example we consider a network of 3 users with a weighted fair resource allocation, i.e., $p^{(u)} = 33$ for all users. Each user experiences a given channel. The used channels are the three channel realizations of Figures 4-6 presented in [28]. More precisely, the channel of class 2 denotes the transmission from the CCo to User 1, the channel of class 5 denotes the transmission from the CCo to User 2, and the channel of class 9 denotes the transmission from the CCo to User 3.

The signal is transmitted respecting the power spectral density mask (PSD) reported in Figure 9. The noise is additive white and Gaussian with a PSD of -110 dBm/Hz.

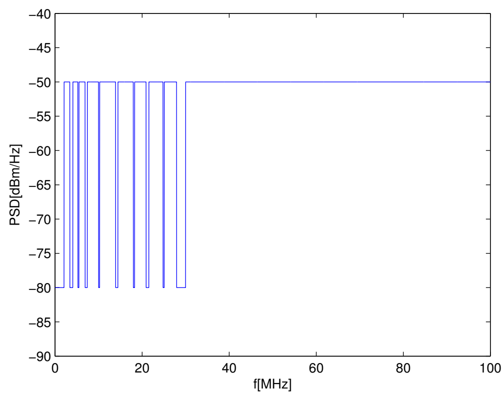


Figure 9: PSD mask for the transmitted signal.

Due to the presence of notches in the PSD of the transmitted signal, as explained in Section 3 of [28], it is convenient to use pulse shaped OFDM (PS-OFDM) instead of simple OFDM. In fact, PS-OFDM allows a better spectral confinement w.r.t. OFDM, and thus it uses a higher number of active tones still respecting the mask.

In the following, we use PS-OFDM with a raised cosine window as MC modulation scheme. The overhead μ equals $\mu = GI + RO$, where GI and RO respectively denote the guard interval and the roll-off duration. Furthermore, we set the RO duration equal to the GI duration. The number of sub-channels is M = 4096 into the frequency band 0-100 MHz. For a detailed treatment of PS-OFDM, please refer to [28]. The corresponding multiple access is accomplished via pulse shaped OFDMA (PS-OFDMA).

Regarding the bit-loading algorithm, we use the one proposed in [29]. The constellations employed are 2-PAM, 4, 8, 16, 64, 256, 1024-QAM.

Figure 10 shows the aggregate bit-rate, and the rate achievable by each user as a function of μ. As we can see, an optimal OH (or equally *GI*) for the multiuser PS-OFDMA scenario can be found. This value corresponds to the one that maximizes the aggregate rate. For the simulated network the optimal OH equals 360 samples when the aggregate rate is equal to 724 Mbit/s.

Usually the OH of OFDM is designed such that it results longer than the "worst case" channel impulse response. As it is well known, by doing this, the received signal is not affected by interference (refer to [28] Section 3). Nevertheless, we notice that the use of an OH equal to 5 μs, that is a representative value for a "worst case" indoor PLC channel duration, yields an aggregate rate of 678 Mbit/s. Thus, the gain given by the use of the proposed optimization procedure w.r.t. the conventional choice of an OH equal to the "worst case" channel length equals 6.7%.

Figure 11 shows the optimal sub-channels allocation for an overhead that is equal to 3.6 μs (this is the value that maximizes the aggregate network rate).

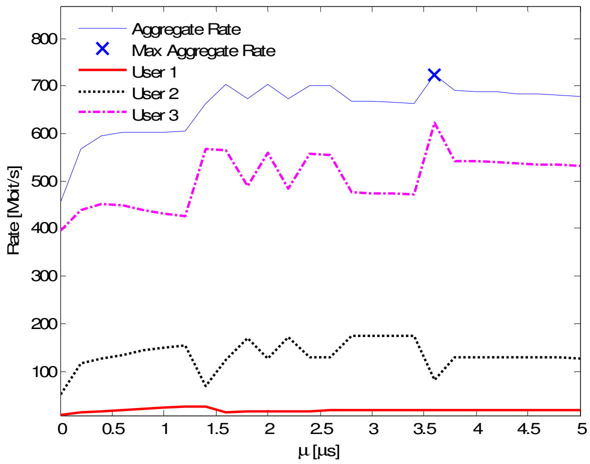


Figure 10: Aggregate and Single User Rates function of the overhead duration for the PS-OFDMA system.

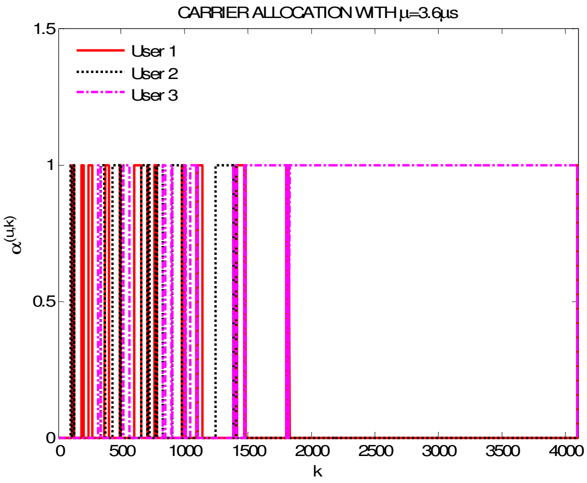


Figure 11: Sub-Channels Allocation for an overhead μ that equals the optimal one (3.6µs).

4.2.2 TDMA: time slot design over PL time variant channels

State-of-the-art PLC devices make use of a hybrid MAC mechanism to satisfy the high QoS required by multimedia services, e.g., video streaming, network gaming, etc.. As an example, in Section 2 we have seen that HPAV uses a persistent time division multiple access (TDMA) scheme to satisfy the QoS based traffic. On the contrary, the best effort traffic is served in a non persistent scheduled region or is left for a contention based region.

In order to maximize the network performances, a cross layer design between the physical (PHY) and the MAC layer has to be considered. In [30] we have proposed a cross layer design method to optimize the TDMA region of an HPAV like system. For a PLC network whose channels are characterized by both cyclostationary noise and cyclic variations of the channel response, the optimal time slot duration in a multi-user scenario has been computed for various amounts of overhead required by the PHY layer. The derived results can be used in practical systems to perform a fast real-time time slot duration selection and scheduling as a function of the number of active users.

4.3 Multi user LP-OFDMA

4.3.1 LP-OFDM system description

The LP technique consists in connecting a subset of subcarriers with precoding sequences to mutually exploit their capacities . In the following, we call this subset of subcarriers block and the subcarriers in one block are not necessary adjacent. The number of blocks is the ratio of the total number of subcarriers N to the precoding sequence length L. We assume the same precoding sequence length L for all blocks. If judiciously done, each resulting block holds an equivalent signal-to-noise ratio (SNR) such that the total supported throughput is greater than the sum of the individual throughputs supported by each subcarrier taken separately. The following expressions for LP-OFDM modulation technique are derived from [39], [41]. The optimum achieved bit rate of the LP-OFDM system, under assumption of perfect synchronization and channel estimation, writes

$$\mathrm{R}_{u,b} = L\log_2\left(1+\frac{1}{\Gamma}\frac{L}{\sum_{n\in S_b}\frac{1}{|h_{u,n}|^2}}\frac{E}{N_0}\right), \tag{4}$$

where $|h_{u,n}|$ is the channel amplitude of user u on subcarrier n, E is the PSD constraint, Γ is the SNR gap, N_0 is the background noise level, S_b is the subset of subcarriers within the b th block of size L. The conventional OFDM system is obtained for $L=1$.

4.3.2 LP-OFDMA

In this section, we present the gain brought by linear precoding technique in OFDMA scenarios. Under satisfaction of the different QoS requirements (rate, delay, error probability, etc.), bits, energies and subcarriers have to be allocated to different users in order to optimize the expected goal. Table I gives the different rate maximization strategies in multi user context [44], [45], [46]; R_k is the data rate a user k gets.

	Objective	Advantage	Disadvantage
Max sum capacity [Jang et al. 2003]	$\max\sum_{k=1}^{K} R_k$	Best sum capacity	No data rate proportionality among users
Max minimum user's capacity [Rhee et al. 2000]	$\max\min_k R_k$	Equal user data rates	Inflexible user data rates distribution
Max weighted sum capacity [Wong et al. 2004]	$\max\sum_{k=1}^{K} w_k R_k$	Data rate fairness adjustable by varying weights	No guarantee for meeting proportional user data rates

Table 1: Bit rate maximization strategies in OFDMA [47]

According to the strategy chosen, different resource allocation schemes have been proposed in the literature. In these different schemes, rather than choosing one subcarrier according to the priority strategy, each user chooses a block of subcarrier for the LP-OFDM allocation.

The maximum weighted sum capacity problem under peak bit error constraint (BER) is considered for the downlink in PLC systems. As in power constraint, a peak constraint is defined in opposition to average constraint [48]. The peak power constraint is the PSD constraint where the power is limited for each subcarrier. In the case of BER, this constraint is applied to each bit. In [44], each user receives a weight according to his required bit rate. This weight does not take into account the user channel conditions and this may lead to no guarantee for meeting proportional user data rate. In [49], each user receives a number of subcarriers that is the ratio of its required bit rate to the number of bits computed with its average channel gain. The users required bit rate will be decreased when the power constraint is too low to meet users required bit rate.

We propose a new channel condition aware proportional fairness (CCAPF) algorithm in multi user context. This algorithm tries first to satisfy users required bit rate when it is possible and then tries to maximize the overall bit rate [47].

In multiuser context, B blocks of subcarriers are distributed among users. A proportional fairness resource allocation scheme is considered where the overall bit rate is maximized under satisfaction of users minimum bit rate requirements

$$\begin{cases} \max\limits_{S_b} \sum\limits_{u=1}^{U} R_u = \max\limits_{S_b} \sum\limits_{u=1}^{U}\sum\limits_{b=1}^{B} s_{u,b} \times R_{S_b}^{u} \\ \text{where } R_{S_b}^{u} = L \times \log_2\left(1 + \dfrac{E}{\Gamma N_0}\dfrac{L}{\sum\limits_{n \in S_b} \dfrac{1}{\left|H_{u,n}\right|^2}}\right) \\ \text{and } s_{u,b} = 1 \text{ if } u \text{ uses the block } b\text{,else } 0 \\ \text{subject to } R_u \geq \tilde{R}_u \end{cases} \tag{5}$$

Due to PSD constraint in PLC systems, all users have the same peak power constraint E on each subcarrier. For simplicity, it is assumed that all users utilize the same precoding sequence length L. The minimum required bit rate $\tilde{R}_u$ is converted into a weight for each user

$$r_u = \frac{\tilde{R}_u}{\sum\limits_{v=1}^{U} \tilde{R}_v}. \tag{6}$$

In [44], a proportional fairness (PF) algorithm is proposed to solve this problem where each user receives a weight $\phi_u = r_u$ according to his bit rate requirement. When extending PF algorithm to blocks, each user will receive $B_u^i = \lfloor \phi_u \times B \rfloor$ blocks of subcarriers. This allocated numbers of blocks do not take into account users channel conditions. Namely, a user u with bad channel conditions may need more blocks of subcarriers than B_u^i or a user v with good channel conditions may need less blocks than B_v^i. As a result, there is no guarantee for meeting proportional user data rate. Our proposed algorithm is applied in LP-OFDM context and users channel conditions are taken into account when allocating blocks of subcarriers. Consequently, the user who reaches his required bit rate will not receive additional blocks and the unallocated blocks will be redistributed among unsatisfied users. Unlike, the max-min user capacity scheme (see [46]), flexibility is introduced when redistributing the unallocated blocks. Therefore, a unsatisfied user who has very bad channel conditions will not receive additional blocks and the more capable user will receive additional blocks. In [49], it is assumed that each user experiences the same channel gain $\overline{h_u}$ over all subcarriers, and $\overline{h_u} = \text{mean}_n \left|H_{u,n}\right|^2$. Therefore, the number of bits per block $\overline{R_S^u}$ can be estimated using for each user. Hence, a unsatisfied user u who has achieved R_u bit rate right now, needs about

$$B_u^2 = \left\lfloor \left(\tilde{R}_u - R_u\right) / R_S^u \right\rfloor \tag{7}$$

blocks to be satisfied. The following algorithm tries to allocate blocks among users in order to satisfy their required bit rate. In the initialization step, each user receives one block of subcarriers. In [44], there is no priority among users but in the proposed algorithm, a priority is introduced among users according to their channel capacities. The more the user channel capacity, the lesser its priority.

Initialize $R_u = 0$, $B_u = 0$, $\Omega = \{1, 2, \cdots, U\}$

compute $\overline{h_u}$ and $\overline{R_S^u}$

sort users according to $\overline{h_u}$

for each sorted user u do

  compute R_S^u where S is its L best available subcarriers

  $R_u = R_u + R_S^u$, $B_u = B_u + 1$,

  if $B_u = B_u^i$ or $\overset{\square}{R}_u \leq R_u$ then

   $\Omega = \Omega - \{u\}$

  end if

end for

while Ω is not empty do

  find $u^* = \arg\min_{u \in \Omega} R_u / \phi_u$

  compute $R_S^{u^*}$ where S is its L best available subcarriers

  $R_{u^*} = R_{u^*} + R_S^{u^*}$, $B_{u^*} = B_{u^*} + 1$,

  if $B_{u^*} = B_{u^*}^i$ or $\overset{\square}{R}_{u^*} \leq R_{u^*}$ then

   $\Omega = \Omega - \{u\}$

  end if

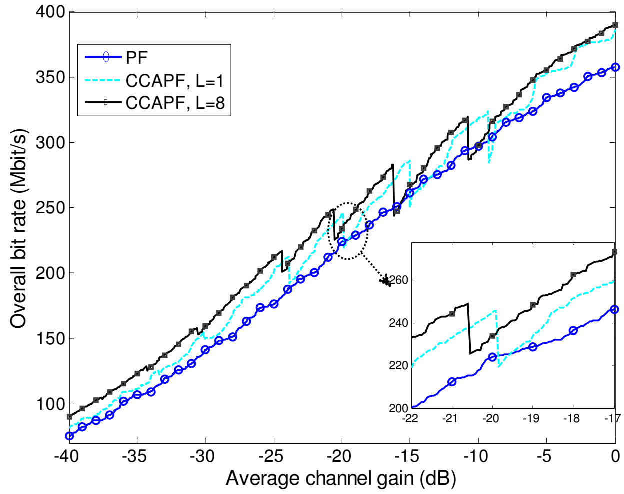
end while

At this step, there are $B^r = B - \sum_{b=1}^{B} B_u$ unallocated blocks. These blocks are redistributed when it is possible to unsatisfied users. For each unsatisfied user v, B_v^2 is computed and if $B_v^2 > B^r$, user v will not receive additional blocks. Else, user v will receive the maximum number of blocks which allows to satisfy its required bit rate.

After this stage, if there are unallocated blocks, the maximum sum capacity (table 1) algorithm is performed. Thus, the best user on each block receives this block.

4.3.2.1 Simulation results

In this section, we present simulation results for the proposed algorithms. The generated signal is composed of $N = 1160$ subcarriers transmitted in the band 1--30 MHz. Perfect synchronization and channel estimation are assumed. A high background noise level of -110 dBm/Hz is assumed and the signal is transmitted with respect to a flat PSD of -50 dBm/Hz. The maximum number of bits per symbol is limited to 15. The multipath channel models of the various in-home measured channels for PLC given in [43] are used. In [43], PLC channels are classified into 9 classes according to their capacities, and a model of transfer function is associated to each class (table 3). The higher the channel class number, the better its channel conditions.



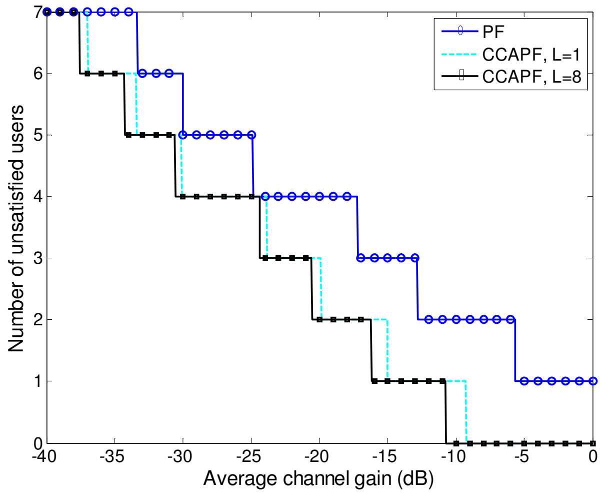


Figure 12: (a) Achieved overall bit rate for 9 users and (b) Number of unsatisfied users for different average channel gain

To maximize the overall bit rate under satisfaction of different user bit rate requirements, the proposed linear precoding based channel condition aware proportional fairness algorithm is performed for 9 users. In addition, it is considered that all users have the same peak BER constraint of 10^{-5} and each user experiences one different channel class and the required minimum bit rate is 20 Mbit/s for all users. Figure 12a and Figure 12b gives respectively the comparisons of the achieved bit rate and the number of unsatisfied users. The CCAPF methods give more overall bit rate than PF algorithm and reduces the number of unsatisfied users. This result shows the performance of blocks redistribution. The jump positions in their curves show the effect of redistribution of blocks, where bit rate is reallocated to a user who can satisfy his required bit rate. One jump position is zoomed out in Figure 12a, and there are gaps between the CCAPF algorithms. These gaps explain what is shown in Figure 12a where CCAPF ($L = 8$) algorithm gives lesser unsatisfied users than CCAPF ($L = 1$) algorithm. The PF algorithm gives more unsatisfied users because a user become satisfied when his channel conditions allow him and there is no effort from best users to ``help'' worst users.

4.4 Multi user multicast scenario

The purpose of OMEGA-PLC is to provide high-quality, multi-stream, entertainment oriented networking over existing AC wiring within the home. It will employ advanced PHY and MAC technologies that provide a 1 Gbit/s class powerline network for video, audio and data. OMEGA-PLC aims to be the network of choice for the distribution of data and multi-stream entertainment including multiple simultaneous HDTV and/or SDTV streams, audio, and streaming AV over IP content throughout the home. Multicasting is interesting in this context. Currently, existing PLC devices use several unicast point-point links to transmit the same data to many users. Multicasting is a network addressing method for the delivery of data to a group of users simultaneously. This technique offers a significant improvement compared to unicasting because it uses less network resources. Over the physical layer, resources have to be allocated in order to satisfy requirements of each multicast user.

Here, we address a PHY-MAC cross-layer resource allocation for multicast OFDM systems in PLC context. Over the PHY layer, resources have to be allocated in order to satisfy requirements of each multicast user. Yet, the difference in link conditions of users makes it difficult to adapt the PHY layer (coding rate, modulation index, etc.) to the link conditions of each user. The conventional resource allocation method in multicast OFDM adapts the PHY layer to the worst user link conditions. Consequently, all users receive the same bit rate and this final multicast bit rate is limited by the worst user channel conditions [31].

To increase the total multicast bit rate and to better fit the channel conditions, we study the heterogeneous multicasting also called multirate multicasting [33]. The conventional multicast system refers to unirate multicast system. In multirate multicast, the receivers of a multicast subgroup are offered service at different rates commensurate with their capabilities (e.g., channel capacities). Therefore, multirate schemes have a great advantage over unirate multicast in adapting to diverse receiver requirements and heterogeneous network conditions [34]. One way of attaining multirate multicast is by hierarchical encoding or layered streaming which is particularly suitable for audio/video traffic. In this approach, the sender provides data in several layers organized in a hierarchy. Receivers subscribe to the layers cumulatively to provide progressive refinement [32], [35]. Over the MAC layer, multicast users are separated into subgroups in frequency domain [31], [32], or in time domain [36]. In frequency domain, each subcarrier is assigned to a subgroup of users which receive the same data symbols on this subcarrier. And the number of loaded bits on each subcarrier is the lowest one of all the users sharing this subcarrier. In time domain, one time slot is allocated for the transmission of one data layer to each multicast subgroup.

Here, we propose a new resource allocation algorithm for multicast OFDM systems in order to increase the bit rate [37], [38]. The proposed algorithm jointly uses the linear precoded OFDM (LP-OFDM) modulation technique and the conventional resource allocation scheme in unirate multicast systems to exploit the channel frequency selectivity experienced by each user. The proposed algorithm is used for unirate multicast systems as well as for multirate multicast systems. First, we maximize the bit rate for unirate multicast systems. Second, we investigate a time domain multirate multicast (TDMM) system where users are gathered into subgroups according to their channel conditions. One time slot is allocated to each subgroup and two modes of grouping multicast users are presented and analyzed. The proposed algorithm is used as resource allocation algorithm within each subgroup. In addition, for comparison purposes, we adapt the frequency domain multirate multicast (FDMM) system, proposed in [31], [32], to PLC context.

4.4.1 Resource Allocation in Unirate Multicast OFDM Systems

4.4.1.1 Multicast system description

In multicast Multicast delivers data to a group of users by a single transmission, which is particularly useful for high-data-rate multimedia service due to its ability to save the network resources. Figure 13 illustrates a simple multicast case in PLC context where source *S* sends multimedia data to three receivers *R1, R2* and *R3*.

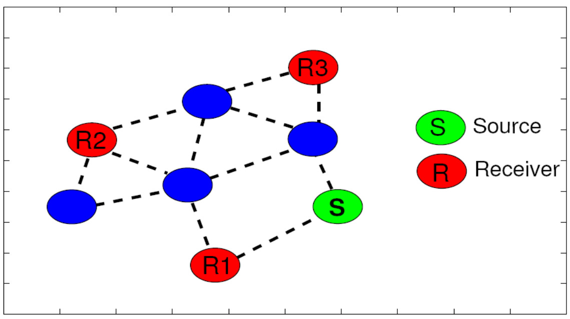


Figure 13: Multicast data delivery scenario

The CCo, in PLC networks, controls the activities of the network, and periodically issues beacon frames containing scheduling information that allocates time to each connection. Since the PLC MAC is connection oriented, all data communications are carried over logical ```connections" identified uniquely by a global connection ID and QoS specification. QoS requirements include the guaranteed bandwidth, quasi-error free service, fixed latency and jitter control [11]. The source S communicates the pertinent requirements to the CCo and asks the CCo for a suitable contention free allocation for communication with stations *R1, R2* and *R3* [11]. If the CCo is able to accommodate the connection request, it will ask the stations to ``sound" the channels. This allows the stations to perform the initial channel estimation. A feedback path from each user to the transmitter reports the channel amplitudes $|h_{u,n}|$ on each subcarrier of each user to transmitter. This channel estimation is assumed to be perfect. Based on this channel estimation and the QoS requirements, the transmitter can perform the multicast resource allocation.

4.4.1.2 Conventional multicast resource allocation

In multicast OFDM systems and in non-hierarchical data context, the modulation should be adjusted to serve all users and especially the user with the worst channel conditions. The conventional method in multicast OFDM, LCG (low channel gain, [31]), consists in allocating resources while satisfying requirements of all users. This method adapts the PHY layer to the worst user link conditions and sets the number of loaded bits per subcarrier with the lowest number of loaded bits over this subcarrier, considering all the channels of users. Hence, the total loaded bits with LCG method in PLC context writes

$$\begin{aligned} \mathrm{R}^{\mathrm{LCG}} &= \sum_{n=1}^{N} \mathrm{R}_n^{\mathrm{LCG}} \\ &= U \sum_{n=1}^{N} \left(\min_u \log_2 (1 + \frac{E}{\Gamma N_0} |h_{u,n}|^2) \right). \end{aligned} \tag{8}$$

We show that the capacity of the conventional multicast system is limited when the number of multicast users increases. Actually, when assuming that users experience independent and identically distributed (i.i.d.) Rayleigh fading channels with parameter σ_u, the expected value of the capacity on subcarrier n writes

$$\mathrm{E}[\mathrm{R}_n^{\mathrm{LCG}}] = -\frac{U}{\log(2)} \exp\left(\frac{U \Gamma N_0}{2E\sigma_m^2} \right) \mathbf{E}_i \left(-\frac{U\Gamma N_0}{2E\sigma_m^2} \right), \tag{9}$$

where

$$\mathbf{E}_i(-h) = -\int_h^{+\infty} \frac{\exp(-t)}{t} dt \tag{10}$$

is the exponential integral and σ_m is the parameter of the minimum amplitude of users. The proof follows that employed in [31]. Therefore, we have

$$\lim_{U \to \infty} \mathrm{E}[\mathrm{R}_n^{\mathrm{LCG}}] = \frac{1}{\log(2)} \frac{2E\sigma_m^2}{\Gamma N_0}. \tag{11}$$

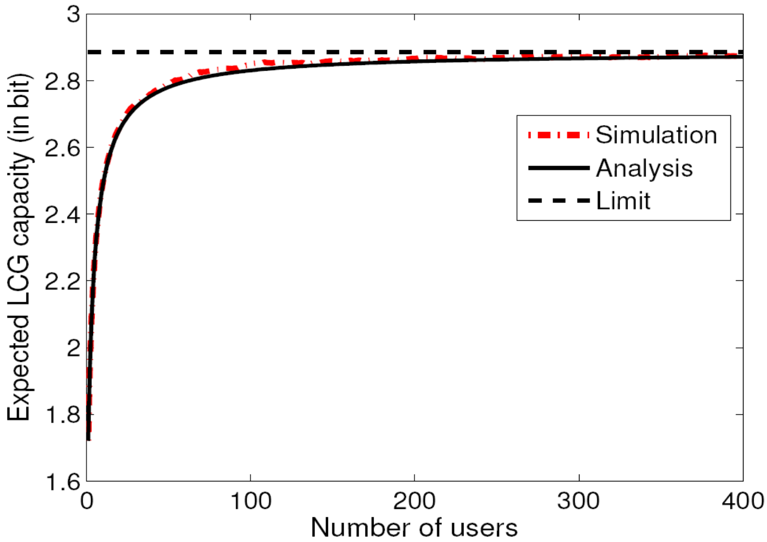


Figure 14: Comparison of analysis and simulation results for the expected LCG method capacity ($E=1$, $N_0=1$, $\Gamma=1$ and $\sigma_m=1$).

Figure 14 shows the comparison of analysis and simulation results for the expected LCG method capacity. Results confirm the fact that the conventional multicast system is limited when the number of multicast users increases.

4.4.2 Bit rate optimization problem in multicast LP-OFDM systems

Here, we jointly use LP-OFDM modulation techniques and LCG method to exploit the channel frequency selectivity experienced by each user. We introduce a $B \times N$ decision matrix $D=(d_{b,n})$ to the optimum achieved bit rate (4). D determines the repartition of the N subcarriers into the B blocks, and D satisfies the following constraints

$$d_{b,n} = \begin{cases} 1 \text{ if } n \in S_b, \\ 0 \text{ else} \end{cases} \quad \text{and } \forall n, \sum_{b=1}^{B} d_{b,n} = 1. \tag{12}$$

Using (4) and (12), the optimum achieved bit rate then writes

$$\mathrm{R}_{u,b} = L \log_2\left(1 + \frac{1}{\Gamma} \frac{L}{\sum_{n=1}^{N} \frac{d_{b,n}}{|h_{u,n}|^2}} \frac{E}{N_0}\right). \tag{13}$$

In multicast systems, when considering the LP-OFDM modulation technique, the loaded bits over the block S_b of subcarriers will be the lowest bit rate of users over this block. This number of loaded bits writes

$$\mathrm{R}_b^{\mathrm{LP}} = \min_u \mathrm{R}_{u,b}. \tag{14}$$

Due to the PSD constraint in PLC systems, all users have the same peak power constraint E on each subcarrier. Hence, there is no power allocation. For simplicity, it is assumed that all users utilize the same precoding sequence length L for all blocks. The optimization problem writes

$$\begin{cases} \max_{D} \sum_{b=1}^{B} \mathrm{R}_b^{\mathrm{LP}} & = \max_{D} \sum_{b=1}^{B} \left(\min_{u} \; \mathrm{R}_{u,b} \right) \\ \text{subject to} & d_{b,n} = \begin{cases} 1 \text{ if } n \in S_b, \\ 0 \text{ else,} \end{cases} \qquad (\mathrm{C1}) \\ & \forall n, \sum_{b=1}^{B} d_{b,n} = 1 \qquad (\mathrm{C2}) \\ & \text{and } \forall b, \sum_{n=1}^{N} d_{b,n} = L. \qquad (\mathrm{C3}) \end{cases} \tag{15}$$

This is a combinatorial resource allocation problem, which is NP-hard [42]. We need to find the optimal D, which maximizes the multicast bit rate. Since this problem of repartition of subcarriers into blocks does not have analytical solution, algorithmic solutions are proposed in the following.

4.4.3 Proposed solutions to the optimization problem

4.4.3.1 Optimal solution: combinatorial solution

The obvious and basic resolution for finding the optimal matrix D is to consider all possibilities of definition of D and then to choose the best case. Looking for all possibilities is a combinatorial problem. For each row of D, we have to set L columns to ``1'' and others to ``0'' taking into account the constraint ($C3$) in (15). As the order of filling the different rows do not change the final result, the number of possibilities writes

$$\frac{\binom{N}{L} \times \binom{N-L}{L} \times \binom{N-2L}{L} \times \cdots \times \binom{2L}{L}}{B!} = \frac{N!}{B!(L!)^B}, \tag{16}$$

where the total number N of subcarriers is a multiple of the number L of subcarriers per block. For a fix precoding sequence length, this solution gives the optimum multicast bit rate, but becomes unfeasible when the number of subcarriers increases. The computation time of this solution can be reduced by using the so-called branch and bound algorithm. This algorithm consists of a systematic enumeration of all candidate solutions, where large subsets of fruitless candidates are discarded by analyzing the properties of the problem. Software tools based on this algorithm can be used to solve this combinatorial problem. Nevertheless, the computation time remains high for larger number of subcarriers.

In this paper, we propose simplified resource allocation algorithm for PLC scenarios. These algorithms jointly use LP-OFDM modulation techniques and LCG algorithm to exploit the channel frequency selectivity experienced by each multicast user.

4.4.3.2 Exploitation of the equivalent channel

Here, we define an equivalent channel, which is the combination of channel conditions of different users. Actually, for each index of subcarrier, the equivalent amplitude of the channel is given by the amplitude of the worst user subcarrier. Let $\left|h_n^{\mathrm{eq}}\right|$ be the equivalent multicast channel amplitude on subcarrier n. Hence,

$$\left|h_n^{\mathrm{eq}}\right|^2 = \min_{u} \left|h_{u,n}\right|^2. \tag{17}$$

Computing this equivalent multicast channel, the multicast resource allocation is the same as the single link resource allocation. Bit-loading algorithms, as proposed in [50] for single user context, can then be applied to this equivalent channel. The LCG method gives results for the conventional OFDM bit loading algorithm

$$\mathrm{R}_n^{\mathrm{LCG}} = U\log_2(1+\frac{E}{\Gamma N_0}\left|h_n^{\mathrm{eq}}\right|^2). \tag{18}$$

To increase the bit rate of this LCG method, we propose to apply the LP-OFDM bit loading in single user context on the equivalent channel and this method will be considered as linear precoding based LCG (LP-LCG) method. Using (13), the corresponding bit rate writes

$$\mathrm{R}_{\mathrm{LP-LCG}} = U\sum_{b=1}^{B} L\log_2(1+\frac{1}{\Gamma}\frac{L}{\sum_{n=1}^{N}\frac{d_{b,n}}{\left|h_n^{\mathrm{eq}}\right|^2}}\frac{E}{N_0}). \tag{19}$$

It has been shown that LP-OFDM outperformed conventional OFDM in single link context [39], [41], and the optimal decision matrix D is such that the distorsion is low for each block. The blocks of subcarriers are then composed of adjacent subcarriers after the sorting operation in descending order. Let O be the vector of sorted indices of $\left|h_n^{\mathrm{eq}}\right|^2$ in descending order. The decision matrix is then

$$d_{b,n} = \begin{cases} 1 \text{ if } n \in \{O_j | (b-1)L+1 \le j \le bL\}, \\ 0 \text{ else.} \end{cases} \tag{20}$$

Here is an example with $N = 8$, $L = 4$, $B = 2$ and $O = [5\ 1\ 8\ 7\ 4\ 3\ 2\ 6]$. We have

$$D = \begin{pmatrix} 1 & 0 & 0 & 0 & 1 & 0 & 1 & 1 \\ 0 & 1 & 1 & 1 & 0 & 1 & 0 & 0 \end{pmatrix}, \tag{21}$$

4.4.3.3 Proposed LBCG (Low Block Channel Gain) method[37], [38]

The LP-LCG method exploits the dimension of precoding for the equivalent channel but does not take advantage of the diversity of users on the blocks. With the LBCG method, we first define the different blocks of subcarriers using the equivalent channel. Then, the number of loaded bits is calculated with the channel of the worst user on each block and not with the equivalent channel.

The proposed LBCG method gives better bit rate than the LP-LCG method. Actually, considering blocks of subcarriers, we show, for all $n \in S_b$, for all b and for all u,

$$\left|h_{u,n}\right|^2 \geq \left|h_n^{\mathrm{eq}}\right|^2 \Leftrightarrow \sum_{n=1}^{N} \frac{d_{b,n}}{\left|h_{u,n}\right|^2} \leq \sum_{n=1}^{N} \frac{d_{b,n}}{\left|h_n^{\mathrm{eq}}\right|^2}$$

$$\Leftrightarrow \mathrm{R}_{u,b} \geq L\log_2\Big(1+\frac{E}{\Gamma N_0}\frac{L}{\sum_{n=1}^{N}\frac{d_{b,n}}{\left|h_n^{\mathrm{eq}}\right|^2}}\Big).$$

Then, $\forall u,b$

$$\min_u \mathrm{R}_{u,b} \geq L\log_2\Big(1+\frac{E}{\Gamma N_0}\frac{L}{\sum_{n=1}^{N}\frac{d_{b,n}}{\left|h_n^{\mathrm{eq}}\right|^2}}\Big) \tag{22}$$

and

$$\sum_{b=1}^{B}\left(\min_u \mathrm{R}_{u,b}\right) \geq L\sum_{b=1}^{B}\log_2\Big(1+\frac{E}{\Gamma N_0}\frac{L}{\sum_{n=1}^{N}\frac{d_{b,n}}{\left|h_n^{\mathrm{eq}}\right|^2}}\Big).$$

From (22), we derived that the bit rate offered by the LBCG method is bounded. We aim at maximizing the left-hand side of inequality (22) according to (15). Maximizing the right-hand side of the inequality increases the left-hand side. The definition of D, in (20), maximizes the right-hand side of (22).

The maximum of the right-hand side corresponds to the LP-LCG method. Then, we can derive that the proposed LBCG method is not optimal, but gives better results than the LP-LCG method. Using (13) and (14), the bit rate offered by the LBCG method writes

$$\mathrm{R}_{\mathrm{LBCG}} = U\sum_{b=1}^{B}\Big(\min_u L\log_2\Big(1+\frac{1}{\Gamma}\frac{L}{\sum_{n=1}^{N}\frac{d_{b,n}}{\left|h_{u,n}\right|^2}}\frac{E}{N_0}\Big)\Big). \tag{23}$$

The following algorithm describes how to compute the multicast bit rate with the LBCG method. Results for the conventional LCG method is obtained for $L=1$.

Data: N , U , B , L and $\forall u,n$ $\left|h_{u,n}\right|^2$ R bit rate per user

Result: $R \leftarrow 0$;

begin

- **for all** subcarrier n , $n \in [1;N]$
  - compute $\left|h_n^{\mathrm{eq}}\right|^2$ from (17);
- sort $\left|h_n^{\mathrm{eq}}\right|^2$ in descending order; let O be the set of sorted indices;
- define D as in (20);
- **for all** block b , $b \in [1;B]$
  - **for all** user u , $u \in [1;U]$
    - compute $\mathrm{R}_{u,b}$ from (13) ;
  - $\mathrm{R}_b^{\mathrm{LP}} \leftarrow \min_u \mathrm{R}_{u,b}$;
  - $R \leftarrow R + \mathrm{R}_b^{\mathrm{LP}}$;

End

Table 2: LBCG method algorithm

4.4.3.4 Comparison of the different methods

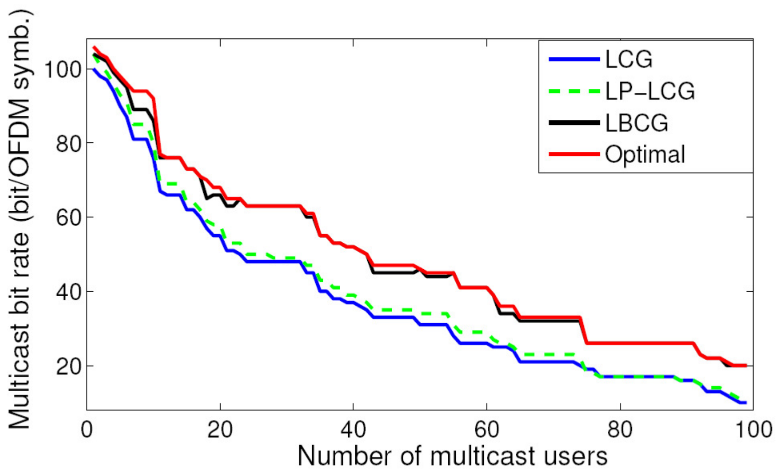


Figure 15: Total loaded bits per user according to the number of multicast users

Here, we address the comparison of the different unirate multicast methods. To compute the optimal solution, i.i.d. Rayleigh fading channels with lower number of subcarrier are used. The total number of subcarriers is $N = 12$ and the precoding sequence length is $L = 4$. Figure 15 shows the multicast bit rate in bit per OFDM symbol. As expected, the optimal solution outperforms the others and the LP component improves the conventional multicast OFDM system (LCG). The LBCG method gives better results than LP-LCG and LCG methods. This LBCG method offers performance close to the optimal one which does an exhaustive search for optimal repartition of subcarriers.

Compared to the conventional LCG method, the additional complexity brought by the LP-LCG method is the utilization of the precoding matrix which is composed of Hadamard orthogonal matrices. In addition to the utilization of th precoding matrix, the LBCG method requires the computation of the different $R_{u,b}$ for each user on each block. Nevertheless, the additional complexity is marginal compared to the initial complexity.

4.4.3.5 Analysis of multicast bit rate between two PLC channel classes

In this part, we analyze by simulation the unirate multicast bit rate (in bit per OFDM symbol) between two users, where each user experiences one different class of channel within the 9 classes [38]. The LBCG method is considered as the resource allocation scheme because it gives the best bit rate in unirate multicast context. Figure 16 shows the comparison of multicast bit rates for all couples of channel classes. These results confirm the fact that the multicast bit rate is limited by the least capable user in unirate multicast context. In addition, these results suggest that multicast users must be separated so that the worst user does not alter the multicast bit rate. The bit saturation explains the lower gaps of multicast bit rates among class 7, 8 and 9. Actually, 10 bits are assigned to each subcarrier, the highest allowed by the HomePlug AV standard.

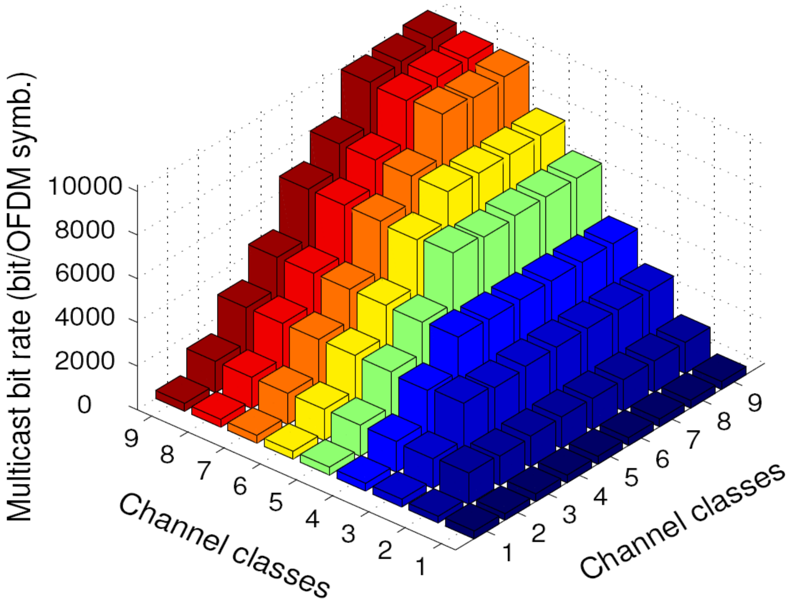


Figure 16: Comparison of multicast bit rate, in bit per OFDM symbol, between two channel classes.

4.4.4 Multirate multicast system

Assuming that the multicast data are encoded into layers and any combination of the layers can be decoded at the receiver, the multicast bit rate can be increased by separating users according to their channel conditions. Under this assumption, the sender provides data in several layers organized in a hierarchy. Receivers subscribe to the layers cumulatively to provide progressive refinement [31][35], [36]. If only the first layer is received by the user with the lowest data rate, the decoder produces the worst quality version. As more layers are received by more capable users, the decoder combines the layers to produce improved quality. Multicast users can be separated into subgroups in frequency domain [31], [32], or in time domain [36].

Figure 17 shows the PHY-MAC cross-layer modules in the transmitter for the multirate multicast data transmission. The channel state information of receivers is used by the multicast scheduler, the multicast subgroup management and the resource controller. The multicast subgroup manager gathers the multicast users into subgroups and subgroups are determined either in frequency domain (on each subcarrier) or in time domain (on each time slot). Moreover, the multicast subgroup manager offers information to the multicast scheduler module such as the bit rate. Multicast scheduler module determines the quantity of data in every frame. The resource controller assigns time slots and subcarriers, and determines the modulation order on each subcarrier. Video source encoder module encodes the streaming video data with the determined data rate and coding rate. A sufficiently large size of buffer to store the real-time data is assumed. In the message frame, the message is made from the combination of encoded video and multicast users management information [34]. Then, after processing in the PHY interface module, the multicast TX bits are transmitted to the multicast users.

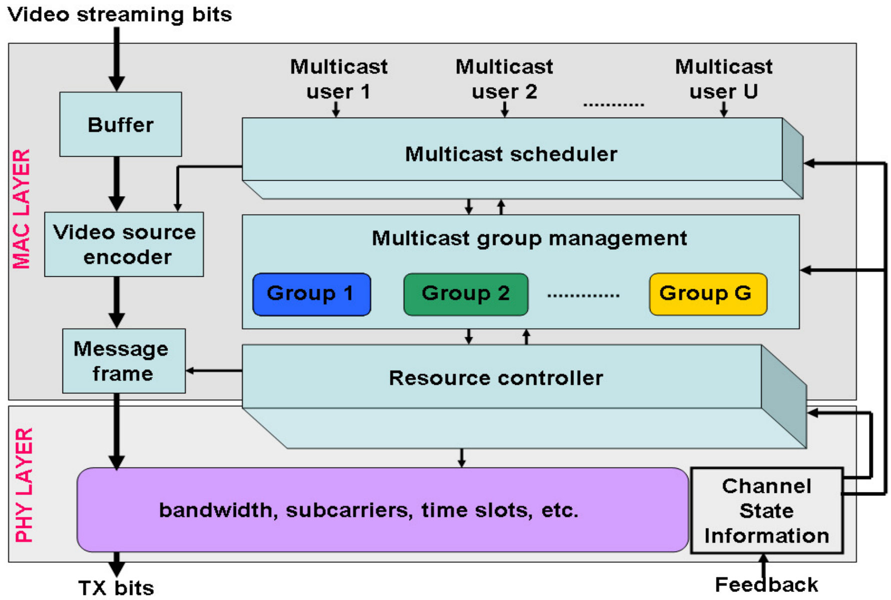


Figure 17: PHY-MAC cross-layer modules in the transmitter side.

4.4.4.1 *Frequency domain multirate multicast (FDMM) systems*

To increase the total multicast bit rate, users are separated in frequency domain [31][32]. Actually, each subcarrier is assigned to a subgroup of users which receive the same data symbols on this subcarrier. And then, the number of loaded bits on each subcarrier is determined considering the lowest one among the channel amplitudes of all the users allocated to this subcarrier. This FDMM method significantly increases the total multicast bit rate compared to the conventional LCG method in wireless communications [31], [32].

For power spectral density constrained systems, as in PLC, the optimization problem in [31], [32], reduces to

$$\max_{b_n} b_n \sum_{u=1}^{U} \mathrm{H}\left(\left|h_{u,n}\right|^2 - f(b_n)\right). \tag{24}$$

Actually, due to PSD constraint, the decision of modulation order is independent for each subcarrier. H is the Heaviside step function, defined by

$$\mathrm{H}(x) = \begin{cases} 0 & \text{if}\, x < 0 \\ 1 & \text{if}\, x \geq 0 \end{cases} \tag{25}$$

and

$$f(b_n) = (2^{b_n} - 1)\frac{\Gamma N_0}{E}. \tag{26}$$

The bit rate offered by the FDMM method then writes

$$R_{\text{FDMM}} = \sum_{n=1}^{N}\left(\max_{b_n} b_n \sum_{u=1}^{U} \mathrm{H}(|h_{u,n}|^2 - f(b_n)) \right). \tag{27}$$

and it is obvious to show that

$$R_{\text{FDMM}} \geq R_{\text{LCG}}. \tag{28}$$

Actually, for each subcarrier n, we have

$$\begin{aligned} \max_{b_n} \; b_n \sum_{u=1}^{U} \mathrm{H}(|h_{u,n}|^2 - f(b_n)) &\geq b_{\min} \underbrace{\sum_{u=1}^{U} \mathrm{H}(|h_{u,n}|^2 - f(b_{\min}))}_{=U} \\ &\geq Ub_{\min} \end{aligned} \tag{29}$$

$$\text{and } Ub_{\min} = U\left(\min_u \log_2(1 + \frac{E}{\Gamma N_0}|h_{u,n}|^2) \right) = \mathrm{R}_n^{\text{LCG}}.$$

Since the considered Heaviside step function H is not continuous at zero, we cannot find a local maximum by using the first derivative test or second derivative test. An algorithmic solution for this optimization problem writes

$$\begin{aligned} &u^* = \arg\max_u \; \log_2(1 + \frac{E}{\Gamma N_0}|h_{u,n}|^2)\sum_{v=1}^{U} \mathrm{H}(|h_{v,n}|^2 - |h_{u,n}|^2) \\ &\text{and } b_n = \log_2(1 + \frac{E}{\Gamma N_0}|h_{u^*,n}|^2). \end{aligned} \tag{30}$$

4.4.4.1.1. Proportional fairness FDMM

The FDMM method does not consider the fairness among users. To enforce the fairness performance while minimizing throughput degradation, it has been proposed subcarrier/bit allocation scheme for proportional fairness (PF) [31]. This method will be considered as FDMM-PF method.

Let $R_u(t)$ be the bit rate of the u th user and b_n be the number of bits that are assigned to the nth subcarrier at time t. For a low computational complexity, a simplified PF algorithm is developed by employing the average data rate, which is given by [51]

$$R_u(t) = (1 - \frac{1}{T_W})R_u(t-1) + \frac{1}{T_W}\sum_{n=1}^{N} b_n \mathrm{H}(|h_{u,n}|^2 - f(b_n)), \tag{31}$$

where T_W indicates the average window size. For power spectral density constrained systems, an algorithmic solution of the optimization problem for the subcarrier n, derived from [31], writes

$$\begin{aligned} &u^* = \arg\max_u \; \log_2(1 + \frac{E}{\Gamma N_0}|h_{u,n}|^2)\sum_{v=1}^{U} \frac{\mathrm{H}(|h_{v,n}|^2 - |h_{u,n}|^2)}{R_v(t-1)} \\ &\text{subject to given } R_v(t-1), \; v \text{ is user index,} \\ &b_n = \log_2(1 + \frac{E}{\Gamma N_0}|h_{u^*,n}|^2). \end{aligned} \tag{32}$$

4.4.4.2 Time domain multirate multicast (TDMM) systems

In this paper, we propose to separate multicast users in time domain in order to reduce the impact of the worst user channel conditions. Hence, we gather multicast users into subgroups according to their channel conditions (classes). In contrast to unirate multicast methods (LCG, LP-LCG and LBCG) where all multicast users share the same time slots, we allocate each time slot to a subgroup of multicast users and each user is part of only one subgroup. Within each multicast subgroup, the LBCG method is used for the resource allocation process. We define the bit rate of the system as the mean bit rate over the time slots used by all subgroups,

$$\mathrm{R}_{\mathrm{TDMM}} = \frac{1}{G}\sum_{g=1}^{G}\left|G_g\right| \mathrm{R}_{\mathrm{LBCG}}^{G_g}, \tag{33}$$

where $|.|$ is the cardinal function, G is the total number of multicast subgroups and G_g is the g th multicast subgroup. We have

$$\begin{aligned} &\bigcup_g G_g = \bigcup_g \{\mathrm{class}\, C_g \text{ to class } C_{g+1}-1\} = \{1,2,\cdots,9\} \\ &\text{and } \sum_{g=1}^{G}\left|G_g\right| = U. \end{aligned} \tag{34}$$

The unirate multicast system is obtained for $G = 1$. The optimization problem in TDMM systems writes

$$\max_{G_g} \mathrm{R}_{\mathrm{TDMM}} = \max_{G_g} \frac{1}{G}\sum_{g=1}^{G}\left|G_g\right| \mathrm{R}_{\mathrm{LBCG}}^{G_g}. \tag{35}$$

This optimization aims at finding the optimal number of multicast subgroups and the optimal repartition of users in each subgroup. As there is no direct relationship between the multicast subgroups and the achievable multicast bit rate, this problem does not have analytical solutions. Based on the analysis of Figure 16, we propose empirical repartition of users taking into account the gaps among the multicast bit rates between two channel classes. Two modes of grouping users are considered.

- Mode 2: 2 subgroups of multicast users use 2 time slots. Subgroup 1 (G21) is composed of class 1, 2, 3 and 4 channels; and subgroup 2 (G22) is composed of class 5, 6, 7, 8 and 9 channels.
- Mode 3: 3 subgroups of multicast users use 3 time slots. Subgroup 1 (G31) is composed of class 1, 2, 3 and 4 channels; subgroup 2 (G32) is composed of class 5 and 6 channels; and subgroup 3 (G33) is composed of class 7, 8 and 9 channels.

Table 3 gives the repartition of the different subgroups of multicast users over 6 time slots according to the used modes.

Time slots	1	2	3	4	5	6
Unirate	G1	G1	G1	G1	G1	G1
TDMM Mode 2	G21	G22	G21	G22	G21	G22
TDMM Mode 3	G31	G32	G33	G31	G32	G33

Table 3: Utilization of time slots by subgroups of multicast users according to the unirate case and the TDMM mode 2 and 3.

4.4.5 Simulation results

Simulation results for the proposed LP methods applied to multicast systems are provided. The performances of the different algorithms are compared with the conventional multicast approach LCG. The generated signal is composed of N=1024 subcarriers transmitted in the [2; 87.5] MHz band. Perfect synchronization and channel estimation are assumed. Through this work, we choose to employ a PSD mask up to 30 MHz similar to the one employed in HPAV [11]. Furthermore, in order to be compliant with electro magnetic compatibility issues (see deliverable D3.3 [3]), we fix the power spectral density (PSD) of the transmitted signal in the frequency band 30-87.5 MHz equal to -80 dBm/Hz. We consider a White Gaussian Noise with PSD of -140 dBm/Hz. Furthermore, in order to compute the achievable rate of the system, we set a fixed target SER of 10^{-3} without channel coding and the maximum number of bits per symbol is limited to 14.

In the first step, we consider a multicast system with 9 users and each user experiences one different class of channel within the 9 classes. Figure 18 shows the cumulative distribution function of the average and minimum bit rate of multicast users as functions of achievable bit rate. Results confirm the fact that the proposed LBCG method outperforms the conventional multicast approach (LCG method) considering both the minimum and average bit rates. This LBCG method gives better result than the multirate multicast methods where users are separated in time domain. However, the frequency domain multirate multicast method gives the best average bit rate compared to other methods. But, in this considered 9-user multicast system where channels of users are very different, the FDMM method yields a good average bit rate without assigning any subcarrier to a given user. As a result, the QoS requirements of users are not ensured and the fairness among users is degraded. By adapting the bit rates of multicast users at each time slot according to previous allocated bit rates, the FDMM-PF method reduces the bit rate offered by the FDMM method, but the minimum user bit rate is improved. As a result, this FDMM-PF leads to better fairness index (Figure 19). Figure 18 also shows that the unirate multicast methods give better minimum bit rates. Since all users have the same bit rate in unirate context, the minimum bit rate is the same as the average bit rate. These unirate multicast methods equally distribute the resources, but, in the considered 9-user multicast systems where channels of users are very different, it is justifiable to give more resources to some users than others.

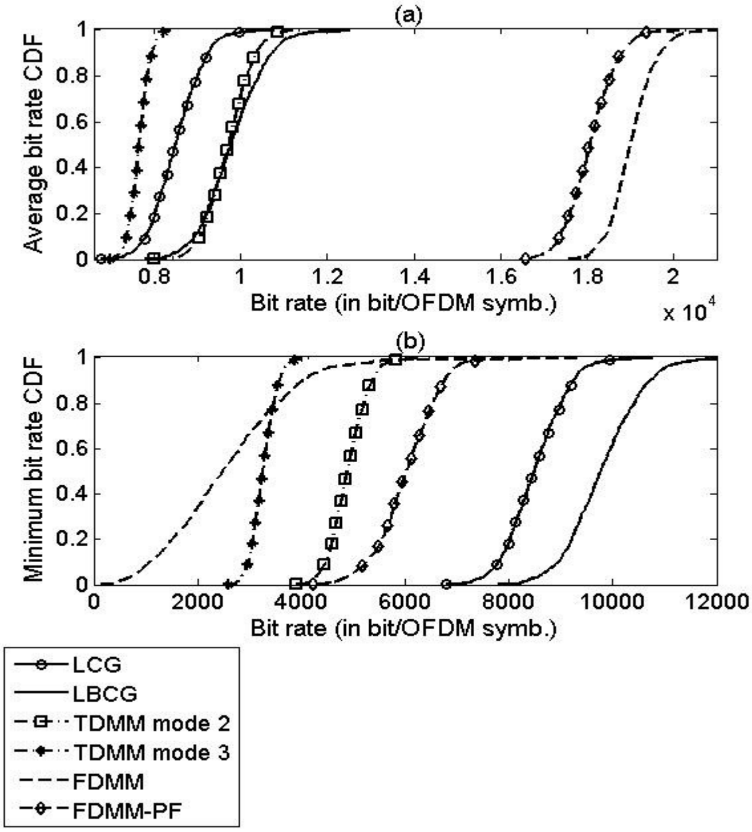


Figure 18: Cumulative distribution function (CDF) of the average and minimum bit rate of multicast users; each user experiences one different channel class and L=32.

As a performance metric, we use the fairness index defined in [52]

$$FI = \frac{(\sum_u R_u(t))^2}{U(\sum_u R_u^2(t))}. \tag{36}$$

Figure 19 shows the cumulative distribution function of the fairness index of multicast users. As expected, the unirate multicast methods give the best fairness index (FI=1). In multirate context, the TDMM methods give better fairness index than the FDMM method. The FDMM-PF method enforces the fairness performance of the FDMM method.

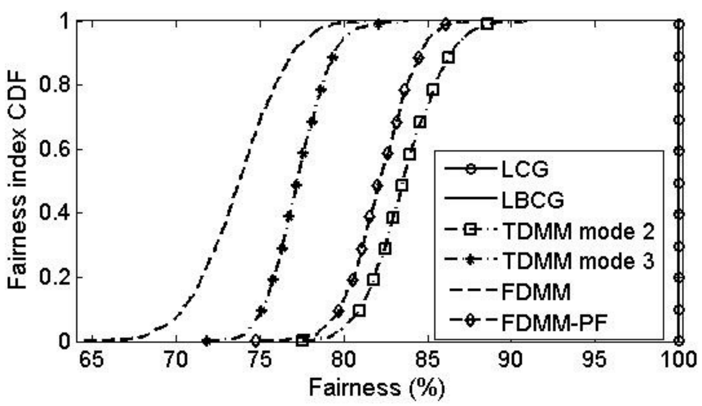


Figure 19: Cumulative distribution function of the fairness index of multicast users; each user experiences one different channel class and L=32.

Besides the comparison of the performance in terms of bit rate and fairness index, the required downlink signaling overheads are compared. In unirate multicast systems, only the modulation order on each subcarrier needs to be signaled to users. In addition to information on the order of modulation on each subcarrier, the multirate multicast systems need to transmit information about the subgroups of users. In FDMM method, the subgroups are not the same for each subcarrier. Thus we can state that, the downlink signaling overhead of the FDMM (or the FDMM-PF) method is higher than the other methods due to the subcarrier/bit allocation information. Furthermore, under the assumption that any combination of layers consisting of multicast data can be decoded at the receiver, an intelligent mapping algorithm for efficiently recovering the original data from different layers is needed [31]. This may bring additional signaling overhead.

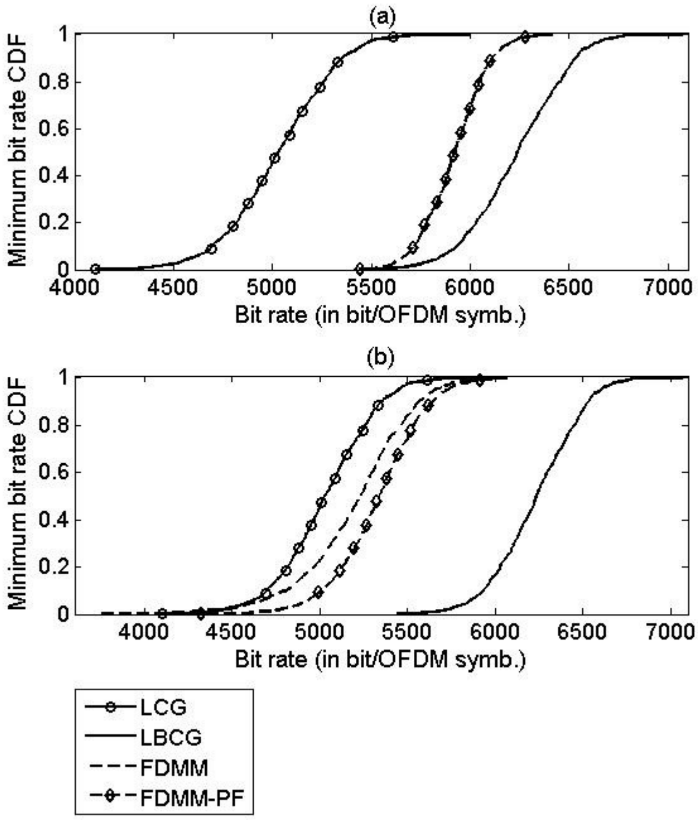


Figure 20: Cumulative distribution function of the average and minimum bit rate of multicast users; each user experiences one class 5 channel and L=32.

In the second step, we consider a multirate multicast system with 9 users and each user experiences the same class 5 channel. This considered case shows the performances of the different multirate multicast methods when multicast users experience similar channels. In this context, the TDMM methods are not performed because they give the same results as the LBCG method. Figure 20 shows the cumulative distribution function of the average and minimum bit rate of multicast users. And, Figure 21 shows the cumulative distribution function of the fairness index of multicast users. In this considered case, the LBCG method is the most suitable method for multicast systems considering both the bit rate and the fairness index. Actually, this method gives the best average and minimum bit rates compared to other methods. However, the FDMM methods remain better than the conventional LCG method in terms of bit rate. Nevertheless, it is worth using the unirate multicast systems with the LBCG method when the multicast users experience similar channels.

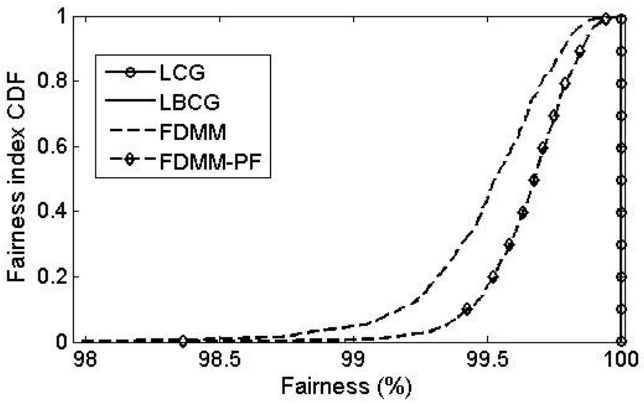


Figure 21: Cumulative distribution function of the fairness index of multicast users; each user experiences one class 5 channel and L=32.

5 Conclusions

This deliverable has presented the adaptation of the HPAV MAC layer to the OMEGA PHY enhancement. It has also analyzed the multi user resource allocation in downlink and multicast contexts. The guard interval has been optimized for the multiuser OFDMA scenario. The optimal value corresponds to the one that maximizes the aggregate rate. With LP-OFDMA systems, the proposed resource allocation algorithms improve the satisfactory of the bit rate requirements of users. In multicast scenarios, results have been given with hierarchical and non hierarchical data encoding. In non hierarchical case, a bit rate gain up to 25% is reached. In hierarchical case, the proposed resource allocation algorithms improve the fairness among users in the case of OFDM systems.

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