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**Broadband Radio Access Networks (BRAN);
Inventory of broadband radio technologies and techniques**

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Contents

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Foreword

This Technical Report (TR) has been produced by ETSI Project Broadband Radio Access Networks (BRAN).

Technical Reports are informative documents resulting from ETSI studies which are not appropriate for Technical Specification (TS), ETSI Standard (ES) or European Norm (EN) status.

A TR may be used to publish material which is either of an informative nature, relating to the use or application of TSs, ESs or ENs, or which is immature and not yet suitable for formal adoption as a standard.

The present document describes the technologies and techniques that are considered applicable to development of normative specifications for BRAN.

Introduction

The present document provides an inventory of technologies and techniques that appear suitable for the implementation of the BRAN as identified in TR 101 031 [4] and TR 101 177 [5]. These networks are intended to support a variety of core networks, including those based on ATM and IP, and provide point to multi-point, multi-point to multi-point networks and point-to-point access at a typical data rate of 25 Mb/s or more. They fall within the following major categories:

- HIPERLAN 2 (HIgh PErformance Radio Local Area Network 2);
- HIPERACCESS (High PErformance Radio ACCESS network);
- HIPERLINK (High PErformance Radio LINK).

HIPERLAN 2, is a complement to HIPERLAN 1 ETS 300 652 [2], ETSI's high speed wireless LAN, provides high speed (25 Mbit/s typical data rate) communication between portable computing devices and broadband Asynchronous Transfer Mode (ATM) and Internet Protocol (IP) networks, aimed at telecommunications access and capable of supporting the multimedia applications of the future. The typical operating environment is indoors. User mobility is supported within the local service area; wide area mobility (e.g. roaming) is supported by standards outside the scope of the BRAN project.

HIPERACCESS is an outdoor, high speed (25 Mbit/s typical data rate) radio access network, providing fixed radio connections to customer premises and capable of supporting multi-media applications (other technologies such as HIPERLAN2 might be used for distribution within the premises). HIPERACCESS will allow an operator to rapidly roll out a wide area broadband access network to provide connections to residential households and small businesses. However, HIPERACCESS may also be of interest to large organizations wishing to serve a campus and its surroundings and operators of large physical facilities such as airports, universities, harbors etc.

NOTE: HIPERACCESS will have no (or very limited) mobility.

HIPERLINK, a very high speed (up to 155 Mbit/s data rate) radio network for sytatic connections and capable of multi-media applications; a typical use is the interconnection of HIPERACCESS networks and/or HIPERLAN Access Points (APs) into a fully wireless network.

Each of the above may be operated in licence exempt or in licensed spectrum. The candidate frequencies range from around 3 GHz up to around 60 GHz. To date only the 5,2 GHz and 17,2 GHz band have been allocated for license exempt applications of BRAN (HIPERLANs and HIPERLINKs). Specific allocations to licensed BRAN (HIPERACCESS) have not been made to date.

Table 1 summarizes the above categories and (provisional) frequency bands.

In view of the variety of frequency bands under consideration, for HIPERACCESS particularly, it is likely that frequency ranges from several areas of spectrum may be designated. These could include frequencies from the microwave range up to the millimetric range and it would seem most likely therefore that a range of BRAN standards may be required encompassing several families of BRAN equipment. As a result the potential number of specifications to be developed for BRAN is very large. Therefore it makes sense to identify commonalties as soon as possible so as to reduce the specification effort. Establishing a common set of technologies as described in the present document should help to identify such commonalties. However, it should be noted that current and future specification work will not be limited to the material covered in the present document.

The technologies and techniques covered are not described in great detail and where possible reference to publicly available sources has been made. However, some of the material contained here is new and public sources are not always available.

A secondary aspect of the present document is that it allows relationships between the various subjects to be identified that may impact the suitability of a given technology for the purposes of a BRAN category.

EXAMPLE: A protocol that relies on broadcasting to maintain synchronization between the network nodes can not be combined with the use of scanning antenna systems that use traffic demands as the main reference for deriving antenna steering data.

Table 1: Summary of current BRAN system types and definitions

BRAN System	Use	Expected majority use	Frequency Band	Mobility	Range	Radio License Regime	Rate Mbit/s	Confi-guration	Comments
HIPERLAN 1	Wireless LAN	Indoor	5,15 to 5,25 GHz [5,25 to 5,3] GHz	Ambulant	50 m	Exempt	20	mp-mp	ERC Decision 96-03 [7]
HIPERLAN 2	Wireless access, ATM or IP	Indoor	Around 5 GHz	Ambulant	50 m	Exempt	25	P-MP	CEPT SE24 currently investigating the possibility of extra spectrum in the 5 GHz area.
HIPERLINK	Wireless infrastructure	Indoor private networks, Outdoor tbd	17,1 GHz to 17,3 GHz	Fixed	150 m	Exempt	155	PP	Was formerly called HIPERLAN 4 in TR 101 031 [4] 100 mW EIRP limit. [6] refers.
HIPERACCESS/E HA/E (Exempt)	Wireless access, ATM or IP	Outdoor, Private Networks	Around 5 GHz	Fixed (support for nomadic users tbd)	0,5 km to 5 km	Exempt	25	P-MP	Previously known as HIPERLAN 3. TR 101 031 [4] defines requirements for HIPERLAN 3. Operation not envisaged in the range 5,15 GHz to 5,25 GHz. See also comment on HIPERLAN/2
HIPERACCESS/U HA/U (Urban)	Urban Fixed Access, ATM or IP	Outdoor, Public Operator	>10 GHz	Fixed	0,5 km to 5 km	Licensed	25	P-MP mp-mp	The amount of spectrum required at various frequencies is under consideration
HIPERACCESS/R HA/R (Rural)	Rural Fixed Access, ATM or IP	Outdoor, Public Operator	<10 GHz	Fixed	0,5 km to 5 km	Licensed	25	P-MP mp-mp	The amount of spectrum required at various frequencies is under consideration

1 Scope

The scope of the present document is identification of technologies and techniques and their characteristics that may prove suitable for the implementation of the physical, Data Link Control (DLC) and subnetwork convergence functions of BRAN as identified in TR 101 031 [4] and TR 101 177 [5]. The BRAN Project is intended to provide specifications for these networks.

The technologies and techniques identified should support the main characteristics of these networks: high data rate, high capacity and uncoordinated deployment. Bringing together these technologies and techniques in a single document facilitates evaluation of their relative merits as well as facilitating the composition and evaluation of combinations.

The present document is not intended to be inclusive or complete: it reflects a state of the art at the beginning of the standardization process. Much of the material is derived from private and public research projects in the field of broadband radio systems. Examples are the EU ACTS projects AWACS, MEDIAN, SAMBA and Magic WAND and the German ATMmobil project.

2 References

The following documents contain provisions which, through reference in this text, constitute provisions of the present document.

- References are either specific (identified by date of publication, edition number, version number, etc.) or non-specific.
- For a specific reference, subsequent revisions do not apply.
- For a non-specific reference, subsequent revisions do apply.
- A non-specific reference to an ETS shall also be taken to refer to later versions published as an EN with the same number.

[1] EN 300 175-3 (V1.4): "Digital Enhanced Cordless Telecommunications (DECT); Common Interface (CI); Part 3: Medium Access Control (MAC) layer".

[2] ETS 300 652 (1996) (including Amendment 1 (1997)): "Radio Equipment and Systems (RES); HIgh PErformance Radio Local Area Network (HIPERLAN) Type 1; Functional specification".

[3] ETS 300 744 (1997): "Digital Video Broadcasting (DVB); Framing structure, channel coding and modulation for digital terrestrial television".

[4] TR 101 031 (V1.1): "Radio Equipment and Systems (RES); HIgh PErformance Radio Local Area Networks (HIPERLANs); Requirements and architectures for Wireless ATM Access and Interconnection".

[5] TR 101 177 (V1.1): "Broadband Radio Access Networks (BRAN); Requirements and architectures for broadband fixed radio access networks (HIPERACCESS)".

[6] CEPT Recommendation T/R 22-06: "Harmonized radio frequency bands for HIPERLAN systems".

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3 Definition and abbreviations

3.1 Definition

For the purposes of the present document, the following definition applies:

access network: A subnetwork providing the means for user devices to access one or more networks, for example an Ethernet network serving as access network to a wide area internet network.

3.2 Abbreviations

For the purposes of the present document, the following abbreviations apply:

AAL	ATM Adaptation Layer
ABR	Available Bit Rate
ACH	Access CHannel
ACK	ACKnowledgement
AM	Amplitude Modulation
AP	Access Point
ARQ	Automatic Repeat and reQuest
ASK	Amplitude Shift Keying
ATM	Asynchronous Transfer Mode
AWGN	Additive White Gaussian Noise
BCH	Bose-Chaudhuri-Hocquent
BDFE	Block Decision Feedback Equalization
BER	Bit Error Rate
BRAN	Broadband Radio Access Networks
BTC	Block Turbo Code
C/I	Carrier to Interference ratio
CAC	Connection Admission Control
CBR	Constant Bit Rate
CLR	Cell Loss Ratio
COFDM	Coded OFDM
CP-OFDM	Constant Peak power OFDM
CRC	Cyclic Redundancy Code
CSMA/CD	Carrier Sense Multiple Access with Collision Detection
DCA	Dynamic Channel Allocation
DDFSE	Delayed Decision Feedback Sequence Estimator
DECT	Digital Enhanced Cordless Telecommunications
DFE	Decision Feedback Equalizer
DIFS	Distributed control function Inter-Frame Space
DLC	Data Link Control
DPCH	Dedicated Packet CHannel
DPSK	Differential Phase Shift Keying
DQPSK	Differential QPSK
DSP	Digital Signal Processing
EC	Error Correction
EY-NPMA	Elimination Yield Non-pre-emptive Priority Multiple Access
FDE	Frequency Domain Equalizer
FDM	Frequency Division Multiplexing
FEC	Forward Error Correction
FER	Frame Error Rate

FFT	Fast Fourier Transform
FOCTC	Frame Oriented Convolutional Turbo Code
GMSK	Gaussian Minimum Shift Keying
GSM	Global System for Mobile communication
HDLC	High level Data Link Control
HIPERLAN	High Performance Radio Local Area Network
IDFT	Inverse Discrete Fourier Transform
IFFT	Inverse Fast Fourier Transform
IP	Internet Protocol
ISI	Inter Symbol Interference
LBT	Listen Before Talk (CSMA)
LOS	Line Of Sight
LSI	Large Scale Integration
MAC	Medium Access Control
MC-CDM	Multi-Carrier Code Division Multiplex
MLSE	Maximum Likelihood Sequence Estimation
mp-mp	multipoint-to-multipoint
M-PSK	M-ary Phase Shift Keying
M-QAM	M-ary Quadrature Amplitude Modulation
MRC	Maximum Ratio Combining
MT	Mobile Terminal
OFDM	Orthogonal Frequency Division Multiplex
OQPSK	Offset QPSK
PA	Power Amplifier
PCF	Point Control Function
PDU	Protocol Data Unit
PHY	PHYsical layer
PIFS	Point control function Inter-Frame Space
PM	Phase Modulation
P-MP	Point-to-MultiPoint
PMR	Peak-to-Mean power Ratio
PN	Psuedo-random Number
PP	Point-to-Point
PSK	Phase Shift Keying
PTS	Partial Transmit Sequence
QAM	Quadrature Amplitude Modulation
QoS	Quality of Service
QPSK	Quadrature Phase Shift Keying
RC	Recursive Convolutional
RCC	Real Channel Connections
RF	Radio Frequency
RM	Reed Muller
RS	Reed Solomon
RSC	Recursive Systematic Convolutional codes
RT-VBR	Real-Time VBR
S-ALOHA	Slotted ALOHA
SAMA	Simple Asynchronous Multiple Access
SC	Single Carrier
SCR	Sustained Cell Rate
SIFS	Short Inter-Frame Space
SISO	Soft Input Soft Output
SNR	Signal to Noise Ratio
SOVA	Soft Output Viterbi Algorithm
TDD	Time Division Duplex
TDM	Time Domain Multiplex
TDMA	Time Division Multiple Access
UBR	Unspecified Bit Rate
U-NII	Unlicensed-National Information Infrastructure (US 5 GHz band)
VBR	Variable Bit Rate
VCC	Virtual Channel Connection

VCI	Virtual Channel Identifier
VPI	Virtual Path Indicator
W-ATM	Wireless Asynchronous Transfer Mode
w-DLC	Wireless Data Link Control
WLAN	Wireless Local Area Network

4 Antennas

This clause briefly introduces a general trend towards the antenna technology which may be applied to the current and future broadband wireless access systems. So far, there have been no antenna technology proposals presented in EP Broadband Radio Access Networks (BRAN) for standardization for BRAN systems. References to relevant documents will be made at the appropriate places in the discussion.

Antenna technology can be specifically applied to combat severe multipath effects, enhance capacity and coverage, reduce interference, and hence increase spectrum efficiency of broadband wireless networks supporting high bit rates.

4.1 Beam patterns

Different antenna types can be categorized with respect to their beam patterns. We distinguish here between "single wide beam antennas" and "narrow beam antennas". The latter type consists of multiple beam antennas as well as single beam antennas.

4.1.1 Single wide beam antennas

Single wide beam antennas can be either omnidirectional antennas or wide sector beam antennas. These antenna types with lower gain offer a wider coverage but cannot distinguish between the desired signal and interfering signals, and are therefore incapable of spatial filtering. Spatial filtering is the process which emphasizes the signal received from a desired direction and de-emphasizes signals received from other directions.

If simultaneous broadcasting from an Access Point (AP) is desired (broadcasting to all mobile terminals at the same time), a single wide beam antenna functionality is the only possible alternative. However, if broadcasting can be performed in a sequence over the different sectors, narrow beam sectorized antennas can be used. Thus, the broadcasting issue in the Medium Access Control (MAC) layer design is closely related to antenna technology.

4.1.2 Narrow beam antennas

The effect of spatial filtering increases with decreasing beam width of the antenna. However, the beam width should only be decreased down to a certain value. Below this value, which is related to the cluster size, the effect is reduced. The cluster size depends on the amount and configuration of scatterers in the vicinity of the mobile terminal [8].

There are several methods by which spatial filtering may be implemented. The first method involves sectorisation, a process in which the cell is divided into a number of angular regions (n). The base station uses directional antennas with beamwidths of (360/n) degrees to cover these regions. The directional antenna "covers" (or beam switches by electronic means) only a specific beamwidth and suppresses signals from neighbouring sectors; thereby reducing the total interference faced by the terminal. Reference [9] shows a measurement study (at 19,37 GHz) investigating the impact of a directional antenna characteristics on delay spread and the K-factor. The results indicate that directional antenna can be used as an alternative to adaptive equalization or multi-carrier transmission. The AWACS project aims to investigate the feasibility of switched beam antenna at the base station.

The results of a multipath measurements study carried out over five different indoor environments using directional antennas at 5,2 GHz showed that a suitably aligned narrow beamwidth antenna makes possible to achieve delay spread reduction through spatial filtering.

However, the study suggests that the system needs to tackle more severe delay spread common to commercial environments which are considered for BRAN applications. Therefore, there will be a need for more sophisticated anti-multipath techniques such as equalization or multi carrier modulations.

Reference [10] describes an electronic beam switching concept using directional antennas known as "time-division beam scanning or time-space multiplexing" in a fixed P-MP system. The analysis indicates that this technique exceeds the performance of conventional P-MP systems by an order of magnitude. The system operates in a semi-duplex mode, where different time slots are used for communication with various fixed terminal stations. In each time slot, the base station redirects its antenna to point to the desired terminal station. The interference in other directions is thereby greatly reduced. In addition, the scanning approach offers flexibility and easy adaptation to new and relocated users.

A sectorised antenna with a switch is closely related to the PHYsical layer (PHY) and MAC layer design. For example, the number of communication channels per antenna sector varies in an uncontrolled way. In general, a complex switch matrix has to be implemented.

Apart from the *fixed lobe* systems described above, an antenna can use *continuous* beam scanning. The advantage of this concept is that the beam direction is not limited by the fixed sector configuration of the antenna. Downlink comparisons between fixed and scanned beams have been performed in field trials [11]. The antenna lobe can thus be directed in exactly the desired direction which leads to increased system performance. This type of antenna could in some sense be called an adaptive antenna (see subclause 4.1.3).

4.1.3 Adaptive antennas

A more sophisticated and advanced technique to introduce spatial filtering involves the use of an adaptive antenna array technology [9]. Adaptive antennas are now regarded as an essential component in future generation mobile networks [12] and as an capacity enhancement in present cellular systems [11]. An adaptive antenna array basically consists of a number of antenna elements combined via an amplitude and phase control network.

Numerous approaches using adaptive antennas have been considered in order to exploit the spatial domain; for example null steering to isolate co-channel users, optimum combining (of phase and amplitude) to reduce multipath fading and suppress interference, and beam steering to focus energy toward desired users.

Thomson CSF Communications suggests exploitation of low cost adaptive antenna technology in both the base station and the mobile terminal for future broadband wireless access systems operating in the 5,2 GHz and 17,2 GHz bands. The adaptive antenna approach is chosen in order to avoid the implementation of complex and power consuming equalizers and /or multi-carrier modulation techniques. This approach is in contrast to ACTS Project TSUNAMI ([12] p.102) where adaptive antennas are placed on the base station side only. The study intends to simulate and implement two types of antennas; 4 to 6 switched sectored antennas with moderate azimuth beamwidth of 70 to 100 degrees and a set of 3 to 6 diversity combining omnidirectional antennas.

4.2 Polarisation

Dual orthogonal polarisation can be used to increase system performance in various ways. Polarisation is used differently depending on the existence of Line Of Sight (LOS) in the particular application.

4.2.1 Single polarisation

In outdoor applications with no LOS the vertical polarisation direction is preferred, since it involves lower path loss [13].

4.2.2 Dual orthogonal polarisation

In the LOS case dual orthogonal polarisation can reduce interference and hence increase system capacity. Interference is reduced by using different polarisations on different communication channels. Cross-polarisation suppression leads to a reduced interference level. For example, by using double radio channels with different polarisations, the system capacity can be doubled within a limited frequency band.

If there is not LOS, orthogonal polarisations can be used for diversity reception. Polarisation diversity has been found a good alternative to space diversity [13].

5 Modulation schemes

The spectrum scarcity in radio communications combined with the increased emphasis on digital transmission has created a need for spectrum utilization techniques which transport the message signal through a radio channel with the best possible quality at a reasonable cost while occupying the least amount of radio spectrum. This goal has led in recent years to the development of new forms of modulation and demodulation schemes applicable to digital radio communications. There are several criteria to compare the performance of digital modulation methods. For example, they may be compared with respect to spectral properties, signalling speed, complexity and the effects of interference, fading and delay distortion on the performance. The decision as to which method is best depends on the specific circumstance of use. Power efficiency and spectral efficiency are among the most important requirements of digital wireless radio systems together with factors such as delay spread and robustness to interference. In the case of broadband systems, signal processing complexity becomes an increasingly important factor. In addition to the above mentioned criteria, the system has to provide Quality of Service (QoS) comparable to fixed networks. Therefore, when selecting a modulation scheme, good interworking with the Data Link Control (DLC) layer has to be ensured.

5.1 Single carrier schemes - Basic characteristics

In this subclause, a brief introduction to the single-carrier schemes is given. The intention is not to compare these techniques, but the presentation of some parameters which should be taken into account when considering them for application in broadband (mobile) radio communications systems.

In a communications system using Single Carrier (SC) modulation the data symbols are transmitted sequentially. It means that the frequency spectrum of each data symbol is allowed to occupy the entire available channel bandwidth. In the case of bandwidth-efficient linear modulation techniques such as M-ary Phase Shift Keying (M-PSK) and M-ary Quadrature Amplitude Modulation (M-QAM), this filtering results in an increased amount of envelope fluctuation in the signal. On the other hand, for power efficiency purpose - it appears that cost-efficient small-size solutions and power efficient solutions are closely related to each other, non-linear transmit Power Amplifiers (PAs) should be used. Due to the Amplitude Modulation (AM)/AM (AM/Phase Modulation (PM) effect is not so considerable for solid state amplifiers) non-linear effect of the transmit PA the envelope fluctuation mentioned above leads to a considerable restoration of the spectral sidelobes that have been previously removed by filtering. To avoid the spectral spreading, either a fairly power inefficient, and as a result fairly expensive, linear PA should be applied or the PA should be operated at a certain output backoff depending on the modulation scheme. For example, in the US digital radio cellular Time Division Multiple Access (TDMA) standard which uses $\pi/4$-Differential Quadrature Phase Shift Keying (QPSK), a 6-dB to 10-dB output backoff is needed in order to increase the spectral efficiency. To cope with the spectral regrowth of non-linearly amplified modulated signal, the class of constant-envelope modulation techniques is well-suited. By using these schemes, power efficient amplifiers can be applied without introducing degradation in the spectrum occupancy of the transmitted signal. Therefore, several practical mobile radio communications systems apply such modulation methods.

As examples, Global System for Mobile communication (GSM) and Digital Enhanced Cordless Telecommunications (DECT) use Gaussian Minimum Shift Keying (GMSK) which is a derivative of MSK (Minimum Shift Keying). The price to be paid for this advantage is the larger bandwidth occupancy by constant-envelope modulation schemes compared to the linear modulation ones. In other words, the achieved power efficiency is at the expense of bandwidth efficiency. Considering the spectral efficiency/power efficiency trade-offs discussed above, it is difficult to determine whether or not constant-envelope schemes provide better power and spectral efficiency than high order linear modulations. In cellular mobile systems there is an additional factor which impacts the spectral efficiency of the whole system: the geographical co-channel reuse. Owing to the re-use, mobile terminals which simultaneously transmit in the same channel in different locations, interfere with each other. This co-channel interference is one of the dominant impairments in cellular wireless systems and has a strong impact on the spectral efficiency of the system. This effect should be taken into account when choosing the modulation scheme. In general, power spectra of modulated signals exhibit sidelobes that may interfere with adjacent channels. For spectral efficiency purposes, a certain amount of filtering is necessary at the transmitter; this needs to be taken into account as well.

Severe multipath propagation is a major problem in mobile communications systems which impacts the signal design (modulation and coding). The spread in arrival times resulting from multipath propagation delays causes transmitted data pulses to overlap, leading to Inter Symbol Interference (ISI). To prevent ISI, the transmission rate of data symbols has to be kept much less than the reciprocal of the delay spread of the channel (equivalently the symbol duration has to be much larger than the delay spread). This assumption is achieved in certain environments, which would therefore provide good opportunities for single carrier modulation schemes: for instance, a fixed wireless access system with a LOS link and directional antennas exhibits generally excellent propagation properties in terms of delay spread. On the other hand, for a Wireless Local Area Network (WLAN) system with data rates of the order of 25 Mbit/s, the use of an equalizer is in most environments indispensable. With increasing data rate, the ISI problem intensifies. This increases the computational complexity of equalization (in the case of Maximum Likelihood Sequence Estimation (MLSE) complexity increases exponentially). A possible solution can be to use single carrier modulation with Frequency Domain Equalizers (FDEs) which have a comparable complexity with multi-carrier modulation schemes and which can cope with a high degree of ISI.

The main implementation difficulty of single carrier systems resides with the digital circuitry used in the equalizer. However, with the advent of rapid progress in the digital signal processing technology, the single carrier technology may offer more flexibility for the future systems.

5.1.1 Single carrier modulation with Decision Feedback Equalizer (DFE)

As explained above, the use of a single carrier modulation scheme in a broadband (greater than 10 Mbits/s) WLAN system will necessitate the use of some form of equalizer in the receivers. The most commonly proposed equalizer types are the MLSE and the DFE [14]. The MLSE complexity increases exponentially with the time dispersion in the propagation, and is generally not regarded as a practical option for a High Performance Radio Local Area Network (HIPERLAN)/2 modem. The complexity of the DFE is however approximately linearly dependent on the time dispersion duration, and consequently the DFE is a very commonly proposed structure for WLAN systems.

A WLAN modem using a single carrier modulation and a "conventional" DFE may have the following characteristics:

- any linear or approximately linear modulation scheme can be used. In particular constant envelope modulation schemes such as MSK and GMSK with moderate time-bandwidth product can be used;

- error performance without coding is good, because the DFE automatically exploits the frequency diversity of the channel;

- signal processing complexity is almost completely concentrated in the receiver. DFE complexity is significant, and increases with data rate or worsening propagation environment;

- a DFE has to be configured every time the propagation channel changes. This may be done by training or by direct calculation of the optimum configuration from a measured channel impulse response. In a WLAN system this will typically occur at the start of every burst reception, and therefore the processing load will be higher than average at the start of a burst;

- typically for DFE systems a significant training overhead is added to the start of every burst, to facilitate the DFE configuration process. The training overhead need only be long enough to measure the channel impulse response, but is often made longer. Longer training overheads may be used for antenna selection, frequency offset measurement, or for training the DFE, if this is preferred to direct calculation of the configuration;

- use of antenna diversity may allow a reduction in the size of the DFE required in a given environment, and hence the complexity/power requirements of the receiver signal processing;

- DFEs are non-linear in operation and, if combined with powerful error control coding, the full coding gain will not be available.

5.1.2 Single carrier with FDE

References: [15], [16] and [17].

Frequency domain equalization provides a generic approach for wideband signal transmission that combines the advantages of (conventional) single carrier and multicarrier transmission schemes. The transmitted signal is organized in blocks with cyclic prefixes. Different modulation schemes can be used (e.g. GMSK, QPSK, 8-PSK, 16-QAM) to meet different channel and hardware requirements while achieving high frequency efficiency. The received signal is equalized in frequency domain by means of Inverse Fast Fourier Transform (IFFT) and Fast Fourier Transform (FFT) operations and complex multiplications. The FDE can cope with a high degree of intersymbol interference (e.g. ISI over 50 symbols) while exploiting multipath diversity which is inherent in single carrier signals transmitted via time dispersive channels.

This modulation scheme can be combined advantageously with antenna space diversity with Maximum Ratio Combining (MRC) in frequency domain. Simulation results show, that diversity can give an Signal to Noise Ratio (SNR) improvement in the order of 10 dB and that an FDE receiver with additional antenna diversity can achieve almost Additive White Gaussian Noise (AWGN) performance.

The main characteristics are as follows:

- constant envelope modulation schemes can be used - it allows for non-linear signal distortions, and no amplifier back-off is required;
- uncoded Bit Error Rate (BER) performance is superior to OFDM because multipath diversity can be exploited without additional channel coding;
- multipath resistance as for OFDM, parameters can be scaled to different application scenarios;
- overall signal processing complexity comparable to OFDM;
- transmitter much simpler than for OFDM because no FFT is required;
- channel estimation can be performed in frequency domain using the same procedures as for signal detection;
- no fine tuning of phase and sampling time is necessary, inherent in channel estimation and equalization;
- no time domain windowing required for spectrum shaping, conventional Nyquist filtering can be applied;
- signal reception requires two FFTs instead of one FFT for OFDM;
- antenna space diversity with MRC requires two receiver chains.

5.1.3 Single carrier with Delayed Decision Feedback Sequence Estimator (DDFSE)

A single carrier modulation scheme with an adaptive equalizer has been demonstrated to meet high speed WATM mobile applications. The device can be realized as a low cost unit with low power consumption, small size and of moderately complex circuitry structure [18], [19] and [20].

The design utilizes a GMSK modulation scheme, which has proved robust against non-linear distortions due to its constant envelope characteristic. The single carrier modulation scheme uses an adaptive equalizer, DDFSE, to counter ISI resulting from multipath effects (time dispersion in the channel). The DDFSE is adopted due to its moderate performance, complexity and fast training. If spectral efficiency is an important factor, offset QPSK (OQPSK) is another choice. OQPSK is not a constant envelope signal, but the envelope fluctuation of OQPSK is smaller than QPSK. That means OQPSK's linearity requirements are much less stringent than those of QAM and OFDM.

The DDFSE can be considered as a combination of the MLSE and the DFE. The complexity of the DDFSE is determined by the number of trellis states (with the minimum value the algorithm reduces to the DFE and at its maximum value the algorithm is equivalent to the MLSE). The DDFSE with GMSK can also operate on a symbol clock rate, without needing a timing control circuit. Decreasing the length of the training sequence is an important factor for a WATM modem to increase the system throughput. The selected training sequence, composed of a repeated sequence of 31 bit Psuedo-random Number (PN) code, is appropriate to transmit 25 Mbit/s over the required ~200 ns delay spread environment. With this training sequence, the DDFSE is able to initialize its internal coefficients within 31 symbol periods.

The 31 bit PN code pattern is also suited for frequency offset compensation. The frequency offset can be easily compensated by detecting the phase difference between the 31 symbol periods.

In the proposed system, antenna selection diversity may be carried out by selecting the receiver antenna with the channel response most appropriate to the DDFSE. This is done using the training sequence . Only one receiver is required. This algorithm is superior to the antenna selection based on signal strength only.

The proposed GMSK-DDFSE system performance has been evaluated by computer simulation and hardware implementation and field test results have confirmed the potential of applying the proposed technology for a high speed WATM system (HIPERLAN 2). Furthermore, the potential of realizing a DDFSE on a single Large Scale Integration (LSI) chip was also evaluated.

5.1.4 Single carrier with frequency ramp

The waveform used in this modulation combines time and frequency. It is based on a fixed frequency short message header followed by a linear frequency swept carrier that covers the entire allocated frequency band. The header is used for the channel access and the frequency ramp contains modulated data and the error correction code.

This single carrier modulation scheme has been demonstrated. The diagram of the demonstrator is shown in [21]. The hardware is classical concerning the digital and radio parts except for the synthesizer which is replaced by a digital ramp generator. The modulation used at the baseband level is a DQPSK. A Reed Solomon (RS) code is used to correct the errors. The proposed solution has so a moderate complexity compared to some equalizers.

One main characteristic of the frequency ramp is to be very efficient against frequency selective fading. As the information is spread over the entire bandwidth, a selective fading corrupts only a very short part of the ramp. A simple Forward Error Correction (FEC) is then able to correct the errors.

5.2 Multicarrier schemes - Basic characteristics

Multicarrier modulation has been proposed as an alternative, which could offer possibilities for alleviating some of the problems encountered with systems using single carrier modulation. In a communication system using multicarrier modulation, the total channel bandwidth is divided into N frequency sub-channels to build a Frequency Division Multiplexing (FDM). The input data stream is then serial-to-parallel converted in order to modulate these N narrowband carriers in parallel. In a frequency selective fading environment, the number of sub-channels N should be chosen in such a way that the fading process in each sub-channel can be considered as a flat fading process, i.e. not frequency-selective.

Coded OFDM (COFDM) technique - characteristics [22], [23], [24].

While the conventional FDM technology prevents the overlapping of transmitted power spectra of the individual sub-channels by applying filtering, the technique called OFDM (orthogonal frequency division multiplexing) enables a more efficient use of the bandwidth: the spectra of the individual sub-channels can overlap provided they respect a specific orthogonality constraint. Due to this orthogonality, the different sub-channels can still be separated in the receiver; the frequency division is achieved not by a classical band-pass filtering, but by doing a dot product between the received signal and a signal base vector. This operation is realized with baseband signal processing.

In fact, the OFDM technique consists in defining elementary modulation symbols as sine waveforms which can be modulated in phase (PSK), in amplitude (ASK), or both (QAM), and on which a rectangular time window is applied. The length of this window corresponds to one OFDM symbol useful duration. Furthermore, it can be shown that the samples of a baseband level OFDM signal are effectively the Inverse Discrete Fourier Transform (IDFT) of the data block to be transmitted in this signal and that the corresponding demodulation process can be realized by taking the DFT of the samples of the received OFDM signal. Hence a bank of real N coherent demodulators is not needed. Both transmitter and receiver can be implemented using efficient FFT techniques.

To avoid any ISI in such a scheme a temporal guard interval (also called cyclic prefix), the length of which has to be as long as the delay spread of the channel, is inserted between consecutive symbols. It has the advantage of being very simple to implement: the guard interval is inserted at the transmitter side and removed by the receiver. It acts like a buffer for multipath absorption.

Since the duration of an OFDM signal is Nfft times (where Nfft is the FFT size, Nfft ≥ N) larger than that of a single carrier signal, it is much more immune to impulse noise.

On the other hand, an OFDM signal is the sum of many independent modulated sinewaves, and as a result its sampled amplitude has an almost Rayleigh distribution. Therefore, its Peak-to-Mean power Ratio (PMR) is much higher than that of a single carrier modulation signal, and it is more subjected to non-linear distortions. This means that, unless specific measures are taken to reduce the peak-to-mean ratio, linearity requirements on the PA or the amount of output backoff is higher compared to the linear single carrier modulation schemes.

Finally, it is important to point out that OFDM technique performs well on frequency selective channels only if it is associated with coding and interleaving. Indeed, OFDM transforms the frequency selective wideband channel into a number of more or less flat fading narrowband channels. As a consequence a number of carriers in a single transmitted symbol may have very different signal to noise ratios. Therefore a coding scheme has to be used to overcome this problem. This is why OFDM is often referred to as COFDM.

Different COFDM schemes have been proposed, corresponding to different frequency multiplex parameters, different sub-carrier modulations and different coding schemes. The PA linearity issues discussed in 5.1 are also relevant for OFDM. Some signal processing can be done on a COFDM signal, in order to lower its PMR, or to reduce its sidelobes, see subclause 5.2.4.1.

Multicarrier modulation can be beneficially incorporated with a diversity technique based on subchannel by subchannel processing, see reference [25].

5.2.1 OFDM parameters - different possibilities

To define the spectrum of the OFDM symbol, the number of modulated carriers as well as the inter-carrier spacing (which is the inverse of the useful OFDM symbol duration) has to be chosen. This choice is directed by the echo handling, the time synchronization aspect, the sensitivity to phase noise as well as the useful bit rate. Depending on these characteristics, an Asynchronous Transfer Mode (ATM) cell could be mapped onto one or multiple OFDM symbols. Some schemes propose the use of "short cells" (less than one ATM cell as payload). The minimum size of a short cell is determined by the amount of data mapped onto a single OFDM symbol.

5.2.1.1 Subvariant 1 - "Small" number of carriers

The main reason for choosing a relatively small number of carriers (from 8 or less to e.g. 32) is given by implementation limits. A small number of carriers will lead to low PMRs, as well as small FFT size. Moreover certain codes and methods applied to reduce this PMR are not suitable/available for a larger number of carriers. On the other hand, for a given net bit rate, and a given spectral efficiency, the less the number of carriers composing the multiplex, the smaller the amount of useful information carried within one COFDM symbol, that is to say the smaller the useful symbol duration. The guard interval has to have a minimal length, given by the environment the system is designed for. Consequently, the smaller the useful symbol duration, the greater the guard interval over symbol length ratio and the more important the loss of spectral efficiency due to this guard interval insertion.

The two following examples give an idea of the parameters which can be chosen for such OFDM schemes:

- A demonstrator exists [26], in which 16 sub-carrier frequencies are used in parallel, with differential 8-PSK modulation, in the time domain, on each one. As the whole symbol duration is short (1,44 µs), the training sequence needed to initialize the differential modulation has the advantage to be short too. A block code, the so-called *complementary* code, is used in the frequency domain. This coding scheme, optimized to provide a lowered peak-to-mean ratio, also provides coding gain (see subclause 5.2.4.1). Time domain raised cosine windowing is used to reduce adjacent channel interference (see subclause 5.2.4.2). The symbol rate is 833 ksymbols/s on every carrier. The coding rate is 0,5. The net bit rate is therefore 833 times 16 carriers times 3 bits per symbol divided by 2, or 20 Mb/s. The 16 sub-carriers are spaced at 1,25 MHz, leading to an overall bandwidth requirement of 20 MHz. This scheme has been designed for indoor applications: it has limited resistance to multipath and is not suitable for short range outdoor.

- A seven sub-carrier multiplex (Nfft=8, the eighth carrier being set to zero) associated to a two stage RS coding scheme, described in subclause 6.3.1.1, has been also proposed to match the perceived frequency selective fading nature of the 5 GHz indoor radio channel. The first RS code protects the system from OFDM sub-carriers lost due to fading events. The second one protects the system against the erasure of a limited number of COFDM symbols within the proposed frame, the structure of which allows for the transfer of single ATM cells. The proposed system is capable of delivering a 50 meter range and about 20 Mbits/sec of peak user data rate using 100 mW EIRPEP, with 7 dB of amplifier back-off. It can also support antenna diversity and Automatic Repeat and reQuest (ARQ).

5.2.1.2 Subvariant 2 - "Large" number of carriers

For a given useful bit rate and spectral efficiency, the more the number of carriers composing the multiplex, the longer the useful symbol duration (i.e. the smaller the inter-carrier spacing), and consequently, for a given guard interval length, the smaller the guard interval over symbol length ratio. Hence, choosing a relatively large number of carriers enables the system to handle safely the propagation echoes with the possibility of minimizing the loss of spectral efficiency due to this guard interval insertion. On an implementation point of view, the larger the number of carrier, the higher the PMR (but the less often this peak is reached), and the bigger the FFT size. The following examples give an idea of the parameters which can be chosen for such OFDM schemes.

- The idea of generating a 117 modulated sub-carrier OFDM signal with an FFT of size 128 has been proposed. In this scheme, the Partial Transmit Sequence (PTS) method (see subclause 5.2.4.1) is used to reduce the PMR.

- A demonstrator exists ([27], [28]) which employs a (55,71) RS codec and a 512-point FFT with DQPSK modulation on each subcarrier (differential encoding is between subcarriers within the same OFDM symbol). A 64 sample preamble and 24 sample postamble are added to fight multipath, ISI and group delay variations. The duration of this 600 sample OFDM symbol on air is 2,667 µs. Synchronization is performed at the portable stations only based upon the autocorrelation of a known sequence broadcast by the basestation. Estimation and correction, in both time and frequency, is all digital. I and Q modulation/demodulation is performed in the analogue domain. Simulation shows the system may offer low BER (10^{-6}) in both line-of-sight (as in the demonstrator) and non line-of-sight/omnidirectional antenna scenarios. This demonstrator works in the 60 GHz band.

- A (relatively) large FFT size (e.g. 512) can be chosen, corresponding to a multiplex of about 350 modulated sub-carriers, allowing both a high immunity against long echoes, and a high spectrum efficiency (i.e. a small guard interval over symbol length ratio, equal to 1/16 or 1/32). Choosing a rather large guard interval (around 800 ns) enables the system to deal safely with echoes due to the propagation in an indoor environment (50 m) as well as in a short range environment (200 m), and moreover to cope with the differences of distance the terminals are from the AP. Hence, there should be no need of any "sophisticated" time synchronization, in which each terminal has to adjust its emission time with respect to the distance it is from the AP (cf. GSM).

  As far as the spectrum shape is concerned, a long symbol duration (around 25,6 µs) induces a rather small inter-carrier spacing (about 40 kHz). This results in reduced out-of-band emissions (due to the sinx/x sidelobes), and the use of time windowing (e.g. of raised cosine type) sometimes proposed to lower these sidelobes can be avoided. On the other hand, the PMR is more important than with less modulated carriers and some specific signal processing might be used to reduce it.

Considering the complexity aspect, the larger the FFT, the more complex the FFT circuit. However, consumer devices based on OFDM with FFT size as large as 8K already exist (DVB-T standards, 3), and with CMOS 0,25 µ technology, a 512 point-FFT circuit is evaluated to be no more than 1,8 mm^2. The net bit rate of the system is about 20 Mbit/s

NOTE: If the number of modulated carriers N is equal to Nfft, inherent aliasing due to the digital signal generation (whatever the FFT size is) occurs. Oversampling the signal could be an option to avoid this. In other terms, the number of modulated carriers, N, should be strictly less than the FFT size, Nfft (0,5 < N/Nfft < 0,75).

5.2.1.3 Scalability applied to OFDM

A fast growing market for radio access networks with all kinds of different requirements and applications will demand low cost, high performance radio technology. This can only be achieved if the radio is build up out of common building blocks that do not have to be (re)designed for each new application area with its own requirements in terms of bandwidth, data rate, delay spread tolerance, error rate or velocity requirements. Instead these common blocks should be scaleable to the specific needs without changing the hardware. OFDM modulation is most suited to such scaling.

5.2.1.3.1 Scaleable OFDM parameters

The scaleable parameters are the (I)FFT size or the number of subcarriers, the guard time, the clock rate, the coding rate and the constellation size. All these parameters influence the characteristics of the OFDM system in terms of rate, bandwidth, delay spread or interference tolerance, power requirements, noise performance or the link budget. For example, data rate can be traded for range, delay spread tolerance, interference resistance or combinations thereof. This also provides adaptability to local interference or noise conditions. In combination with variable multi-level modulation (Differential Phase Shift Keying (DPSK) through n-ary QAM) this inherent flexibility can be further exploited.

Examples have been given for 24 MHz channels including a DPSK system giving 16 Mb/s, 200 ns delay spread tolerance, with good interference resistance ($74{,}10^6$ butterflies/s) and a 64 QAM system giving 72 Mb/s, 2 µsec delay spread tolerance but less interference tolerance ($108{,}10^6$ butterflies/s). It is shown that the Digital Signal Processing (Digital Signal Processing (DSP)) processing power to realize scaling are limited and compare favourably with single carrier solutions.

This scaleable modulation scheme has the usual OFDM characteristics, and besides, it enables adaptability to many environments and requirements, with a minimal effort of specification (only Tx needs to be specified for multiple applications). Hence, selective implementations for niche markets (cost, range, power driven) are possible. This maximizes the scope for innovation. With such a scheme, further development to meet future requirements should be possible.

5.2.1.3.2 Scalabilty achieved with coding

A highly flexible coding scheme [29], based on Reed-Muller (RM) codes, which supports binary, quaternary and octary modulation has been presented. It is suitable for any 5 GHz radio physical layer based on OFDM modulation. It allows simple adaptive changes to the error correction/detection capability, the PMR (see subclause 5.2.4.1), the number of carriers and the choice of binary/quaternary/octary sub-carrier modulation, according to the current channel constraints and system requirements. It provides straightforward encoding and decoding in all cases. With this scheme, simple changes to various code properties in order to deal adaptively with varying noise resistance and delay spread in the channel are possible: if needed, the physical layer can hence evaluate in a simple way.

5.2.2 Sub-carrier modulation

Any linear modulation scheme can be used on the sub-carrier. For instance, if M-ary modulation schemes are used in individual sub-channels, a bandwidth efficiency of $\log_2$ Mbits/s/Hz can theoretically be achieved, which can lead to a very high spectral efficiency. This sub-carrier modulation can be coherent, or differential. In the latter case, this differential modulation can be either on the time or on the frequency axis.

5.2.2.1 Subvariant 1 - Coherent modulation

In the case that coherent demodulation is applied, the estimation of the flat fading envelope in individual sub-channels is necessary. For this purpose, two solutions are possible: insertion of pilot carriers within the multiplex, or insertion of a reference OFDM symbol at the beginning of a Protocol Data Unit (PDU).

Pilot insertion requires the channel frequency response to be almost constant during one OFDM symbol duration. For the system to remain spectrally efficient despite the insertion of a few pilots, this method requires a "relatively" large number of modulated carriers. If a reference OFDM symbol is inserted let say every S symbols, S has not to be too large so as to ensure that the channel is constant over the S symbols duration.

5.2.2.2 Subvariant 2 - Differential PSK in the time domain

Differential phase modulation in the time domain removes the need for channel estimation and carrier phase equalization. For this to be efficient the carrier phase has to be invariant over the order of a symbol period. In most environments this is easily satisfied.

5.2.2.3 Subvariant 3 - Differential PSK in the frequency domain

Differential phase modulation in the frequency domain also removes the need for channel estimation and phase equalization. For this to be effective, the carrier phase has to be invariant over the inter-carrier spacing. Channel phase variation will be rapid near a fade and consequently this will only work well with a large number of carriers, i.e. with a small inter-carrier spacing. If combined with differential encoding along the frequency axis, one OFDM signal is self-containing, that is to say, useful information can be retrieved without the need of other OFDM symbols to get a phase reference. This can lower the receiver complexity and it is well suited for burst transmission.

5.2.3 Coding associated to OFDM

As already pointed out, the OFDM technique has to be associated to a coding scheme. Indeed, this association allows a kind of sub-carrier weighting, which means that in the presence of frequency selective fading, sub-carriers with poor signal-to-noise ratio may be ignored, without any loss of data. Different possibilities are proposed hereafter. When using an error correcting code for this purpose, an important parameter is the code length which should be as long as possible to ensure the best performances. Note that techniques mentioned in subclauses 5.2.3.1 and 5.2.3.2 are more precisely described in subclause 6.3.

5.2.3.1 Subvariant 1 - Concatenation of traditional codes

When the transmitting conditions are severe (frequency selective channel varying in time), the serial concatenation of two codes can be efficient. The inner code, usually powerful, is helped by an outer code which corrects residual erroneous bits:

- concatenation of two RS codes (see subclause 6.3.1.1);
- concatenation of a RS code and a convolutional code;

  this concatenation scheme has already been often used in digital systems. The inner convolutional code is quite robust (e.g. coding rate of about 2/3). The outer RS code deals with the error bursts the Viterbi algorithm can create.

5.2.3.2 Subvariant 2 - Turbo-coding issues

These powerful codes described in subclause 6.3.3.2 (see Block Turbo Codes (BTC) and Frame Oriented Convolutional Turbo Codes (FOCTC)) can also be used as inner codes in a COFDM modulation scheme. Due to the very good performance achieved by such coding schemes, an outer code may not be needed (to be studied by simulations).

5.2.3.3 Subvariant 3 - Sub-carrier selection method

Leaving out weak sub-carriers can enormously reduce the bit error rate. A new algorithm, based on weak sub-carrier selection, was presented. It also enables to reduce the PMR of an OFDM signal (see subclause 5.2.4.1). Feedback from the receiver is needed by the modulator.

5.2.4 Out-of-band emission reduction techniques

5.2.4.1 Subvariant 1 - PMR reduction techniques

Several approaches have been suggested to reduce the PMR of the OFDM signal and thus minimize the out-of-band radiation arising with its amplification.

5.2.4.1.1 Complementary code

The PMR can be controlled by a so-called *complementary* coding scheme [30]: the useful information to be conveyed by the different carriers of the OFDM multiplex is encoded by this code, the coding rate of which is 0,5, and thus the peak power is reduced to just 4 times the average. This corresponds to a gain of 6 dB in peak transmit power and amplifier efficiency [26]. As well as controlling the PMR, the coding provides a performance gain. The minimum distance of the code is four symbols. Therefore any 5 of the 8 carriers in a codeword suffice to recover the data. In practice most combinations of 4 carriers are also sufficient. To remain implementable, the length of this code has to be relatively small. This introduces a restriction on the number of sub-carriers.

5.2.4.1.2 PTS method

This method, described in [28], consists of partitioning the N carriers composing the multiplex into V sub-blocks of N/V carriers. Each sub-block is processed with an IDFT. Each complex vector resulting of each IDFT is then rotated, so as to reduce the PMR in an optimal way. On the implementation point of view, the partitioning can be done in such a way that this method requires V IDFTs of size N/V (and not N) plus some post processing. In that particular case, the complexity with respect to the several IDFTs is approximately the same as that of conventional OFDM. This method can be used for coherent and differentially sub-carrier modulation either in time or frequency direction. The type of modulation alphabet to be used for each sub-carrier is arbitrary. The PTS method is applicable whatever the number of sub-carriers is and is nearly optimum in the sense of approaching the theoretical limit of maximum crest factor reduction vs. redundancy. For this scheme to work, the set of rotation factors applied to the initial signal has to be transmitted (explicitly as side information) to the receiver, so that the data can be recovered by applying the inverse operation. However, if differentially sub-carrier modulation in the frequency direction is used, the system can avoid the explicit transmission of this side information.

5.2.4.1.3 Crest factor reduction by selecting sub-carriers

Considering a low out-of-band radiation the PMR (crest factor) has to be reduced. A new algorithm was presented based on the allocation of non information carrying subcarriers with a reducing function. It is shown, that the bit error rate of an OFDM system can be reduced by leaving out weak subcarriers, so these subcarriers can be used to reduce the peak-to-mean ratio of a muticarrier signal.

5.2.4.2 Constant peak-power OFDM technique

The CP-OFDM scheme features:

1) the constant peak-power signal generation; and

2) the random start-symbol-set generation.

It detects the envelope-peak magnitude of the IFFT output in a symbol-by-symbol manner. It then scales up or down the IFFT output signal of the corresponding symbol according to the detected peak magnitude so that the peak magnitude always becomes an allowed maximum peak value. Based on the assumption of employing differential encoding in the time domain, a set of random data is set to the differential encoder as the start symbol in the beginning of every packet transmission. This avoids a situation in which a data-block comprising a specific information data-pattern (e.g. the unique word, control word, etc.) always ends up having a large peak and is hence error-prone as a consequence of the constant peak-power signal generation method. This also enables ARQ to perform well in the re-transmissions, when the initial packet happens to be faulty because of the large envelope-peak.

From the out-of-band spectrum suppression point of view, CP-OFDM is beneficial, since it can be incorporated with time-domain wave-shaping which allows out-of-band spectrum reduction with a simple implementation. This method gives satisfactory bit/packet error rate performance when the number of sub-carriers is not too large, for example, 32-sub-carrier CP-OFDM exhibits excellent bit/packet error rate performance coupled with high-rate forward error correction.

5.2.4.2.1 Envelope gaussian weighting

This method is described in [31]. Its implementation is possible completely by digital signal processing without any interaction with the HF section of the amplifier. In addition, a predistortion for the linearization of the characteristic of the transmitter amplifier has been integrated into this procedure.

5.2.4.2.2 RM code technique

The coding scheme presented in subclause 5.2.1.3.1, which is based on RM codes, enables to control the PMR, while providing appropriate error correction and detection.

5.2.4.3 Subvariant 2 - time windowing technique

Time domain raised cosine windowing can be used to reduce adjacent channel interference.

5.2.5 Spread spectrum associated to OFDM

5.2.5.1 Subvariant 1 - OFDM with frequency ramp

When the OFDM multiplex is composed of a small number of carriers, a frequency ramp can be used in order to help to combat frequency selective fading. The system consists of the OFDM modulation at the baseband level and then the addition of a frequency ramp instead of a fixed frequency as in conventional OFDM modulations. By adding the frequency ramp, the frequency diversity is increased and hence efficiency against frequency selective fadings is improved. A simple FEC is then able to correct errors. As the coding overhead is reduced thanks to the ramp, the solution increases also the number of available channels within the whole bandwidth. Due to the ramp, the channel response is easily identified and discarded from the received signal before demodulation. Such a scheme, in spite of its simplicity, outperforms the efficiency of an equalizer. An OFDM modulation with 4 subcarriers and a 8-PSK modulation on each subcarrier is proposed. Each subcarrier is spaced by 3,2 MHz. With this solution the gross bit rate is only about 36 Mb/s for a 20 Mb/s payload data rate.

5.2.5.2 Subvariant 2 - Multi-Carrier Code Division Multiplex (MC-CDM) technique

In [33] (see also reference [32]) it is pointed out how OFDM is extended to MC-CDM by employing additional frequency diversity gained from spreading data symbols in the frequency domain. Together with the use of channel coding in frequency direction, a twofold diversity scheme is therefore applied to smooth the severe fading.

Applying sufficiently long symbol spreading (in frequency direction) at the transmitter, together with iterative Block Decision Feedback Equalization (BDFE) at the receiver, approximately transforms the rayleigh fading channel into a set of parallel AWGN channels. Since the spreading can be carried out with a fast orthogonal transform, the additional complexity is moderate (equal to that of the FFT). And because the symbols are spread only over one block in frequency direction (every subchannel contains a portion of each input symbol), no bandwidth extension is necessary. However, the rayleigh fading amplitudes introduce non linear distortion. Orthogonality of the spreading sequences is then destroyed and has to be recovered by the BDFE. The symbol spreading and de-spreading extends the OFDM system to a MC-CDM system.

6 Coding schemes

An engineering solution to error control for BRAN networks should reduce complexity and minimize bandwidth expansion, while at the same time satisfying all QoS requirements. With error control, we mean physical layer FEC and logical link layer ARQ techniques. Due to various conflicting requirements existing for integrated communications in a wireless environment, it is quite complicated to compare ARQ and FEC techniques and difficult to obtain a consensus on a solution fulfilling the aforementioned objectives.

FEC is the only choice, if feedback channels are not available or retransmission is not desirable for some reason. A system using FEC maintains constant throughput which is equal to the code rate regardless of the channel error rate. This is one of the main disadvantages of FEC; the constant bandwidth expansion even when there are no errors. Due to delivery of the decoded message to the user regardless of correctness of decoding result, it is hard to achieve high system reliability with FEC. To obtain high system reliability, a long powerful code has to be used. This results in another disadvantage of FEC; low throughput as well as complex and expensive decoding. ARQ techniques are considered as appropriate schemes to achieve very low bit error rates required by ATM networks. In an ARQ system, a code with good error-detecting capability is employed. Using a proper linear block code, the probability of an undetected error can be made very low. ARQ systems suffer sever drawbacks; firstly, their throughputs fall rapidly with increasing channel error rates, and secondly, in a multiple access system the immediate accessibility of the feedback channel and forward channel cannot always be accommodated. The latter issue can be solved by using DLC protocols that reserve slots for the transmission of ACKnowledgements (ACKs) and the received packets having been detected in error. Note that these schemes need extra capacity and are similar to FEC in spirit. Compared to an ARQ system, the time management of a FEC system is much simpler. A system using FEC transmits its packets serially according to the channel capacity assigned to it. Such a system does not require a feedback channel and its immediate accessibility. Generally, FEC is often preferred over ARQ for services with a timing between source and destination where a medium bit error rate is required. However, for services that are not delay-sensitive and require a very low bit error rate, ARQ is the more proper choice.

The drawbacks in both FEC and ARQ could be overcome if the two schemes are combined in an appropriate manner. A system using such a combined technique consists of an FEC subsystem contained in an ARQ system. The FEC subsystem provides for the reduction of the retransmission frequency by correcting the error patterns that occur most frequently which results in the increase of the system throughput. The function of the ARQ portion is to retransmit the packets in which a less frequent error pattern has occurred and is detected. This results in the increase of the system reliability. Since only a small set of error patterns have to be corrected, a code with a simpler decoder can be used. For FEC systems, both types of codes, i.e. block codes and convolutional codes as well as their combination can be employed.

6.1 Block codes

With very few exceptions, the only block codes of practical importance are linear codes. There is an important subset of the set of all linear codes called cyclic codes. Each code of this subset can be generated by a polynomial. Bose-Chaudhuri-Hocquent (BCH) codes, RS codes, generalised RM codes and quadratic residue codes are some examples of cyclic codes. The encoder of cyclic codes can be implemented in hardware by using a relatively simple feedback shift register. Furthermore, there exist several decoding algorithms that enable one to decode certain of these codes by applying only moderate amounts of hardware. Consequently, cyclic codes are important for practical implementation. Among the cyclic codes mentioned above, BCH codes and RS codes are the most important ones and have been used in several standards and practical systems. Besides error-correcting capability, these codes have also an excellent error-detecting capability which can be used in the ARQ process.

6.2 Convolutional codes

Convolutional codes are the counterpart to block codes. Unlike block codes, where the algebraic properties are very important in constructing good classes of codes and in developing decoding algorithms, convolutional codes are mainly based on engineering approaches and have been developed by computerised searches. In contrast to block codes, the implementation of the maximum-likelihood decoding algorithm for convolutional codes, called viterbi algorithm, is simple. In addition, this algorithm can easily use demodulator soft information allowing relatively large coding gains to be achieved. Consequently, convolutional coding with the soft decision maximum-likelihood algorithm has become in recent years one of the most widely used FEC techniques. Due to the lack of algebraic structure, convolutional codes do not have any error-detecting capability. For a combined FEC and ARQ technique, an additional cyclic redundancy check (Cyclic Redundancy Code (CRC)) code is necessary.

6.3 Concatenated codes

Code concatenation is a practical technique for obtaining a code with a very long block length and a large error-correcting capability. This is accomplished by combining two elementary codes. These codes have two distinct levels of encoding and decoding. The advantage of this coding scheme is that sequential decoding of the different codes can be performed. Thus, the decoding complexity of the overall code depends on the complexity of the decoder associated to each separate code used, this leads to a reduction of the decoding complexity compared to the decoding of a code that have identical parameters of those of the concatenated code.

6.3.1 Conventional concatenation

Conventional concatenation is done by combining in serial two codes. The inner code, usually powerful, is helped by an outer code which corrects residual erroneous bits.

6.3.1.1 Concatenation of two RS codes

This scheme has been proposed associated to a OFDM modulation (see subclause 5.2.3).

The protection of the useful bytes carried within one ATM cell can be, for instance, as follows:

- 53 bytes encoded in 55 bytes with an RS (55, 53). This enables an ARQ process on the ATM cell;
- 1 RS (7, 5), dealing with groups of 3 bits, is applied as an inner code on the 15 bits an OFDM symbol would convey, in this proposed scheme.

6.3.2 Concatenation of a RS code and a convolutional code

This scheme can be associated either to an OFDM modulation (see subclause 5.2.3), or to a single carrier modulation.

In this concatenation scheme, the inner code is chosen to be a convolutional code and the outer code is a RS code. The convolutional code is decoded according to the maximum likelihood criteria by utilizing demodulator soft decision. It enables to lower the BER at the input of the outer decoder. An interleaver is generally placed between the inner code and the outer code in order to break burst errors. However, with the example of parameters given above, this interleaver would be useless because the length of the inner and outer block codes are equal.

Regarding the requirements for transmission on HIPERLAN/2, the global scheme can consist of a RS (86,70) code, obtained by the shortening of the RS (255,239) code, for the outer code and a convolutional 64-state code, of rate R=2/3, for the inner code. To ensure a burst-based transmission, this convolutional code is used as a block code by means of trellis closing. The classical way to close the trellis consists in adding m zeros (if a 2^m state trellis is used) at the end of each block to be transmitted, prior to the coding ("zero-trellis-closing" technique). To avoid the addition of these dummy bits, a cyclic trellis closing has been proposed (tail biting method, [34], [35]): the memory of the convolutional encoder is initialized with the last m bits to be transmitted.

For a transmission over a frequency-selective channel, an optimal performing of the channel decoder is achieved by insertion of a binary random-type block interleaver, of size 1 032, after the inner code, so that the errors are scattered before entering the viterbi decoder.

6.3.3 Turbo codes

The concept of turbo codes is the iterative decoding of two codes concatenated either in parallel or in serial using a Soft Input Soft Output (SISO) elementary decoder. Each elementary decoder therefore provides a decision and a likelihood ratio which quantifies the probability that the decision is correct. This information is passed to the next decoding stage in order to improve the BER at each iteration. The turbo decoder is implemented as a certain number of pipelined identical elementary decoders depending on the number of iterations.

Turbo codes are able to achieve performances with a signal to noise ratio close to shannon's theoretical limit, provided that the code is long enough and that a sufficiently large the number of iterations in the iterative decoding process is used.

NOTE: Concatenation concerns the building of a single powerful code. This code can then be used as the inner code in a concatenation scheme similar as the ones proposed in subclause 6.3.1.

6.3.3.1 Convolutional turbo codes

Convolutional turbo codes are built using a parallel concatenation of two Recursive Systematic Convolutional codes (RSC) separated by a large random interleaver. The elementary decoder used in the iterative decoding process consists of two constituent SISO decoders, one for each RSC encoder, an interleaver and a deinterleaver [36].

6.3.3.1.1 The Soft Output Viterbi Algorithm (SOVA) for conventional turbo codes

In this scheme the two elementary decoders use the SOVA which is a modified version of the viterbi algorithm [37]. A trellis size of 256 states has been used. A frame of 64 information bits is decoded within e.g. 8 iterations. To meet the burst transmission requirement in WATM, this turbo code has been used as a block code, using the "zero-trellis-closing" technique.

6.3.3.1.2 Frame Oriented Convolutional Turbo Codes (FOCTC)

The design of an FOCTC [38] relies on the basic idea that a convolutional code can be used in a burst mode, if we can ensure that the state of the encoder (i.e. the content of the internal registers) is equal to zero (i.e. to its initial value) at the end of each burst. This is obtained by using only one Recursive Convolutional (RC) encoder, in which the data is entered twice, once before interleaving and once after interleaving. For this to work, the interleaver size is highly related to the period of the scrambler associated with the encoder.

The proposed (1 020, 680) block code is composed of a 16 state RC encoder with an interleaver size of 680. As an example, the decoding is performed in 7 iterations.

NOTE: The code dimension of 680 (85 bytes) is suitable for the transmission of one ATM cell plus a few extra bytes, that would have firstly been protected by an RS type outer code, for instance.

6.3.3.2 BTC (iterative decoding of product codes)

A BTC basically consists of a product code which is decoded by an iterative decoding process ([39], [40]). A product code is obtained by concatenating in serial two linear block codes (with parameters (n, k, d)) leading to a matrix with parameters (n^2, k^2, d^2). The iterative decoding of product codes is based on soft decoding and yields to a soft decision of the rows then the columns of the coded matrix. These BTCs have exceptional performances for high code rate (> 0,7) and high coding gain is expected at low BER (< 10^{-5}). In addition, no flattening effect is expected for this block turbo code scheme (serial concatenation) compared to the conventional turbo code scheme (parallel concatenation).

Two examples of BTC were proposed and simulated as a possible (inner) coding scheme for the HIPERLAN/2 physical layer.

6.3.3.2.1 BTC

Simulations of a BTC coding scheme associated to a 8-PSK modulation over a rayleigh fading channel has been simulated. The use of the product code (1 024, 676, 16), composed of two hamming (32, 26, 4) codes, has been proposed as a first attempt to fit the following requirements:

1) code rate about 2/3, in order to obtain a spectral efficiency of 2 bit/s/Hz in conjunction with the 8PSK modulation;

2) block length as long as possible because the longer the code the better the performances.

The block size of 676 bits includes one ATM cell and additional information (external (RS) code, a service channel...). This should be about the maximum useful information to be sent within one codeword, to ensure a maximum granularity of one ATM cell.

The decoding can be performed in e.g. 4 iterations.

6.3.3.2.2 BTC with variable parameters

A solution for designing BTC for variable block size and code rate from a product code (1 024, 676, 16) is proposed in order to fit more accurately the system specifications. The BTC with variable parameters is obtained by applying a shortening and a puncturing technique on the initial product code. This solution can, for instance, yield to coding one ATM cell of 424 bits without any overhead.

Using this coding scheme, a detection scheme can be performed where no additional bits (CRC) are required. This detection scheme performs better at high Frame Error Rate (FER) where a frame consists here of a code word.

6.4 Unequal error protection

Channel coding with unequal error protection is useful for the coding of messages with different sensitivities to channel errors. In DLC layer there are different kinds of messages from the higher layer which can be packed in different PDU. In physical layer of wireless systems we have in general limited redundancy for channel coding. Therefore it is meaningful, channel coding is carried out in terms of sensitivities of PDU from DLC layer. So the most important PDU can be more efficiently protected from channel error than the less important PDU. Several methods for the channel coding with unequal error protection have be suggested and investigated in the literature (e.g. puncture convolutional codes) [37].

7 Medium access techniques

For the purposes of the present document, medium access techniques for multiple access are of interest. There are many excellent texts on this subjects where the most important techniques are explained. A few specific techniques that have been applied to wireless systems warrant mention.

For the layered architecture and the location of the DLC functions including MAC please refer to TR 101 031 [4] and the ETSI/ATM common reference model TR 101 177 [5].

7.1 Distributed control techniques

See references [41] and [42] for a survey of distributed control multiple access techniques. The most relevant systems to be taken into account for BRAN systems are those used in the [44] and HIPERLAN type 1 standards. Both of these systems use carrier sensing in a Listen-Before-Talk (LBT) protocol to support the requirements of the MAC sub-layer protocols.

7.1.1 Passive contention resolution: Carrier Sense Multiple Access with Collision Detection (CSMA/CD)

In carrier sense schemes a device has to sense the radio channel before starting its transmission. This transmission is attempted only if the channel is sensed idle for an appropriate time period, otherwise the device follows a deference or back-off strategy: it waits a predetermined time before trying again to gain access to the channels. The scheme is thus called "Carrier Sense Multiple Access with Collision Avoidance".

The [43] multiple access scheme implements 4 priority levels by defining 4 different listen times before transmission permission is obtained. The highest priority (Short Inter-Frame Space (SIFS)) is used primarily to transmit ACKs, the next priority (Point control function Inter-Frame Space (PIFS)) is used to give access priority to a central controlling function over the main data priority (Distributed control function Inter-Frame Space (DIFS)).

In the de-centralized mode of operation, multiple contenders are separated by randomizing their access delays over a slotted time frame. Should a transmission collide (i.e. two or more devices selected the same slot) the time frame is increased using an exponential backoff scheme similar to [50].

As with [51], the channel performance of [44] is good for bursty data. Performance degrades under excess load owing to the increase in re-transmissions caused by collisions.

Support for time-bounded data via the Point Control Function (PCF) is based on a polling scheme with a contention period for capacity reservation requests and other management and signalling data. The overhead of the PCF protocol is relatively high making it inefficient for the transmission of small units of data.

7.1.2 Active contention resolution: Elimination Yield Non-pre-emptive Priority Multiple Access (EY-NPMA)

EY-NPMA was developed as part of the work on the HIPERLAN type 1 standard 2. EY-NPMA defines both channel free and synchronized channel accesses. The synchronized channel access consists of prioritization, contention and transmission phases. The channel free access consists only of the transmission phase.

In the synchronized channel access, five priority levels and a contention resolution mechanism are coded into a single variable length pulse consisting of a priority assertion and an elimination extension burst. This pulse is transmitted ahead of the data.

The priority assertion pulse is extended in the elimination phase using a geometrically distributed probability of selecting the number of extension slots. The elimination burst is capable of separating a large number of simultaneous contenders. The final yield phase complements the elimination phase to provide a very low residual collision rate.

The HIPERLAN type 1 MAC offers a connectionless data transfer service to the MAC service user, with channel access priority as the mechanism to support time-bounded data. Parameters that can be specified with data transmission requests include packet lifetime and user priority. These parameters are used to determine the priority with which the channel access and control sub-layer will attempt to transmit the packet.

The EY-NPMA mechanism will select for transmission packets from the highest available priority level from the set of contenders and thus provides non-pre-emptive, hierarchical independence of the priority levels and acts as a global scheduler for the distributed HIPERLAN MAC. Packets that cannot be transmitted before their lifetime parameters expire are discarded.

EY-NPMA can be characterized as a stable multiple access system with good overall performance for bursty traffic. The contention resolution mechanism replaces the usual re-transmission-after-collision mechanism for improved channel efficiency.

7.1.3 Frame based distributed control

7.1.3.1 Simple Asynchronous Multiple Access (SAMA)

Within a spectrum allocation where non interoperable systems operate, there is need for an effective method to share the spectrum. Listen-Before-Talk (LBT) has been used as a medium or Radio Frequency (RF) spectrum access method to facilitate sharing. In decentralized RF environments where non-interoperable systems has to share spectrum, LBT has been used with limited success. The problem stems from the fact that not all participants intending to transmit can effectively detect transmissions from other participants, which may either be sources of interference to the intended transmission or be vulnerable to interference from the intended transmission.

If sectored, directional antennas are used in HIPERLAN 2 deployments, the effectiveness of LBT is expected to be further diminished. SAMA is more effective spectrum sharing methodology even when directional antennas are utilized.

In principle, the segment period can be equal to the frame period, however, it is expected that the segment period, t_s, will be a fraction of the frame period, t_f. Participants or pairs of communicating partners would utilize equal length segments, t_s, in sequential frames. To achieve access to the media, a probe signal having the length, t_s, would be transmitted to the intended recipient. The recipient would acknowledge the probe with a response having the same length, t_s, exactly one frame period later. This constitutes channel setup. If the sender receives the ACK, then the portion of the frame period used by the pair of partners during channel setup will be used for the duration of the communication session. If the ACK is not received, then subsequent probe attempts are made at random points in time during subsequent frames. It is possible that this pair of communicating partners will disrupt an existing communications session. If this occur, the interrupted communicating pair would acquire a new channel.

7.1.3.2 Real Channel Connection (RCC) multiple access

A decentrally organized MAC protocol that is proposed for a wireless ATM network (WANET) uses physical channels that result from dividing the given frequency band into FDM channels. FDM channels are separated in Time Domain Multiplex (TDM) channels based on periodic time slots (see figure 1).

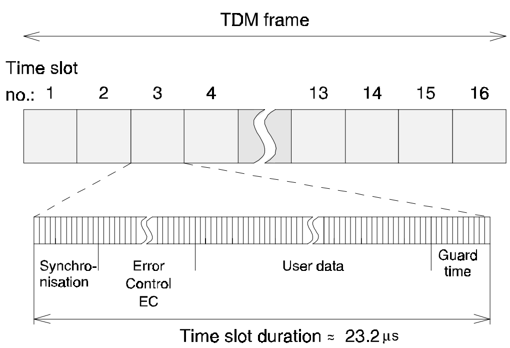

Figure 1: Frame structure

A time slot carriers a burst containing field for synchronization, error control Error Correction (EC) and user data and has some unused space called guard time. The example TDM frame comprises 16 slots.

A logical channel is defined by the carrier frequency and the number of slots used in parallel.

There exist several logical channels, e.g. the Dedicated Packet CHannel (DPCH) is used to multiplex packets of Virtual Channel Connection (VCC), e.g. ATM cells. One or more Access CHannels (ACHs) are used to acquire a DPCH. The number of ACHs is dynamically adjusted depending on the traffic characteristic. For end-to-end data transmission there are dynamically allocated RCC between stations that are made up of DPCHs. Some of the periodic slots might be reserved for use as an RCC to support channel switched connections across single hops of the radio network for ATM Constant Bit Rate (CBR) services.

As illustrated in figure 2 sub-multiplexing of physical TDM channels is possible so that, e.g. one ACH per frame is used (ACH 1/1) or one DPCH is used every third frame (DPCH 1/3).

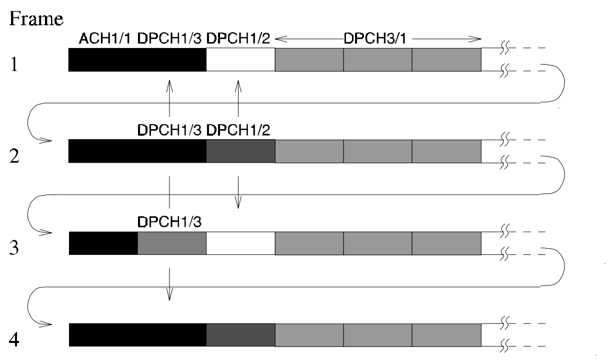

Figure 2: Logical channel structure

A DPCH might use more than one slot per frame, e.g. DPCH 3/1.

Logical channels are assigned dynamically to stations that want to communicate. These RCC are established, used and released for point-to-point communication between stations.

To establish a one-hop RCC to the next station in the network to be able to transport the next packets (ATM cells) a signalling packet is transmitted via the ACH. The packet contains a packet identifier, the address of source and destination stations, the addresses of the transmitter and receiver of the one-hop connection, a connection identifier, a list of sufficiently silent RCC proposed as backward channels, and signalling information of higher layer protocols, e.g. the required QoS. The signalling packet is preceded by a key , e.g. of eight fields each containing 40 bit. Instead of 40 bit information a station sends an energy burst in a field or listens to the channel resulting into 256 different energy burst and listening combinations, resp. keys that can be distinguished. All competing stations that sense another station transmitting during their listening phase defer from transmitting an access PDU in the current slot (see CSMA/CD above). Alternatively the access can be established by means of the S-ALOHA protocol. The surviving station, say S1, sends its access message to the addressee, cf. figure 3, and all other stations in the receive range with radius R_{rx} of station S1 mark the proposed channels contained in that message as reserved for a time duration T_{res}.

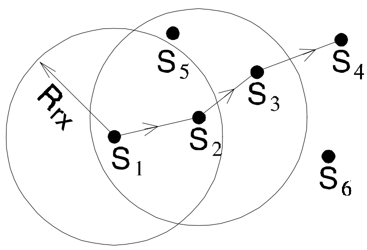

Figure 3: Connection set-up avoiding the problem of hidden stations

The addressed station, say S2 selects one out of the proposed channels according to a minimum required RSSI margin value out of its local channel occupancy list and responds to the calling station on the respective DPCH with an acknowledge packet. By this procedure it is guaranteed that S2 will reach S1 safely and vice versa avoiding the problem of hidden stations.

After an RCC is established the ATM cells are transferred transparently as payload of the data PDU in an contention free Time Division Duplex (TDD) mode.

Bursty traffic sources tend to use a channel that can be described by a packet train model. After an inactive phase of the source, a train of packets (ATM cells) is generated and is characterized by a sequence of consecutive packets until the train ends. For such packet-trains RCC are established. The inter-train gap is a parameter to control the life time of an RCC. A connection is considered as released by all the stations and nodes in the receive range, if the pre-defined inter-train gap has been exceeded. This enables co-ordinated data transmission on VCCs over RCC. The latter can be shared in the frequency or time domain between different stations, resulting into a flexible approach for sharing-rules.

Under asymmetric traffic load a slot in the frame used alternating to carry forward and reverse traffic in a TDD mode of operation. To support asymmetric traffic flows, any combination of forward-to-reverse slots may be agreed by neighboured stations, e.g. 10:1 would assign ten times the capacity forward compared to backwards.

7.2 Centralized control techniques

In a centralized control system one entity (e.g. the AP) controls the bandwidth used by all the other radio units (the terminals). The bandwidth has to be allocated in response to offered traffic. Therefore a signalling protocol is required between the AP and the terminals to inform the AP of resource requests, and to inform the terminals of resource allocations.

In its most general structure, the wireless-Data Link Control (w-DLC) protocol includes two main mechanisms: one for resource request and the other for resource allocation.

The resource request mechanism is utilized by a terminal to request resources for transmission on the uplink; the resource allocation distributes radio resources between the terminals requesting access to the medium taking into account the available amount of transmission resources and aiming at meeting the traffic profile and QoS requirements of all the established connections.

Several schemes have been suggested for the air interface data structure. The schemes considered so far can loosely be split between frame based schemes, and request/response schemes without a regular frame structure. Naturally there are still many possible variations in both approaches.

See references [44], [45], [46] and [32].

7.2.1 Frame based schemes

A frame is a period of time where the access to the channel is pre-organized by the AP. A frame consists of several consecutive slots.

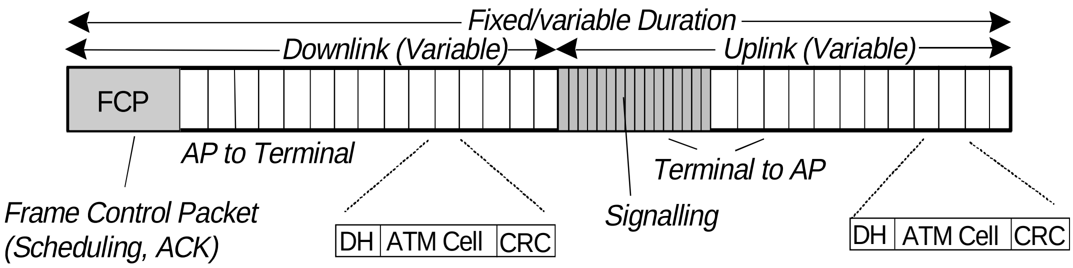

Figure 4: An example of a frame based MAC structure

The transmission frame is split into several sub-parts:

- a broadcast message where the AP transmits system information (FCP). It includes information on how the rest of the frame is used;
- a period for downlink data (AP to Mobile Terminal (MT)) transmissions;
- a period for some part of uplink signalling, e.g. DLC signalling, ATM signalling. Access to the uplink signalling is described in subclause 7.3;
- a period for contention free uplink transmissions from individual MTs to the AP. Access by each MT is controlled by information in the broadcast period. This period may include both data and signalling information.

Each uplink and downlink period consists of several consecutive DLC PDUs, which contain a DLC Header (DH) and one or multiple (cell train) wireless ATM (W-ATM) cells.

The frame structure is affected by the protocol data structure. All proposals include a PDU size appropriate for a single ATM cell payload. Some proposals include the concept of cell trains which are effectively variable sized PDUs equivalent to an integer number of ATM cells. Short PDUs may be used for DLC signalling information (e.g. ARQ ACKs) which cannot effectively fill a complete ATM cell payload.

The length of the overall frame may be fixed or variable. The length of each period can also be fixed or variable (at least for a TDD system). Where variable length transmission periods are used, DLC signalling information is needed to inform the MTs of the exact frame timing. The MTs are synchronized to the frame structure, which is controlled by the AP.

In this scheme, a slot allocation is only valid for one frame. So resource allocation is done on a frame to frame basis.

7.2.2 Request/response schemes

The basic cycle consists of alternating transmissions between the AP and the MTs. Access by the MTs to the channel is controlled by information broadcast by the AP in the immediately preceding transmission. The AP transmissions have to allow for allocating transmissions to specific MTs, to groups of MTs and contention for new MTs (on power-up, or on entering a coverage area).

In its simplest version, medium access is according to a simple ping-pong scheme: the AP sends and the identified terminal answers. Also known as polling for data to send.

By allocating a sharing state to each poll from the AP and adding a sharing request to each MT transmission, the AP can control statistical multiplexing among users of the radio channel. The sharing states may be "empty" (i.e. available for contention), "reserved" (i.e. temporarily assigned to an MT) and "owner" (i.e. assigned to the original owning MT). Then, depending on the traffic load, the system can perform like slotted ALOHA (very low delay), reservation TDMA (intermediate performance) or pure TDMA (maximum throughput).

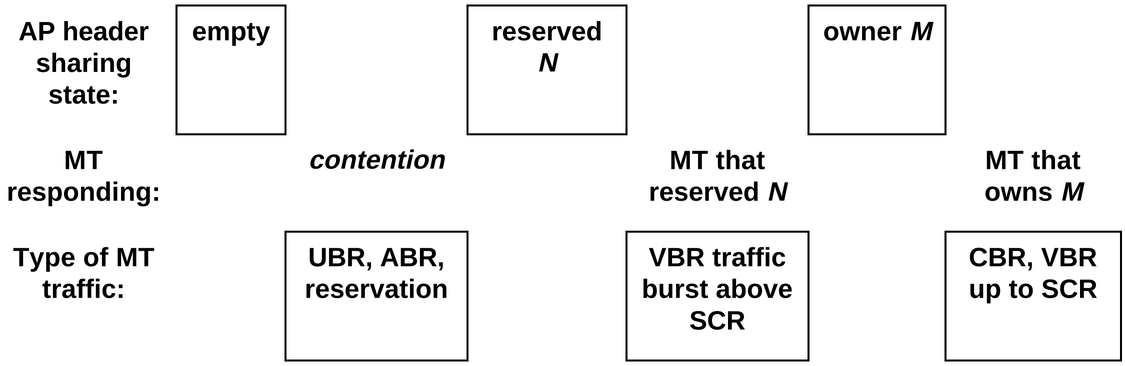

Figure 5: Request/response states behaviour

MTs are quite simple as they basically respond to the most recent AP transmission. Capacity is not wasted in unnecessary polls. As there is no frame structure, the MT may not suffer long delays when critical AP transmissions are not received correctly. Traffic agreements may be supported. However, this approach does not minimize turn-around - the adoption of "cell-trains" for data is desirable for good channel use efficiency.

7.3 DLC signalling techniques

As already mentioned in the introduction of this chapter, centralized control techniques allocate resources to terminals/connections based on load information collected from each terminal.

One of the many tasks of the resource allocation mechanism is to allocate resources for resource request signalling as well. The resource request signalling may be done in a contention or contention-free manner. Hybrid solutions have also been proposed in the literature, based on the combined utilization of polling and random access.

The amount of resources reserved for resource request signalling may be allocated according to either a static or a dynamic discipline: the former allocates a fixed pattern of contention opportunities, irrespective to the status of the current contention attempts and to the level of offered load. The latter adapts the rate of allocation of the contention opportunities to the status and the evolution of the contention attempts.

It is important to note that only a few attempts to compare the performance of the alternative approaches exists: they consider different scenarios and their results cannot be straightforwardly generalised.

Data and control can be strictly separated, for each data transmission a prior signalling had to take place.

Piggybacking involves the transfer of "in band" status information about a connection, e.g. by means of a field in the header of the transmitted w-DLC data units. This is a rather simple approach, whose performance are affected by the need to limit the number of available bits in order to contain the overhead.

Combined control techniques first use out of-band signalling, then piggyback as long as possible. In order to re-enter the transmission cycle out of-band signalling is used again.

7.3.1 Contention based DLC signalling

Multiple MTs are contending for the transmission of DLC signalling. The terminals do not have any dedicated signalling resource , therefore contention resolution mechanisms have to be applied to it, as there are for example: Multiple priority schemes, ALOHA, Slotted Aloha (S-ALOHA), splitting and so on. For further detail please refer to [41] and [48].

7.3.2 Contention free signalling

MTs use a dedicated resource for transmission of their DLC signalling.

7.3.2.1 Polling for resource requests

In order to collect the load information the AP polls the terminals for this information.

NOTE: In comparison to the polling technique described in subclause 7.2.2 the terminals are not polled to transmit data, but only signalling information which in turn is used to allocate slots for data.

Advantage of this kind of scheme is the avoidance of possibly time consuming and complex mechanisms for contention and contention resolution. This kind of polling may be a sub-case of the splitting mechanisms, if each splitting slot is allocated for each terminal/connection. Capacity is wasted when a polled terminal has nothing to transmit on the up-channel.

7.3.2.2 Piggybacking

DLC signalling can be put upon data traffic if available. Piggybacking is limited to active terminals which transmit data.

7.3.2.3 OFDMA/CDMA for resource request signalling

A specific part of frame structure is used for signalling using OFDMA/CDMA.

7.3.2.4 Energy-burst signalling

Eburst is an energy burst of very short duration, which is used to send a "Yes" or "No" signalling message, or effectively a 1-bit signalling message. Typically, the direction of this message is from a mobile terminal to a central administrator such as a base station. Eburst does not require synchronization of the PHY and thus, reduces turnaround time for sending eburst messages. Furthermore, eburst strategy does not require any packetization and hence, can be applied for both fixed and variable-length packets.

Eburst detection uses an energy detection scheme at the PHY. Timing, equalization, demodulation, FEC decoding are not necessary to detect eburst. Eburst detection is more robust than data detection, since the threshold for eburst detection can be set much lower than the threshold for data detection.

There are many ways for transmitting eburst. The simplest implementation is to send a brief spurt of energy, which can be detected by the intended receiver. For OFDM modulation, this spurt of energy can be sent by using only one OFDM symbol plus additional overheads for rise time and fall time. For a single-carrier modulation, eburst is implemented by sending several single-carrier symbols. Thus, the eburst concept can be generalised across many PHY.

7.4 Power management

Increasing use of battery powered devices requires the efficient use of energy for the operation of broadband radio devices. Although this is obvious in the context of portable devices, the same applies for fixed access applications where user devices may have to survive power failure of considerable duration without loss of essential configuration information and operating data.

There are two main areas of power management: transmitter (RF) power management and receiver power management. In the case of broadband systems, the receiver processing is typically complex and requires a large amount of energy. Further, the receiver circuits are typically active for a longer time then the transmitter circuits. Therefore, the energy savings achievable with receiver power management significantly exceeds the energy savings achievable with transmitter power management.

7.4.1 Transmit power control

Power control seeks to reduce the use of RF power at each transmission event to the minimum necessary to achieve a successful information transfer. This reduction may go so far as not transmitting at all if the receiver is subject to significant interference. This form of power management not only reduces energy consumption but also increases spectrum efficiency. The effective implementation of transmitter power management requires knowledge at the sender of the pathloss as well as of the interference conditions at the receiver. Collection of this information requires:

1) carrying the transmitter power level in each message transmitted;

2) communicating local interference conditions;

3) measurement of the quality of the received signal.

Power control is typically implemented by user devices only although spectrum reuse would benefit from applying transmitter power management at the network AP.

Power control is also used in IS95 and GSM.

7.4.2 Power saving

Receiver power management requires knowledge of when a signal will be transmitted and may have to be processed and recognised. Receiver power management may be exercised at two levels: the micro-level in which decisions are made on a per event basis, and macro-level in which decisions are made that span many events.

The knowledge needed for micro-level power management may be implicit as in cellular TDMA systems where devices are assigned a time slot, or it may dynamically determined, e.g. by decoding a header that carries some address information. Micro level power management has minimal implications for the design of medium access and medium sharing techniques.

Macro-level power management requires a message exchange between the two stations involved that describes the time intervals during which a station will not activate its receiver.

Macro-level power management has significant implications for medium access and medium sharing techniques and vice versa since it affects the ability of a device to receive broadcast information. It also requires buffering of traffic at the AP. Obviously, it affects the ability of a device to handle constant bit rate traffic. The same is not true for micro-level power management.

There are also links between power management and user level broadcast and multicast transmissions: for example, an Internet Protocol (IP) level multi-cast can be "replicated" at the DLC level only if all intended recipients are aware of the multi-cast transmissions. This applies to micro-level as well as to macro-level power management.

Macro-level power management techniques are described in references ETS 300 652 [2], [43] and [45].
ETS 300 652 [3] uses unconfirmed multi-casts to announce so-called sleep/wake periods to the environment. The sleep/wake period declaration is made relative to some reference transmitter that regularly transmits a synchronization signal.

Examples of battery saving techniques are described in references [43] and [45].

It should be noted that each of the macro-level power control protocols can also be used to allow a device to declare inactivity so as to be able to scan other RF channels. Therefore there are links between receiver power management and various forms of spectrum re-use.

7.5 Spectrum re-use

7.5.1 Etiquette

HIPERLAN- and Unlicensed-National Information Infrastructure (US 5 GHz band) (U-NII)-systems are expected to work in unlicensed spectrum in the 5 GHz band. Several different types of systems will be allowed to operate within this spectrum. In order to allow different systems to work on the same frequencies, frequency sharing rules have to be applied. In this paper we discuss different options for sharing rules, their impact on different types of systems and define requirements on them for operation of systems guaranteeing QoS to the user.

Systems which offer services with a guaranteed QoS to the users can only be realized with reservation based access protocols, particularly, if a limited maximum delay time is required. Those systems have to be protected and separated from systems which only offer Available Bit Rate (ABR)/Unspecified Bit Rate (UBR) (Available/UBR) services.

Due to their totally different access methods it is difficult to share a common frequency channel between reservation based and non-reservation based systems. A channelisation of the available bandwidth has been proposed. Certain frequency channels should be assigned to each type of system exclusively. E.g. the lowest frequency channel shall be assigned to non-reservation based systems and the highest frequency channel shall be assigned to reservation based systems.

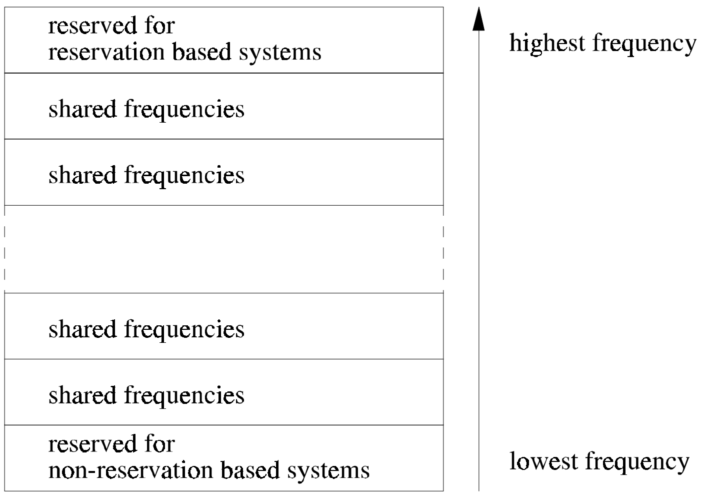

Figure 6: Frequency channelisation

The frequency channels in between are shared between the two types of system using the following rules:

Non-reservation based systems:

1) Look at the lowest frequency channel. If it is silent, start operation here, otherwise go to 2.

2) Go to next higher frequency channel. If it is silent, start operation here. Otherwise go to 2, if there are still frequency channels left, or go to 3.

3) If all frequency channels are occupied, the non-reservation based system starts sharing a frequency channel with another non-reservation based system. Thus it has to determine whether a frequency channel is occupied by a non-reservation based system or by a reservation based system. The non-reservation based system starts sharing a frequency channel occupied by other non-reservation based systems following the rules for non-reservation based systems listed below. This is always possible because at least at the lowest frequency channel a non-reservation based system is working.

Reservation based systems:

1) Look at the highest frequency channel. If it is silent, start operation here, otherwise go to 2.
2) Go to next lower frequency channel. If it is silent, start operation here. Otherwise go to 2, if there are still frequency channels left, or go to 3.
3) If all frequency channels are occupied, the reservation based system has two opportunities: withdraw from communication or share a frequency channel with another reservation based system if this is possible and allowed.

7.5.2 Spectrum sharing

Licensed exempt radio access networks belonging to different owners have to be able to exist side by side in the same frequency band. It has to also be easy to increase the capacity or expand the system by installing new APs. Therefore it is necessary to avoid any traditional frequency planning. One technique that achieves a flexible sharing of the radio resource is Dynamic Channel Allocation (DCA). In DCA all radio resources are available at all APs. An AP can temporarily allocate a channel and the selection of a suitable channel is performed based on a certain quality criteria, e.g. received signal strength , Carrier to Interference ratio (C/I). To obtain a relevant quality criteria both the MTs and the AP needs to make regular measurements and report this to the entity that makes the selection. The main advantage of DCA is through traffic adaptation, i.e. the distribution of radio resources between APs can be made dynamically dependent on the current traffic load. Thus, by using a DCA scheme it is possible for an AP to only use a fraction of the capacity which is offered by a frequency and the remaining capacity may be utilized by another AP. DCA schemes may be performed either centralized or in a distributed form. An overview of the most interesting DCA schemes can be found in reference [47].

7.5.2.1 DCA using containers

In this subclause a channel allocation scheme for wireless ATM that supports the statistical multiplexing is described. The capacity of the physical channel is divided into small parts of equal length, which will be called *container* in the following. Several containers build a frame which is repeated periodically (figure 7). The AP is able to allocate several containers according to the capacity requests of its MTs.

Inside the allocated containers, the access of the AP and the MTs on the physical channel is co-ordinated using a standard DLC protocol. The signalling periods of the DLC protocol are mapped on the allocated containers (figure 8). Observe that there is no fixed relation between a signalling period and a container, a signalling period may consist of several containers. This results in a two level multiplexing. Inside the allocated containers the DLC protocol performs a multiplexing of the traffic of its MT, which results in a dynamic capacity assignment to the MTs. The allocation of a new container happens with a reduced dynamic, since the capacity requirements of an AP are steadier. The allocation of the containers for an AP is similar to the allocation of channels in DECT (Dynamic Channel Selection (DCS)) ETS 300 175–3 [2] for an MT, and similar algorithms may be applied. The main difference with the container approach and more classic DCA systems is that the MTs are clustered into a container in contrast to assigning an individual channel to a MT.

It is not absolutely necessary that the APs are synchronized with each other. Asynchronous APs will lead to a reduction of the maximum traffic load, nevertheless the DCA scheme will still work. It has to be examined under which conditions a (partial) synchronization of the APs is possible. Furthermore, as for every distributed system it is difficult to control the impact a newly assigned channel has on other neighbouring radio cells in terms of interference. It is also of interest to find mechanisms for re-allocation of the available radio resource.

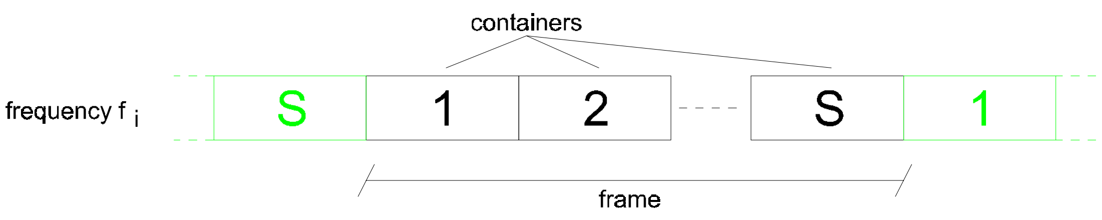

Figure 7: Containers in a frame

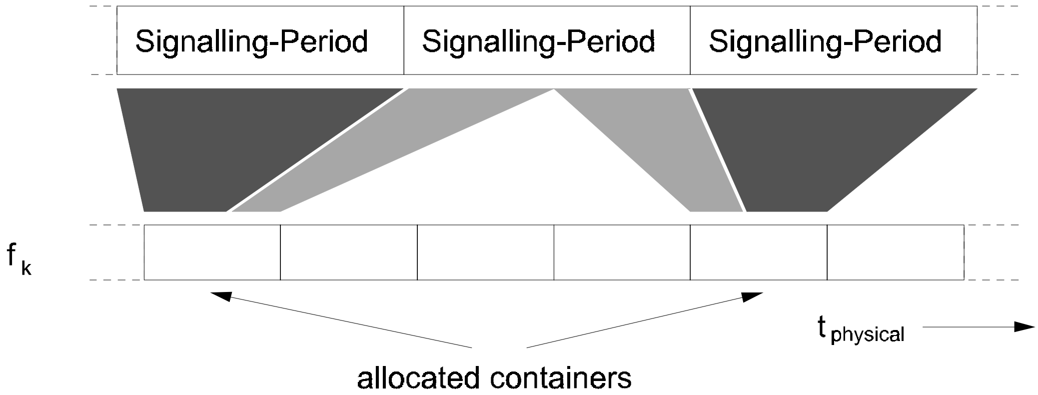

Figure 8: multiplexing of a signalling-period on a container

8 QoS control functions

One of the main goals that DLC should achieve in a W-ATM network is to preserve ATM functionality over the wireless link, while remaining transparent to the ATM connections. Thus, DLC should cope with several impairments of the wireless link and provide QoS to the active ATM connections. Thus, the traffic contract of each ATM connection has to be respected and preserved by the DLC layer.

This clause refers mainly to ATM, but similar issues will apply for non-ATM/BRAN systems.

Some issues concerning QoS control are:

- traffic contract notification;
- means to preserve the traffic contract;
- interaction with ATM Connection Admission Control (CAC) and overall resource management.

8.1 Scheduling

One entity responsible for traffic contract preservation is the scheduler. This entity comprises of the algorithm that decides the cell allocation, the functions gathering the information needed for the algorithm, the functions informing other DLC entities of the algorithm output.

When a new ATM connection is about to be setup, DLC has to have information about its traffic contract. Then, a scheduler will have a complete view of the connection QoS requirements, and will allocate bandwidth for that connection, considering the QoS requirements of the other active ATM connections.

One way to achieve this requirement is by having a scheduling algorithm that can make fair bandwidth allocation, depending on the QoS requirements of each ATM connection. Another way is to apply error control techniques within DLC, to correct or retransmit erroneous information.

DLC is the main entity that has knowledge about the real bandwidth used over the wireless link. This bandwidth refers to the resources needed by the ATM connection, plus the resources needed by DLC/PHY headers and possible retransmissions, resources from DLC control messages, resources allocated for contention. The DLC should notify the resource management entities about the real bandwidth used to allow allocation policies and avoid network congestion.

The scheduling functions may be fully centralized in the AP, partly distributed between AP and MTs or fully distributed among AP and MTs.

The scheduling entity uses the services of the PHYor/and the DLC layer to fulfil the bandwidth requirements of the virtual connections with respect to the desired QoS. If an ARQ scheme is employed within the DLC layer, also the additional DLC signalling information in terms of ACKs has to be considered by the scheduling entity since the transmission of ACKs directly affects the QoS that can be achieved.

8.2 Automatic repeat and request

Compared with fixed networks wireless links offer a poor transmission quality in terms of error rates. But the users of mobile services request the same QoS as in fixed networks normally. Therefore, additional error protection schemes have to be applied in mobile systems.

With an ARQ scheme the ATM Cell Loss Ratio (CLR) is reduced by retransmitting faulty packets. Assuming a perfect error detection code it is possible to achieve a CLR of zero. In other words the CLR with ARQ is only limited by the error detection code, but this is paid with unlimited delays. See references [42] and [41].

The main disadvantage of ARQ protocols is the need to transmit ACKs on the reverse channel. The lack of capacity on the reverse channel slows the forward transmission.

An ARQ protocol for real-time services can retransmit ATM cells as long as a service specific maximum delay is not exceeded. When exceeding its maximum delay, an ATM cell may be discarded.

Discarding ATM cells contributes to avoid and resolve congestion events, since the delay of the following cells can be shortened and the probability to exceed further maximum delays is reduced. Therefore, special procedures may be implemented in order to forcefully resynchronize the receiver and sender state of an ARQ connection to accommodate ATM cell discards.

The achievable performance of the ARQ scheme is highly determined by the efficient and in time transmission of ACKs. In order to develop an appropriate ACK strategy a comprehensive view has to taken, since the performance depends on the employed ARQ scheme, on the services available for ACK transmission by the PHY and/or DLC layer and on the appropriate use of this services by the scheduling entities.

ARQ protocols within the DLC layer are on a link basis. The re-transmission basis is one ATM cell and therefore much shorter than the ATM Adaptation Layer (AAL) packets. Furthermore, the round-trip delay on one link is much shorter then on an end-to-end basis. This enables ARQ protocols within the DLC layer to deal with much higher net cell losses than end-to-end ARQ protocols.

The ARQ protocols within the DLC layer can be executed:

- per ATM connection (per Virtual Path Indicator (VPI)/VCI);
- per ATM service class (CBR, VBR, ABR, UBR);
- for real-time and non-real time connections separately;
- per physical link; or
- combinations of all above.

In general, it is possible to use one type of ARQ for all possibilities, which will be adapted to the needs during protocol execution, this is denoted as "adaptive ARQ". On the other hand it is possible to use special ARQ for each purpose. These are denoted as "special ARQ".

Execution per ATM connection gives the best option to adapt the ARQ to the QoS requirements of each connection. This requires an adaptive ARQ protocol as it is not possible to predict all types of connections. On the other hand it leads to a huge number of parallel instances of ARQ within one terminal and therefore to a huge number of parallel ACKs to be transmitted on the reverse link.

Performing different ARQ for different service classes limits the number of parallel ARQ instances and gives the possibility to use special ARQ per service class. Furthermore, it is possible to use adaptive protocols within some service classes, e.g. Real-Time Variable Bit Rate (VBR) (RT-VBR), to adapt a special VBR ARQ protocol to the actual needs of each VBR connection, e.g. the transfer delay.

The use of two different ARQ is more or less the same as using ARQ per service class. For non-real time connections a standardized ARQ may be used, e.g. High level Data Link Control (HDLC). For real time connections ARQ have to meet special needs.

Using just one ARQ per terminal puts high requirements on this ARQ. It has to be adaptive in the sense, that it has to be able to meet several QoS requirements within one ARQ. Note that this is different from adapting ARQ to one set of QoS parameters. Here, ARQ has to be able to handle different QoS sets within one protocol run. The big advantage is that only one ACK per terminal is necessary. But this may be paid by a larger ARQ send window which leads to longer sequence numbers to be transmitted together with each cell.

Candidate ARQ protocols are go back N and selective repeat protocols (see references [41] and [42]).

8.2.1 ARQ protocols for real-time requirements

An ARQ protocol for real-time services has to retransmit ATM cells as long as a service specific maximum delay is not exceeded. When exceeding its due-date, an ATM cell may be discarded.

Discarding old ATM cells contributes to avoid and resolve congestion events, since the delay of the following cells can be shortened and the probability to exceed further due-dates is reduced. Therefore, special procedures have been developed in order to allow discarding ATM cells within an ARQ protocol which has been designed for no losses at all.

With a go back N ARQ this is no problem as the receiver does not know exactly which packet is missing. Therefore, it is possible to re-assign a sequence number to a different packet.

In conventional selective repeat ARQ protocols discarding of cells or packets is not implemented. After the assignment of a sequence number to an ATM cell, taking the ATM cell out of the sending procedure by discarding it results in a gap in the receive sequence. The receiver will react on this by incessantly requesting a retransmission of the missing cell. Finally a reset of the connection will resolve the deadlock situation. To avoid this, the sender has to inform the receiver after discarding an ATM cell, to which a sequence number has been already assigned to.

Three possible solutions have been proposed and investigated:

- A packet being assigned a sequence number may be discarded. In this case the window will be shifted without waiting for an ACK, enabling further transmissions of newer ATM cells. When receiving the newer cells, the receiver will synchronize to the window shift automatically. This means that the exact execution of the ARQ protocol is temporarily disabled, enabling fast transmissions without error control, until the congestion event has been resolved see reference [49].

- A packet being assigned a sequence number may be discarded. The receiver is informed about the discarded cell by sending a special discard ACK, which in contrast to normal ACKs is sent in the forward direction. As a consequence, discarding ATM cells is only useful if subsequently an efficient transmission of the discard ACK is possible.

- Within the receiver a timer is set which controls the time a packet is requested for retransmission. After that time the window is shifted and the receiver does not wait for the missing packet any longer.

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