Digital Radio Mondiale (DRM) Digital Sound Broadcasting in the AM Bands

Frank Hofmann, Christian Hansen, and Wolfgang Schäfer

***Abstract*—This paper describes the new world-wide broadcasting system Digital Radio Mondiale™ (DRM™) for the long, medium and short wave bands. It is originated by the goal of the consortium members to develop a flexible and efficient audio and data broadcasting standard. Better sound quality, more reliable reception in combination with additional service information make DRM a promising successor to the analogue AM. Achieving good audio quality becomes a challenging task due to the limited bandwidth of 9 or 10 kHz, especially in often strong impaired channels. Many other requirements, which were the basis for the system development, are illustrated. This article gives an overview of the system architecture including the components multiplexing, modulation, channel coding and source coding. Important receiver aspects such as synchronization and channel estimation are also described.**

***Index Terms*—Broadcast, digital radio mondiale, DRM.**

I. Introduction

THE wording Digital Radio Mondiale (DRM) stands for a world-wide initiative to bring AM radio into digital era [1]. This initiative is carried by an international consortium, founded in March 1998, which developed a nonproprietary standard for the broadcast frequencies below 30 MHz. The members are composed of broadcasters, equipment manufacturers, research institutes, regulatory bodies and network providers. The joint effort met in September 2001 in the publication of the ETSI standard [2] and gained approval by the ITU in April 2001.

The long, medium and short wave bands give coverage to large and remote areas as well as in house reception. In combination with better sound quality and more reliable reception as the analogue AM as well as the possibility of additional service information and data services, DRM constitutes high interest of national and international broadcasters.

The channel bandwidth in the long, medium and short wave bands is 9 and 10 kHz respectively. Regarding the state of the art of source coding algorithms a data rate of 20 to 24 kbit/s is required for good sound quality. This results in a needed bandwidth efficiency of more than 2 bit/s/Hz. With these numbers one can deduce, taking also channel coding and signaling overhead into account, that 6 bits per modulation symbol are reasonable, resulting in a 64-QAM modulation. In combination with short-wave channels which suffer from high Doppler spread and delay spread, system design gets a challenging task.

DRM is seen from its members as a complementary system to Digital Audio Broadcasting (DAB) [3] and Digital Video Broadcasting terrestrial (DVB-T) [4]. On the one hand the data rates are much lower, but on the other hand only few transmitting sites are needed for large national or international coverage areas.

To make system design comprehensible as well as to present receiver algorithms, the article is organized to follow the signal flow from the receiver input to the audio output. Beginning more general, Section II describes the most important reasons why DRM is advantageous in comparison to analogue AM. The requirements for system development and a system overview are given in Section III. The basis for each system design is the transmission channel which is explained in Section IV. The more detailed system description starts in Section V explaining the different logical channels used in the signal multiplex. The following Section VI describes how the signal is built and explains the four robustness modes suited for the different transmission conditions. Sections VII and VIII give information about synchronization and channel estimation strategies needed in the receiver. The last two Sections IX and X deal with channel coding and source coding and its adaptation to DRM. The channel coding is based on a multilevel coding scheme which is, despite its performance, rarely used in systems. Source coding consists of three different MPEG-4 compression techniques. To conclude the paper, information about the current status of field tests and a short summary in Section XII is given.

Manuscript received August 1, 2002; revised May 22, 2003.

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Digital Object Identifier 10.1109/TBC.2003.818718

II. Advantages of DRM in Comparison to AM

The analogue AM suffers from poor audio and reception quality. This opens the chance for huge improvements with a digital transmission scheme. Its main advantages are explained in the following:

- **Robustness in fading channels.** In the analogue AM all disturbances as multipath fading, noise as well as co- and adjacent channel interference cause audible impairment. A well designed digital system, which is the main topic of this article, can cope much better with these disturbances.
- **Better audio quality.** The audio bandwidth in double sideband (DSB) AM is less than the half of the transmission bandwidth. Considering additional filters in the receiver an audio bandwidth of 2.4 kHz is typical. The used state of the art source coding algorithms overcome this limitation and deliver a good sound quality at the available bit rate.
- **Power savings at the transmitting side.** When using DSB AM most of the transmitting power is in the carrier. In dependency of the degree of compression of this carrier huge power savings are achievable while maintaining the coverage area.
- **Service related data.** Some years ago it was tried to establish AMDS (Data System in Monophonic AM Sound

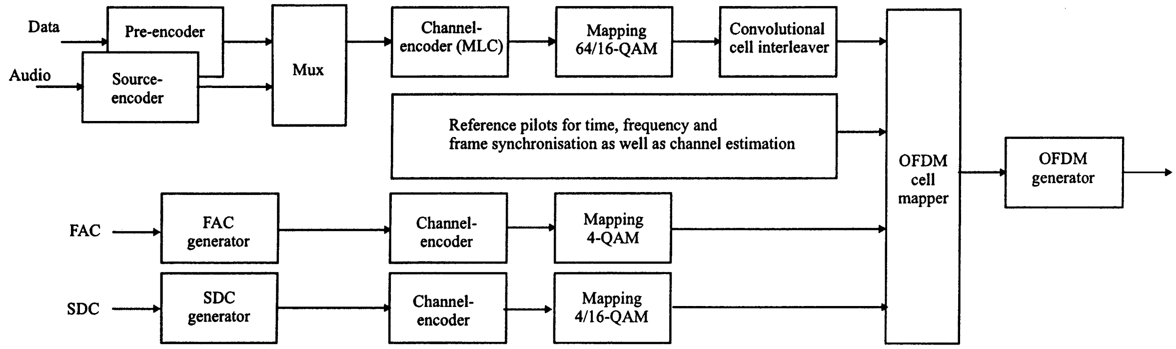

Fig. 1. DRM system overview.

Broadcasting) which failed. The capacity in DRM for service related data as service label, alternative frequency lists and language information is much higher resulting in easy to use radios.

- **Data services.** Digital broadcast systems are independent of the type of content. Besides audio programs data services as traffic and text information as well as pictures are possible.

III. System Overview

A. Requirements for System Development

In order to enable the market success a series of requirements have to be fulfilled. Most of them were adopted by the ITU [5] and were carefully considered when developing the system specification.

- The system is fully compatible with the ITU channel spacing and RF bandwidth. In medium wave the channel grid in ITU region 1 and 3 is 9 kHz, in region 2 it is 10 kHz whereas in short wave the spacing is 5 kHz with 10 kHz bandwidth world-wide.
- To keep the investment for broadcasters moderate, existing modern transmitters are re-usable and expandable with digital modulators.
- It should be possible to build low cost receivers with low power consumption to enable a fast market penetration world-wide. Therefore the system complexity is an important factor.

Additional features are supported to facilitate the introduction of the DRM system:

- Single frequency networks (SFN) are an important feature in digital broadcast systems to reach a high spectral efficiency.
- Some frequency bands below 30 MHz suffer from dramatic changing reception conditions. To cope with this situation alternative frequencies, used by receivers to switch to and from, can be used.
- To facilitate broadcasters with only few frequencies the transition from analogue to digital transmission, simulcast, that means the analogue and the digital signal share one channel with 9 or 10 kHz bandwidth, is possible.
- For the gradual transition from AM to digital a multicast scenario is also feasible. Multicast means the analogue and the digital signal occupy their own separate adjacent channels transmitting from the same station.
- To allow a reorganization of the multiplex during a transmission or the adaption of the transmission parameters reconfiguration possibilities are included.

B. System Parameters

The different transmission modes permit broadcasting with variable channelization constraints and propagation conditions. Besides the nominal bandwidth of 9/10 kHz, the system supports also half channel modes (4.5 and 5 kHz) to allow for simulcast with analogue AM as well as double channel modes (18 and 20 kHz) where planning constraints enable for such facility resulting in larger capacity. The trade-off between capacity and ruggedness to noise, multipath spread and Doppler spread can be defined by the constellation, code rate and the OFDM mode. Considering all parameters a typical data rate in 9/10 kHz channels is 20–24 kbit/s, the maximum data rate in 20 kHz is 72 kbit/s. More detailed parameters are explained in the refering sections.

C. System Architecture

The DRM system is sketched in Fig. 1 which describes the general flow of information on the encoding side.

Up to four audio or data services are source encoded or pre-encoded. For adaption to the transmission capacity and the program content (audio or speech) different audio encoders, namely AAC (Advanced Audio Coding), CELP (Code Excited Linear Prediction) and HVXC (Harmonic Vector eXcitation Coding), are available. They are all included in the MPEG-4 standard. To reach FM-like sound quality the AAC is extended by SBR (Spectral Band Replication), an enhancement tool, which uses information of the spectral envelope of the audio signal to enlarge the audio bandwidth.

The two information channels FAC (Fast access channel) and SDC (Service description channel) carry information about the configuration, that means how the receiver has to decode the signal, as well as service information like the label and alternative frequencies.

For channel coding a Multilevel Coding (MLC) scheme is used in combination with a 16- or 64-QAM modulation and two different depth for the interleaver. Unequal error protection (UEP) and hierarchical modulation are supported.

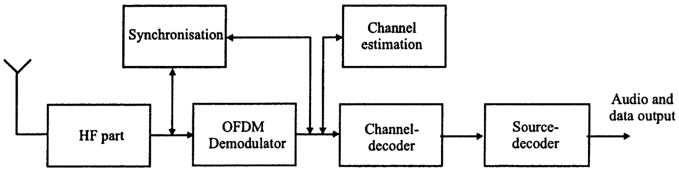

Fig. 2. DRM receiver architecture.

Together with reference pilots for channel estimation and synchronization the OFDM signal is built. Caused by the different propagation conditions throughout the frequencies below 30 MHz, four different OFDM modes are supported.

For the receiver the signal flow has to be mirrored as shown in Fig. 2. The analogue frontend receives the signal including all channel impairment with the antenna.

Some of the synchronization algorithms work in time and some in frequency domain. Also channel estimation is performed after the OFDM demodulation. Channel decoding and source decoding are done with the equalised signal.

IV. Transmission Channel

Digital Radio Mondiale will be applied in the AM-Bands. The AM-band frequencies used for broadcasting contain [2]

- the low frequency band (LF, long wave) from 148.5 to 283.5 kHz, in region 1 only
- the medium frequency band (MF, medium wave) from 526.5 to 1606.5 kHz, in ITU regions 1 and 2 and from 525 to 1705 kHz in ITU region 2
- the high frequency bands (HF, short wave) from 2.3 MHz to 27 MHz world-wide.

In long- and medium wave one can often observe only ground wave propagation represented by the AWGN (additive white Gaussian noise) channel or Rice Channel. In short wave however, reflections at the ionosphere occur. The ionosphere is composed of several layers (D, E, F), which reflect, absorb or attenuate waves depending on their wavelength. The ionosphere structure varies widely over the earth's surface. There is also a dependence on season and solar activity determined by sunspot number. If the wave is also reflected at the earth's surface several reflections can occur as shown in Fig. 3 resulting in a multipath channel.

The movement of the ionospheric layers causes Doppler shift and Doppler spread of the signal. Observations showed that a Gaussian shape for HF-Doppler spread is a good approximation. HF ionospheric channels are nonstationary in both frequency and time, but if consideration is restricted to band limited channels (say, 10 kHz) and sufficiently short time (say, 10 minutes), most channels are nearly stationary and can be adequately represented by a stationary model [6][1]. One suitable model is the wide sense stationary uncorrelated scattering (WSSUS) model. Using the WSSUS model the instantaneous channel impulse response can be written as [7]

$$h(\tau;t) = \sum_n \alpha_n c_n(t) \delta(\tau - \tau_n),$$

[1]Exceptions are typical for sunrise and sunset.

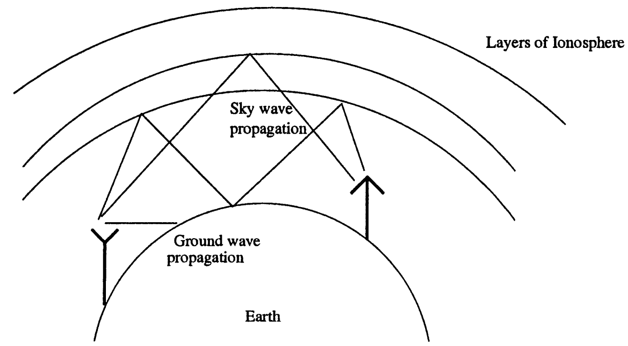

Fig. 3. Wave propagation in the AM bands.

TABLE I
DRM Channel Model 5

parameter	path 1	path 2
delay τ_n	0 ms	4 ms
path gain α_n (rms)	1	1
Doppler shift $D_{sh,n}$	0 Hz	0 Hz
Doppler spread $D_{sp,n}$	2 Hz	2 Hz

where α_n is the attenuation of the n-th path; τ_n describes the delay of the respective path. The time variant tap weights $c_n(t)$ can be described by a complex-valued stationary Gaussian random process. The time variant behavior of each tap weight is characterized by its power density spectrum (PDS). From [6] the PDS of the n-th path can be modeled as

$$S(\tau_n; f) = \frac{1}{\sqrt{2\pi}\sigma_{d,n}} e^{-(f-D_{sh,n})^2/2\sigma_{d,n}^2}, \quad (1)$$

where $D_{sh,n}$ represents the Doppler shift of the n-th path. The Doppler Spread of the n-th path is defined via the standard deviation $\sigma_{d,n}$ in (1):

$$D_{sp,n} = 2\sigma_{d,n}. \quad (2)$$

In order to make the DRM system robust to cope with most transmission channels in the AM bands, a lot of field measurements in different regions of the earth have been carried out. One outcome of these measurements is a set of channels, which is specified in the annex of [2]. These channels were used for performance evaluation of the DRM system. The maximum expected Doppler spread and delay spread are important parameters in the design of the DRM system. As an example Table I shows the DRM channel 5, which represents a severe transmission channel in the high frequency bands due to its large Doppler and delay spread.

V. Multiplex

The DRM multiplex has to meet different requirements. On the one hand a fast service selection in the whole frequency range should be possible, on the other hand a lot of information associated with the program has to be transmitted. Consequently three logical channels are introduced in the multiplex. The Main Service Channel (MSC), the Service Description Channel (SDC) and the Fast Access Channel (FAC).

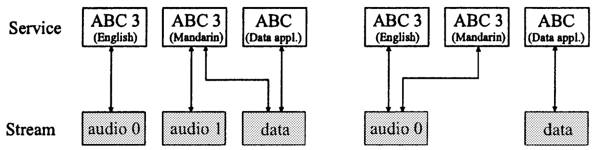

Fig. 4. Two examples of MSC configuration.

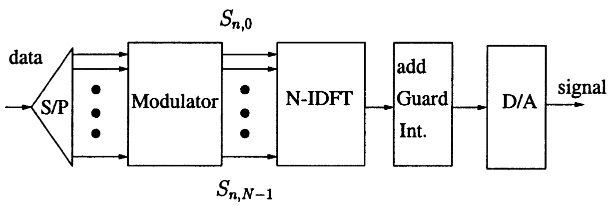

Fig. 5. Structure of an OFDM transmitter.

A. Main Service Channel

The MSC has the highest capacity and contains the data for a maximum of four streams with audio or data content. The services (audio programs and data applications) are mapped to these streams. This structure allows for different services pointing to the same stream and the other way round. Fig. 4 shows an example before and after a reconfiguration.

B. Fast Access Channel

The FAC contains all information which are helpful for service selection when the receiver scans the frequency bands. The two parts of the FAC consists of the channel parameters as spectrum occupancy and constellation of the modulation and the service parameters as a 24 bit service identifier. The FAC within one transmission frame (400 ms) consists of 64 bits including the channel parameters and the most important service parameters.

C. Service Description Channel

The SDC includes the information how the MSC has to be decoded as well as the attributes of the service. These attributes consists of alternative frequencies, service label, frequency schedule etc. The capacity of the SDC is dependent on different parameters as spectrum occupancy and robustness mode and consists of two respectively three OFDM symbols in one transmission super frame (1200 ms). The content of the SDC is divided into data entities with variable size.

VI. Transmission Structure

The DRM system is based on the OFDM transmission technique. The basic principles of the OFDM transmission technique have been already proposed in several publications (e.g., [8], [9]). A principle structure of an OFDM transmitter is shown in Fig. 5. An OFDM signal consists of a sum of subcarriers which are modulated by using phase shift keying (PSK) or quadrature amplitude modulation (QAM)[2]. The individual subcarriers are spaced by the frequency distance ΔF. Considering the transmitted OFDM signal inside the bandwidth of $B = N\Delta F$ and using the sampling theorem, the

[2]DRM uses 4,16 or 64 QAM symbol constellations.

TABLE II
DRM Transmission Modes

parameter	DRM mode			
	A	B	C	D
T_U (ms)	24	21 1/3	14 2/3	9 1/3
T_G (ms)	2 2/3	5 1/3	5 1/3	7 1/3
FFT Size N	288	256	176	112
used carriers (in 10 kHz channel)	226	206	138	88

transmitted OFDM signal must be sampled with the sampling time $t_s = 1/B = 1/N\Delta F$. This leads to an OFDM symbol duration $T_U = N\Delta t = 1/\Delta F$. The samples of the discrete time transmit signal $s_{n,i}$ are calculated as follows:

$$s_{n,i} = \frac{1}{N}\sum_{k=0}^{N-1} S_{n,k} e^{j(2\pi/N)ik} \text{ with } -\infty < n < \infty; \quad 0 \le i \le N-1 \tag{3}$$

where $S_{n,k}$ are the subcarrier modulation symbols inside a single OFDM block. It is worth noting that the complex baseband OFDM signal $s_{n,i}$ is in fact nothing else than the inverse discrete Fourier transform (IDFT) of N input symbols $S_{n,k}$.

In order to avoid distortions from multipath fading channels, the OFDM signal is cyclically extended by a guard time of T_G. As a result, multipath signals with delays smaller than the guard time cannot cause ISI (inter symbol interference).

Because of the difficult propagation conditions in the frequency bands below 30 MHz a set of different OFDM modes is specified in order to cope with different transmission channels. The related OFDM symbol parameters are shown in Table II. It can be seen that only for mode B the IDFT can be replaced by a classical power of 2 FFT. For all other modes a prime factor FFT is used.

The different OFDM modes vary in kind of robustness. Mode A is applied for transmissions over Gaussian channels or channels with minor fading, which are typical for ground-wave transmissions (medium wave, long wave). The other modes can all cope with more time and frequency selective channels, which are typical for sky-wave transmissions (short wave and medium wave at night). Comparing mode C with mode B it can be stated that both have the same robustness against delay spread because of their identical guard interval lengths, but mode C has a higher robustness against Doppler spread because of a shorter T_U. Of course, for a given bandwidth B the possible data rate with mode C is lower than the possible data rate with mode B. Mode D has the greatest guard time and the shortest useful symbol length T_U. Therefore mode D has the highest robustness against severe delay and Doppler spread of all DRM modes for the price of the lowest data rate of all DRM modes. The choice of the best mode is an important issue for the broadcasters. In comparison to other broadcasting applications which also rely on the OFDM principle like DAB and DVB-T it can be stated that for DRM the number of subcarriers is low. As a consequence of the limited channel bandwidth the subcarrier spacing equals approximately only 50 Hz, whereas DAB and DVB-T have a

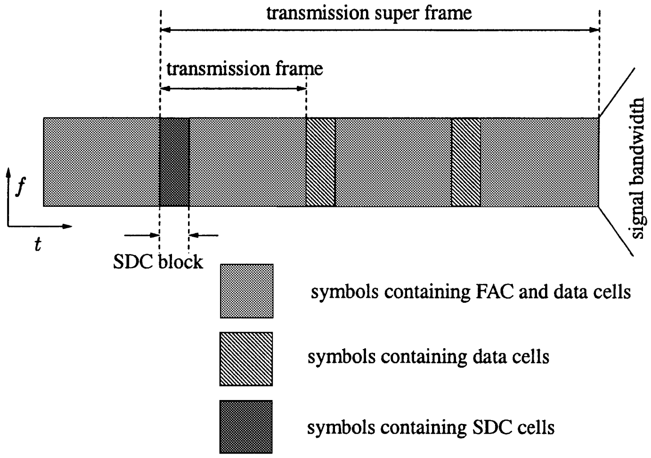

Fig. 6. DRM transmission frame format.

subcarrier spacing in the range of kilohertz. The low subcarrier spacing corresponds to a relatively large OFDM symbol duration (in the range of several ms). Whereas in the most common OFDM systems the guard time is chosen to be $T_G \leq T_U/4$, the very robust mode D is characterized with a very large guard interval of almost the useful symbol duration T_U.

Fig. 6 explains the DRM transmission frame structure. It can be seen that each transmission super frame consists of three transmission frames. Each transmission frame consists of N_S OFDM symbols[3]. It is important to note that all OFDM symbols contain always data and reference information. A complete OFDM reference symbol like in DAB is not available. In more detail, one OFDM transmission frame contains *data cells, control cells and pilot cells*. The control cells consist of FAC and SDC as described in Section V. The SDC is repeated every transmission super frame.

The purpose of the *pilot cells* is twofold: on one hand they can be used for transmission frame, frequency and time synchronization. On the other hand, they are used for channel estimation. Here we want to describe only the structure of the pilot cells. In the following chapters it is outlined, how the pilot cells can be exploited for the various synchronization tasks and channel estimation. The DRM system distinguishes three kinds of *pilot cells*:

- frequency reference cells
- time (transmission frame) reference cells
- gain reference cells

Mathematically a *pilot cell* can be expressed as

$$P_{n,k} = b \cdot e^{j2\pi\vartheta(n,k)}, \tag{4}$$

where b is a real valued pilot boost factor. $2\pi\vartheta(n,k)$ denotes a predefined phase rotation of the pilot cell. The index n represents the OFDM symbol number inside a transmission frame; k denotes the frequency index.

The frequency reference cells are phase continuous pilot tones, which are always present during the DRM signal transmissions on certain subcarriers. The frequency spacing between the DC carrier and the corresponding frequency reference equals 750, 2250 and 3000 Hz, respectively. If these cells are exploited for frequency estimation, the frequency estimation unit can be implemented mode independent. In comparison to the average data cells every frequency reference has a boost factor of $b = \sqrt{2}$.

[3]Mode A,B: $N_S = 15$; Mode C: $N_S = 20$; Mode D: $N_S = 24$

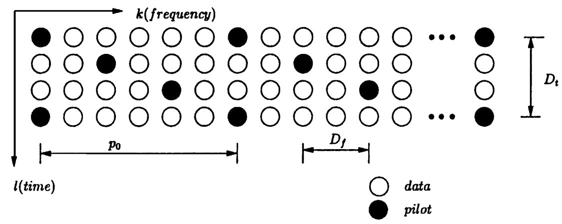

Fig. 7. Pilot pattern for DRM mode B.

TABLE III
GAIN REFERENCE CELLS

DRM mode	p_0 [SC]	D_f [SC]	D_t [OFDM symbols]
A	20	4	5
B	6	2	3
C	4	2	2
D	3	1	3

The time reference cells have a boost factor of $b = \sqrt{2}$, too. In contrast to the frequency reference cells they are only located in the first OFDM symbol of each transmission frame. Their main purpose is to achieve a reliable transmission frame synchronization. In addition they can also be used for frequency offset detection. The exact positions of the time reference cells are depicted in [2].

The largest amount of pilot cells are **the gain reference cells**. These pilot cells are spread equally in time and frequency direction. Fig. 7 depicts the time frequency lattice for mode B.

The gain reference cells are used mainly to get a proper estimate of the channel transfer function. As DRM is a coherent OFDM transmission system a proper estimate of the channel transfer function is necessary for coherent demodulation. From theory it is clear that the number of pilot cells, which are necessary for channel estimation, depends strongly on the expected Doppler spread and delay spread. Therefore the frequency distance distance p_0 of two neighboring pilot cells in one OFDM symbol is correlated with the *wanted* robustness of each transmission mode. Table III summarizes the important issues of the gain reference cells.

In order to give the transmitted pilot signal a more *random-like* waveform, the phase rotation $\vartheta(n,k)$ is chosen in such a way that every gain reference cell gets a unique random phase rotation. Again the boost factor b is chosen to be $\sqrt{2}$ for most of the gain references. Only gain reference cells at the edges of the spectrum and near the DC carrier show a boost factor $b = 2$.

VII. SYNCHRONIZATION

The first task of the synchronization process is to achieve **a coarse timing synchronization**. It is commonly known that a wrong selection of the DFT-demodulation window can lead to strong ISI and has a major impact on all post DFT algorithms (for example the channel estimation). Coarse time synchronization can be achieved by calculating the correlation of parts of

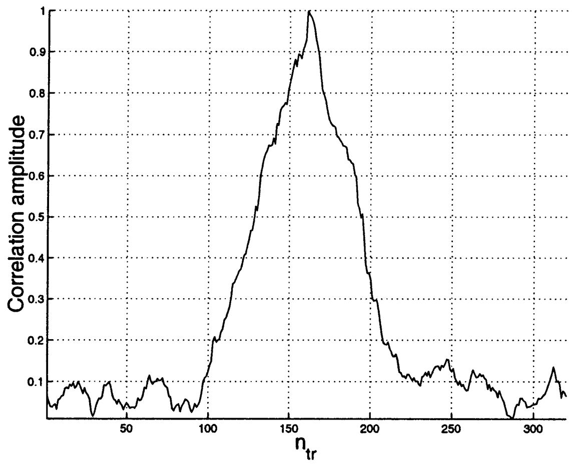

Fig. 8. Example of guard interval correlation amplitude for DRM mode B.

the guard interval with the corresponding parts at the end of the useful symbol. The guard interval based correlation results are visualised for DRM mode B in Fig. 8. An estimate $\hat{n}_\epsilon$ of the start position of the DFT demodulation window is calculated by

$$\hat{n}_\epsilon = \arg\max_{n_{tr}} \left| \sum_{i=n_{tr}}^{n_{tr}+N_G-1} r^*(i) r(i+N) \right|, \tag{5}$$

where r denotes the received signal samples and N_G the duration of the guard interval in samples. The index n_{tr} represents the trial position for the correlation window in (5). The calculation of the correlation in (5) is repeated until a sufficient accuracy is achieved.

Frequency synchronization is the next important task. A frequency offset Δf between the transmit signal s and the received signal r can be introduced by inaccuracies of the receiver's local oscillator or by the transmission channel. The frequency offset Δf can be split into an integral part Δf_I of the subcarrier spacing and a fractional part Δf_F

$$\Delta f = \Delta f_I + \Delta f_F.$$

The fractional part introduces ICI [10], whereas the integral part leads to a shift of all subcarriers at the DFT output.

The frequency synchronization strategy is to establish subcarrier orthogonality as fast and accurately as possible (acquisition) and then maintain orthogonality (tracking). Therefore it is preferable to use the pre-DFT correlation in (5) also for generating an estimate of the fractional frequency offset Δf_F:

$$\Delta \hat{f}_F = \frac{1}{2\pi} \frac{1}{T_U} Angle \left(\sum_{i=\hat{n}_\epsilon}^{\hat{n}_\epsilon+N_G-1} r^*(i) r(i+N) \right).$$

In the next step the received signal samples r are corrected by $\Delta \hat{f}_F$. The integer frequency offset can be estimated by using the time reference cells in every first OFDM symbol of a frame and some neighboring pilot cells in the same symbol. To be more precise, one can calculate a post DFT correlation between the DFT-transformed received signal and the time reference cells. Fig. 9 depicts these correlation results for mode B in DRM channel 5. It can be seen that the correlation gives a distinct peak

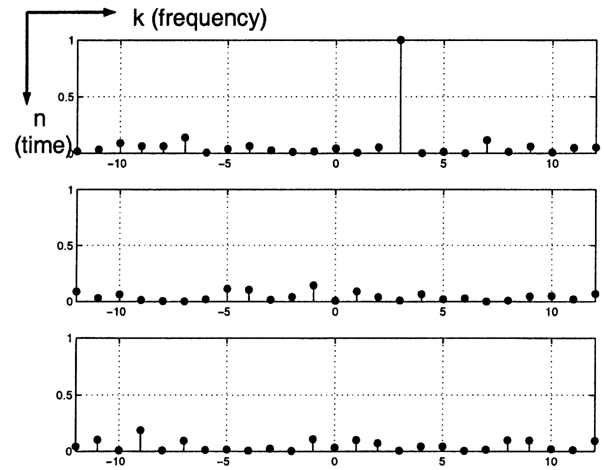

Fig. 9. Snapshot of post DFT correlation for transmission frame synchronization and integer frequency offset detection in DRM channel 5; $\mathrm{SNR} = 5$ dB; $\Delta f_I = 3$; k = trial position in frequency direction; n = trial position in time direction; the indices of the correlation maximum give an estimate for the integer frequency offset and the starting position of a transmission frame.

when the correct received pilot sequence is evaluated and is low for all other values. The position of the correlation maximum in frequency direction corresponds to the integral frequency offset Δf_I; the position of the correlation maximum in time direction corresponds to the beginning of a transmission frame. Therefore the task of **transmission frame synchronization** can be fulfilled with the same method.

It should be mentioned that the three phase continuous frequency references could also be used for the frequency synchronization. In this case the receiver searches only for the frequency positions of the three tones in the received signal r and compares the result with the expected positions of the frequency references. But due to severe frequency selective channels (which is a normal case in short wave) these frequency references could be heavily damaged, so that the frequency synchronization relying only on three frequency references might fail and might delay the whole synchronization process.

After finishing the acquisition phase, the receiver switches into the tracking phase. The tracking phase can be separated into a timing tracking phase and a frequency tracking phase. Frequency tracking can be done for example by using (5) as an error signal for a digital PLL. Fine timing synchronization can be achieved by using an estimate of the channel impulse response (CIR). The optimum method for fine timing synchronization is to maximize the energy of the CIR within a window of size N_G for all possible trial positions n_{tr}.

VIII. Channel Estimation

Coherent modulation demands channel estimation in the receiver to make channel equalization possible. The DRM transmission scheme as presented in Fig. 7 shows OFDM cells with known amplitude and phase, called gain reference cells, in the two-dimensional time-frequency grid. It is well known that the interpolation task can be split into 2 one-dimensional filters, which work sequentially [11]. Considering the formulas for the sampling theorem for the time direction, (7) and the frequency direction, (6) one can deduce that for the expected channel conditions (for example Table I) the interpolation has to be done first in time and than in frequency direction

$$\tau_{\max} \leq \frac{1}{D_f \Delta F} \tag{6}$$

$$f_{D\max} \leq \frac{1}{2 D_t T_S} \tag{7}$$

$T_S = T_U + T_G$ is the OFDM symbol length, $\Delta F = 1/T_U$ the carrier spacing, D_t the distance between pilot cells in time direction and D_f the distance between sampling points in frequency direction (see Table III). Not to violate the sampling theorem, the maximum delay $\tau_{\max}$ between the first and the last path in the delay spectrum of the channel must not be exceeded. Due to the Gaussian shape of the Doppler spread of the individual paths of the channel the maximum Doppler frequency $f_{D\max}$ can be kept only approximatively.

Channel estimation can be done in frequency as well as in time domain whereas the second method is often called DFT-based estimator [12]. Both methods have in common that they estimate the transfer function $\hat{H}$. For practical implementations the three main criteria are performance, complexity of the algorithm and the resulting delay in the receiver. The last point is responsible for the limited possibilities for the filtering in time direction.

The estimators in frequency domain are more common because of the availability of the signal in frequency domain so that no extra transformation is needed. One well known representative is the Wiener filter which is optimum in terms of mean-squared error (MSE). Precondition is the knowledge of the channel correlation function at the gain reference pilot locations. So the Wiener-Hopf equation can be solved

$$\mathbf{W} = \mathbf{\Phi}^{-1}\mathbf{\Theta} \tag{8}$$

where $\mathbf{\Phi}^{-1}$ is the autocovariance matrix of the channel transfer function and $\mathbf{\Theta}$ is the crosscovariance matrix between the channel transfer function and the ideal channel transfer function. The estimated channel transfer function is consequently

$$\hat{\mathbf{H}} = \mathbf{W}^{\mathbf{T}} \cdot \tilde{\mathbf{H}}_{\mathbf{p}} \tag{9}$$

where $\tilde{\mathbf{H}}_{\mathbf{p}}$ is the noisy transfer function at the pilot positions.

Time domain estimators transform the noisy observations of the frequency domain into the time domain by an IDFT. Consequently the fact that the channel power in time domain is concentrated in a few number of samples can be used. Taking the results from [13] one can conclude that the performance is comparable when spending the same complexity with proper design. The mean-squared error (MSE) of two the different methods is illustrated in Fig. 10 for a couple of OFDM carriers.

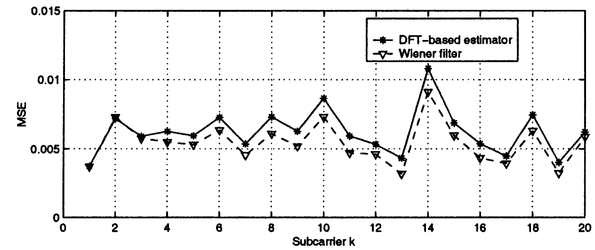

Fig. 10. Comparison of MSE between Wiener filter and DFT-based estimator for channel model 5.

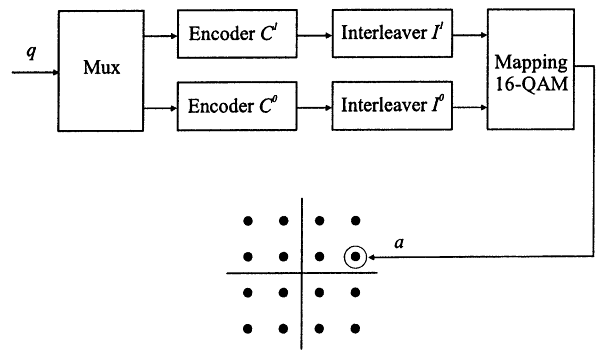

Fig. 11. Multilevel encoder for 16-QAM (4-ASK).

TABLE IV
DRM CONSTELLATIONS AND CODE RATES

Channel	Constellation	Code Rates
FAC	4-QAM	0.6
SDC	4- /16-QAM	0.5
MSC	16- /64-QAM	0.5/0.62 (16-QAM) 0.5/0.6/0.71/0.78 (64-QAM)

IX. CHANNEL CODING AND MODULATION

In DRM a multilevel coding (MLC) scheme is used, whereby channel coding and modulation are jointly optimized, to reach optimum transmission performance. The development goal was a BER of less than 10^{-4}, which is demanded by the source decoder for almost undisturbed audio. This aim has to be achieved with a decoder of low complexity at low as possible SNR values. Additional features as unequal error protection (UEP) and hierarchical modulation are supported. The higher and the lower protected part of the UEP scheme can be used in a flexible ratio to reach a high degree of freedom when assigning the two parts to the multiplex. That means different protections can be reached within one service as well as between different services.

To combat transmission errors also in fading channels a cell interleaver for the MSC with the choice of two different depths besides the bit interleavers of Fig. 11 is applied. Dependent on the predicted time-selectivity of the channel low or high interleaving depth can be chosen with an overall delay of approximately 0.8 or 2.4 seconds. An energy dispersal is used to avoid unwanted regularity in the transmitted signal. Therefore the signal is scrambled by a modulo-2 addition with a pseudorandom binary sequence (PRBS). The different logical channels require different error robustness resulting in various constellations and code rates as presented in Table IV. Additionally the combination of constellation and code rate provides a large degree of flexibility over a wide range of transmission channels.

Initially introduced in [14] and adapted to DRM [15] MLC is a powerful coded modulation scheme. The core of MLC is that more error prone bit positions in the QAM mapping get a higher protection in channel coding. Considering for example the 16-QAM, split into its quadrature components resulting in

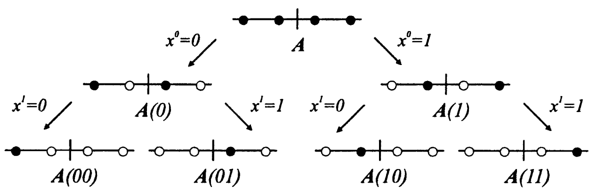

Fig. 12. Partitioning of one-dimension of 16-QAM (4-ASK).

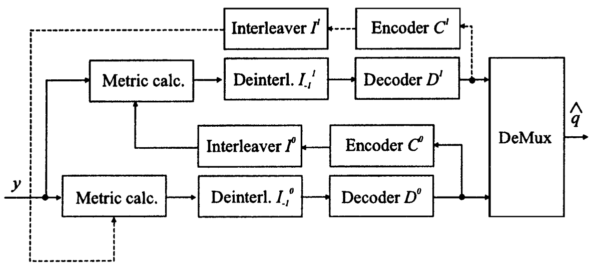

Fig. 13. Multistage decoder for 16-QAM (4-ASK).

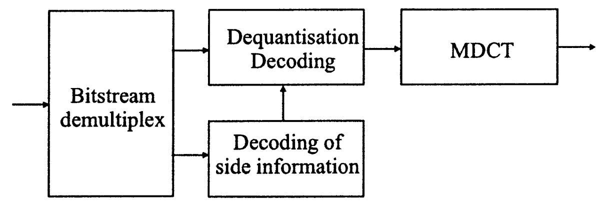

Fig. 14. Principle of audio decoder.

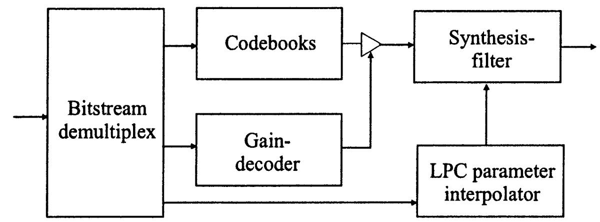

Fig. 15. Principle of speech decoder.

two 4-ASKs, the MLC scheme needs two parallel encoders with two different code rates as depicted in Fig. 11. This one-dimensional consideration of the signal space relaxes substantial the MLC design requirements. The encoder C^0 requires the lower code rate which can be justified with the signal partitioning explained below. DRM uses convolutional codes with constraint length $cl = 7$ as component codes C^i. These have the advantage that one can profit easily from channel state information (CSI) and their well known good performance. The different code rates are obtained by puncturing the mother code. Signal points taken from the signal set $\boldsymbol{A} = \{a_0, a_1, \ldots, a_{M-1}\}$ with M-ary signal alphabet can be described by their address vectors $\boldsymbol{x} = (x^0, x^1, \ldots, x^{l-1})$, $x^i \in \{0,1\}$, where $M = 2^l, l > 1$. Fig. 12 shows the partitioning suited for the 4-ASK. First the signal set is divided into two parts $\boldsymbol{A}(x^0 = 0)$ and $\boldsymbol{A}(x^0 = 1)$ then each part is divided again. Caused by the increasing Euclidean distance the protection of the levels should decrease. The crucial point for the performance of MLC is the appropriate choice of the code rates of the individual levels. Several design methods are explained in [16].

Maximum likelihood decoding of MLC is not realistic because of its huge complexity which is the reason to use multistage decoding (MSD) as depicted in Fig. 13. Each stage receives information of the former stage to reduce multiplicity of possible signal points as well as CSI. The first stage obtains *a priori* information only in the second iteration, which is depicted with the dashed line. Consequently the performance of the decoder can be increased with the number of iterations. The bit-wise interleaving operations between the stages allow the spreading of burst errors between stages.

X. Audio Coding

Since the typical data rates within the 9/10 kHz channels are between 20 and 24 kbit/s, the coding efficiency of the source coders have to be very high to achieve good audio quality. A second requirement for the source coders is that they have also to work in error prone channels due to the fact, that wireless broadcast systems never obtain error free transmissions. Based on several listening tests within the DRM consortium, partly performed together with MPEG [17], speech and audio coding algorithms were chosen. These algorithms are all part from the MPEG-4 [18] standard, namely the Advanced Audio Coding (AAC), Code Excited Linear Prediction (CELP) and Harmonic Vector eXcitation Coding (HVXC). AAC is a generic audio coder working at a wide range of bit rates, whereas the other two are natural speech coders suitable for low and ultra low bit rates.

Speech and audio coding algorithms follow different principles. Perceptual audio coders reach irrelevance and redundancy reduction with the help of a psychoacustic model. Thus only the perceivable signal parts are transmitted. The decoder block diagram is drawn in Fig. 14. After demultiplexing the bitstream is dequantized according to the content of the side information. The 960-point MDCT (modified discrete cosine transform) provides high frequency resolution, resulting in high coding efficiency for tonal and harmonic signal components. For critical time domain signal parts the TNS (temporal noise shaping) tool is able to shape the quantization errror in time domain without changing the frequency resolution.

For speech coding algorithms the human vocal tract is simulated by an excitation generator and a synthesis filter bank. Fig. 15 depicts the speech decoder. The spectral envelope of the signal is described with LPC (Linear Predictive Coding) parameters. The excitation generator consists of the codebooks, containing periodic and random signal components. Both can be amplified independently by gain factors. Synthesis filter and post filter form the output signal.

AAC has been developed within MPEG-2 to improve the compression efficiency to its predecessor algorithms and to enable various bit- and sampling rates. Since its good technology AAC became part of MPEG-4 with new functionality as error robust coding. This format specifies a new bitstream syntax, offering higher resilience in case of error prone channels. In combination with UEP, as explained in Section IX, also graceful degradation is achieved. To reach alignment to the DRM transmission frames of 400 ms, as explained in Section V, a transform length of 960 is used, corresponding to audio frames of

TABLE V
CHARACTERISTICS OF DRM AUDIO CODECS

Coder	Sampling frequency f_{sa}	Bit rates
AAC	12, 24 kHz	wide range, granularity of 20 bit/s
CELP		
Narrowband	8 kHz	3.85 to 12.2 kbit/s
Wideband	16 kHz	10.9 to 23.8 kbit/s
HVXC	8 kHz	2.4 and 4.66 kbit/s

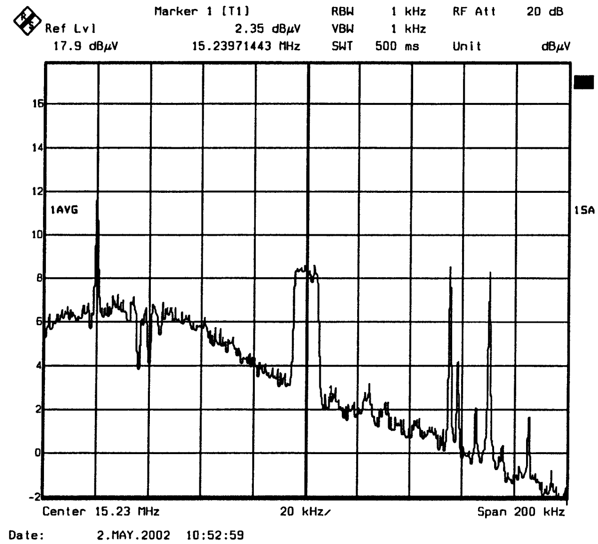

Fig. 16. Spectrum of received DRM signal.

80 ms ($f_{sa} = 12$ kHz) and 40 ms ($f_{sa} = 24$ kHz), respectively. Considering 5 or 10 audio frames with variable bit length, a block with fixed length is build, called super audio frame. Consequently no additional synchronization is needed for the audio coding. AAC supports generic stereo, mono and joint stereo.

Additionally the enhancement tool SBR (Spectral Band Replication) is used to increase the audio bandwidth of the codec to more than 15 kHz. The concept uses the effect that the harmonic series of the audio signal, truncated in the codec, can be extended based on the relation between lowband and highband components. Thus only information about the spectral envelope of the highband has to be transmitted to reconstruct the highband in the decoder. AAC and CELP can profit from SBR at low bit rates.

The MPEG CELP speech coder is offered to allow reasonable speech quality at low bit rates (see Table V). Therefore two major advantages can be seen. First, speech applicatations with two or three languages can be simultaneous transmitted, which is of great interest in international broadcasting. Second, if the available bit rate is low because of simulcast transmission or extremely bad channel conditions, the CELP coding is the preferred candidate. The two used versions of the CELP coder, namely wideband and narrowband, provide scalability in bitrate and bandwidth. The audio bandwidth is 100 to 3800 Hz and 50 to 7000 Hz, respectively.

The third algorithm, the HVXC opens up new applications for DRM as multilanguage services and cheap storage possibilities of radio programs in Flash memory. For robust decoding in error prone channels a new error concealment tool consisting of CRC and intra-frame interleaving was defined.

XI. FIELD TESTS AND SYSTEM PERFORMANCE

The field tests have been scheduled in several phases. While the first phase was focused on verifying that the used channel models were realistic and representative for typical broadcast channels phase II was looking at the system performance on the defined modes. This phase also included tests in Ecuador with near vertical incidence for sky wave (NVIS) propagation. NVIS channels fall into the category of the most critical channels as they suffer from time to time from both high delay and high Doppler spread and the propagation conditions vary a lot during 24h. Delay spreads reached typically 5 ms and sometimes even 7 ms. The Doppler spread was often larger than 2 Hz. Furthermore some long distance transmissions over 6500 km were made showing a satisfactorily performance even for the least robust mode. Phase III is currently looking on the long term performance of typical broadcast links for proving that the quality of service is sufficiently stable during the HF planning seasons of 6 months. The tests have been made with first prototype equipment. The transmitted sequence of signals covered various signals for channel sounding, digital modulation and AM modulated signals for reference purposes and lasted 30 minutes. A measured link was always tested at least with one frequency at the same period of the day with repetitions over a whole week. The results are generally looking very promising so far although some difficult links need optimising with SFNs or alternative frequencies. During an exhibition in August 2001 in Berlin live demonstrations on medium and short wave even with mobile reception could be succesfully shown.

Fig. 16 shows the received spectrum of a digital signal on short wave with some adjacent analogue signals. Due to the sharp drop off at the edges of the digital spectrum the channel bandwidth can be used much more efficiently.

During the system development a lot of simulations and labratory test were performed to optimize the system for different channel characteristics. Fig. 17 gives an overview about the data rates for one service in the MSC in dependency of the SNR to achieve a BER less than 10^{-4}. The curves represent the two robustness modes A and B in the channels 1 and 5. The simulations were done for a channel bandwidth of 10 kHz with different transmssion parameteres for the code rate and the constellation (see Section IV).

XII. SUMMARY AND OUTLOOK

Significantly improved audio quality in existing channel bandwidth, whereas smaller and larger bandwidths are also

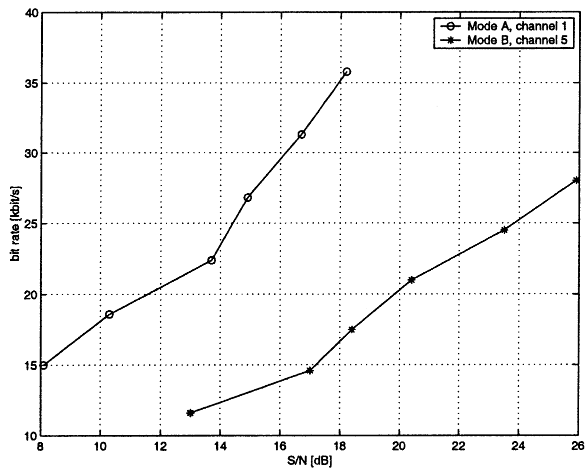

Fig. 17. Data rate for $\mathrm{BER} = 10^{-4}$ dependent on channel characteristics and transmission parameters.

supported, combined with the possibility of huge coverage areas with few transmitters, a new chapter of national and international broadcasting will start. In this article an introduction to the DRM system is given. This article contributes to the understanding of the algorithms used in receivers. First prototype receivers are available and live demonstrations can be admired at several consumer shows.

Acknowledgment

The authors would like to acknowledge the suggestions of Prof. Kuchenbecker from University of Hannover, their colleagues at Robert Bosch GmbH and their colleagues within the DRM consortium.

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