Radiocommunication Study Groups

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Working Party 6A
(Sub-Working Party 6A-1-Sound-DRM)

PRELIMINARY DRAFT REVISION TO RECOMMENDATION ITU-R BS.1114-6

Systems for terrestrial digital sound broadcasting to vehicular, portable

Scope

This Recommendation describes several systems for terrestrial digital sound broadcasting to vehicular, portable and fixed receivers in the frequency range 30-3 000 MHz.The main features of each system such as source coding, channel coding, modulation, transmission structure and threshold levels to achieve good quality of service are described:.

The ITU Radiocommunication Assembly,

considering

a) that there is an increasing interest worldwide for terrestrial digital sound broadcasting (DSB) to vehicular, portable and fixed receivers in the frequency range 30-3 000 MHz for local, regional and national coverage;

b) that the ITU-R has already adopted Recommendations ITU-R BS.774 and ITU-R BO.789 to indicate the necessary requirements for DSB systems to vehicular, portable and fixed receivers for terrestrial and satellite delivery, respectively;

c) that Recommendations ITU-R BS.774 and ITU-R BO.789 recognize the benefits of complementary use of terrestrial and satellite systems, and call for a DSB system allowing for a common receiver with common processing very large scale integration (VLSI) circuits and manufacturing of low-cost receivers through mass production;

d) that Digital System A described in Annex 2 meets all the requirements of Recommendations ITU-R BS.774 and ITU-R BO.789, and that the system has been field-tested and demonstrated in various frequency bands between 200 MHz and 1 500 MHz in a number of countries;

e) that Digital System F described in Annex 3 meets the requirements of Recommendation ITU-R BS.774, and that the system has been field-tested and demonstrated in the 188-192 MHz and 2 535-2 655 MHz bands in more than one country;

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f) that Digital System C described in Annex 4 meets the requirements of Recommendation ITU-R BS.774, and that the system has been field-tested and demonstrated in the 88-108 MHz band;

g) that Digital System "G" described in Annex 5 meets the requirements of Recommendation ITU-R BS.774, and that the system has been field-tested and demonstrated in the 47-68 and 87.5-108 MHz band;

h) that at the 7th World Conference of Broadcasting Unions (México, 27-30 April 1992), the World Broadcasting Unions unanimously resolved:

"1 that efforts should be made to agree on a unique worldwide standard for DAB and

2 to urge administrations to give consideration to the benefits for the consumer of common source and channel coding and implementation of Digital Sound Broadcasting on a worldwide basis at 1.5 GHz;"

j) that the World Administrative Radio Conference for Dealing with Frequency Allocations in Certain Parts of the Spectrum (Málaga-Torremolinos, 1992) (WARC-92) has allocated the band 1 452-1 492 MHz to the broadcasting-satellite service (BSS) (sound) and complementary terrestrial broadcasting service for the provision of DSB. Also, additional footnote allocations were included for specific countries in the band 2 310-2 360 MHz and in the band 2 535-2 655 MHz Nos. 750B and 757A (currently Nos. 5.393 and 5.418) respectively of the Radio Regulations (RR). In addition, Resolution 527 (WARC-92) addresses the subject of terrestrial VHF;

k) that the MPEG-2 transport stream (MPEG-2 TS) is widely applied as containers of digitally coded information;

l) that a standardization process in Europe has resulted in the adoption of Digital System A (Eureka 147 as an ETSI Standard ETS 300 401) for BSS (sound) broadcasting sound to vehicular, portable and fixed receivers;

m) that a standardization process in Japan has resulted in the adoption of Digital System F for integrated services digital broadcasting-terrestrial for sound broadcasting (ISDB-T_{SB}) for digital terrestrial sound broadcasting system to vehicular, portable and fixed receivers;

n) that ISDB techniques can be used to implement services exploiting the full advantages of digital broadcasting, and that Recommendation ITU-R BT.1306 includes the ISDB-T system for digital terrestrial television broadcasting;

o) that a standardization process in Europe has resulted in the adoption of Digital System "G" (DRM as an ETSI Standard ES 201 980) for digital terrestrial sound broadcasting system to vehicular, portable and fixed receivers,

noting

a) that a summary of digital systems is presented in Annex 1;

b) that the condensed system descriptions for Digital Systems A, F, C and "G" are given in Annexes 2, 3, 4 and 5, respectively;

c) that complete system descriptions of Digital Systems A, F, C and "G" are contained in the Digital Sound Broadcasting Handbook,

recommends

1 that Digital Systems A, F, C and/or "G", as described in Annexes 2, 3, 4 and 5, respectively, be used for terrestrial DSB services to vehicular, portable and fixed receivers in the frequency range 30-3 000 MHz;

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2 that administrations that wish to implement terrestrial DSB services meeting some or all of the requirements as stated in Recommendation ITU-R BS.774, should use Table 1 to evaluate the respective merits of Digital Systems A, F, C and "G" in selecting systems.

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TABLE 1

Performance of Digital Systems A, F, C and "G" evaluated on the basis of the recommended technical and operating characteristics listed in Recommendation ITU-R BS.774

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Characteristics from Recommendation ITU-R BS.774 (condensed wording)	Digital System A	Digital System F	Digital System C	Digital System "G"
Range of audio quality and types of reception	Range is from 8 to 384 kbit/s per audio channel in increments of 8 kbit/s. MPEG-2 Layer II audio decoder typically operating at 192 kbit/s is implemented in receivers The system is intended for vehicular, portable and fixed reception	Range is from phone quality to CD quality. It is also capable of 5.1 multi-channel audio. MPEG-2 advanced audio coding (AAC) decoder typically operates at 144 kbit/s for stereo The system is intended for vehicular, portable and fixed reception	Range is from 48 kbit/s to 96 kbit/s using the HD Codec[(1)]decoder The system is intended for vehicular[(2)], portable and fixed reception	Range is from 8 kbit/s to 180 kbit/s using the MPEG-4 HE AAC v2 audio decoder The system is intended for vehicular[(2)], portable and fixed reception
Spectrum efficiency better than FM	FM stereo quality achievable in less than 200 kHz bandwidth; co-channel and adjacent channel protection requirements much less than those for FM. Efficiency is especially high in the case of repeaters reusing the same frequency. (Orthogonal multi-carrier modulation with convolution error correcting coding, coded orthogonal frequency division multiplex (COFDM))	FM stereo quality achievable in less than 200 kHz bandwidth; co-channel and adjacent channel protection requirements much less than those for FM. Efficiency is especially high in the case of repeaters reusing the same frequency. It can be more effective by using 16/64-quadrature amplitude modulation (QAM) carrier modulation. (Orthogonal frequency division multiplex (OFDM) with concatenated block and convolutional error correcting coding)	FM stereo quality and data achievable without additional spectrum; co-channel and adjacent channel protection requirements much less than those for FM. System is interleaved to mitigate first adjacent channel issues and is more robust in the presence of co-channel analogue digital interference	FM stereo quality and data achievable within 100 kHz bandwidth; co-channel and adjacent channel protection requirements much less than those for FM. Efficiency is especially high in the case of repeaters reusing the same frequency. It can be more effective by using 16-quadrature amplitude modulation (QAM) carrier modulation. (Orthogonal frequency division multiplex (OFDM) with multilevel error correcting coding)

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TABLE 1 (*continued*)

Characteristics from Recommendation ITU-R BS.774 (condensed wording)	**Digital System A**	**Digital System F**	**Digital System C**	**Digital System "G"**
Performance in multipath and shadowing environments	System is especially designed for multipath operation. It works on the basis of a power summation of echoes falling within a given time interval. This feature allows use of on-channel repeaters to cover terrain shadowed areas	System is especially designed for multipath environment. It works on the basis of a power summation of echoes falling within a given time interval. This feature allows the use of on-channel repeaters to cover terrain shadowed areas	System is especially designed for multipath operation. It is OFDM modulated thereby achieving a high degree of performance in multipath. This feature allows the use of on-channel repeaters to cover terrain shadowed areas	System is especially designed for multipath environment. It works on the basis of a power summation of echoes falling within a given time interval. This feature allows the use of on-channel repeaters to cover terrain shadowed areas
Common receiver signal processing for satellite (S) and terrestrial (T) broadcasting	Not applicable. Terrestrial only	Not applicable. Terrestrial only	Not applicable. Terrestrial only	Not applicable. Terrestrial only

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TABLE 1 (*continued*)

Characteristics from Recommendation ITU-R BS.774 (condensed wording)	**Digital System A**	**Digital System F**	**Digital System C**	**Digital System "G"**
Reconfiguration and quality vs. number of programmes tradeoff	Service multiplex is based on 64 sub-channels of capacity varying from 8 kbit/s to about 1 Mbit/s, depending on the error protection level, and is totally reconfigurable in a dynamic fashion. Each sub-channel can also contain an unlimited number of variable capacity data packet channels	Multiplexing of payload data is based on MPEG-2 systems. Audio data rate can be selected in any step in order to trade off programme audio quality against the number of services. Transmission parameters such as modulation and error correction are dynamically reconfigurable by transmission and multiplexing configuration control (TMCC)	Bits can be dynamically re-allocated to audio or data using the HDC transport functionalities at the discretion of the broadcaster within the range of 48 to 96 kbit/s for audio to increase or decrease the data rate. The receiver dynamically re-configures to match the transmission mode of operation	Service multiplex is based on up to 4 streams of capacity varying according to broadcaster need, and is totally reconfigurable in a dynamic fashion. Each stream may carry audio or data content with the packet size configurable by the broadcaster to maximise efficiency. The receiver dynamically re-configures to match the transmission mode of operation

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Extent of coverage vs. number of programme trade-offs	Five levels of protection for audio and eight levels of protection for data services are available through using punctured convolutional coding for each of the 64 sub-channels (forward error correction (FEC) ranges from 1/4 to 3/4)	Four kinds of modulation and five levels of protection are available. (Carrier modulation: differential quaternary phase shift keying (DQPSK), QPSK, 16-QAM, 64-QAM, coding rate: 1/2, 2/3, 3/4, 5/6, 7/8)	The system maintains uniform coverage for all programs. Secondary carriers may have reduced range in presence of adjacent channel interference. (Carrier modulation: QPSK)	Two levels of protection may be chosen from a possible eight levels for the multiplex and each stream may be dynamically configured (carrier modulation: 4-QAM, 16-QAM, forward error correction (FEC) ranges from 1/4 to 5/8)

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TABLE 1 (*continued*)

Characteristics from Recommendation ITU-R BS.774 (condensed wording)	Digital System A	Digital System F	Digital System C	Digital System "G"
Common receiver for different means of programme delivery – Terrestrial services	Allows local, subnational and national terrestrial services with the same modulation with single transmitter or multiple transmitters operating in a single frequency network to take advantage of a common receiver	Allows local, subnational and national terrestrial services with the same modulation with a single transmitter or multiple transmitters operating in a single frequency network to take advantage of a common receiver	System uses common antenna and front end that is compatible with existing analogue FM broadcast services. Allows for local service as well as subnational and national terrestrial services with a single transmitter or multiple transmitters operating in a single frequency network in the case of the digital portion of the hybrid mode or the all digital mode. Allows for common delivery of FM programming that makes a seamless transition from digital to analogue and back	Allows local, subnational and national terrestrial services with the same modulation with a single transmitter or multiple transmitters operating in a single frequency network to take advantage of a common receiver

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– Mixed/hybrid	Allows the use of the same band as terrestrial sound broadcasting (mixed) as well as the use of terrestrial on-channel repeaters to reinforce the satellite coverage (hybrid) resulting in all these channels being received transparently by a common receiver.	Allows the use of the same band as terrestrial sound broadcasting (mixed) as well as the use of terrestrial on-channel repeaters to reinforce the satellite coverage (hybrid) resulting in all these channels being received transparently by a common receiver.		
– Cable distribution	Signal can be carried transparently by cable	Signal can be carried transparently by cable	Signal can be carried transparently by cable	Signal can be carried transparently by cable

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TABLE 1 (*continued*)

Characteristics from Recommendation ITU-R BS.774 (condensed wording)	Digital System A	Digital System F	Digital System C	Digital System "G"
Programme-associated data (PAD) capability	PAD channel from 0.66 kbit/s to 64 kbit/s capacity is available through a reduction of any audio channel by the corresponding amount. Dynamic label for programme and service identification showing only receiver alphanumeric display is available to all receivers. Basic hypertext markup language (HTML) decoding and Joint Photographic Experts Group (JPEG) picture decoding is available on receivers with graphic displays (1/4 video graphic array (VGA)), etc.	PAD multiplexing is based on MPEG-2 systems	PAD is an integral part of the system and can be provided through opportunistic data without any reduction of audio quality or data channels. Dynamic label for programme and service identification showing on any receiver alphanumeric display is available to all receivers	PAD with broadcaster selected capacity is available. Dynamic label for programme and service identification showing on any receiver alphanumeric display is available to all receivers. Electronic Progamme Guide and travel data is available; images may be displayed on receivers with suitable graphic displays.

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TABLE 1 (*continued*)

Characteristics from Recommendation ITU-R BS.774 (condensed wording)	**Digital System A**	**Digital System F**	**Digital System C**	**Digital System "G"**
Flexible assignment of services	The multiplex can be dynamically re-configured in a fashion transparent to the user	The multiplex can be dynamically re-configured in a fashion transparent to the user	The system automatically reconfigures between audio and data in a fashion transparent to user	The multiplex can be dynamically re-configured in a fashion transparent to the user
Compatibility of multiplex structure with open system interconnection (OSI)	The system multiplex structure is compliant with the OSI layered model, especially for the data channels, except for the unequal error protection features of the MPEG-2 Layer II audio channel	The system multiplex structure is fully compliant with MPEG-2 systems architecture	The system is based on an OSI layered model including both data and audio except for the unique error protection afforded the audio codec	The system multiplex structure is compliant with the OSI layered model for all services.

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Value-added data capability	Any sub-channel (out of 64) not used for audio can be used for programme-independent data services. Data packet channels for high priority services available to all receivers tuned to any service of the multiplex can be carried in the fast information channel (FIC). Total capacity is up to 16 kbit/s. Receivers are equipped with a radio data interface (RDI) for data transfer to a computer	Capacity at any rate up to the full payload capacity can be assigned to independent data for the delivery of business data, paging, still pictures graphics, etc. under conditional access control if desired	Capacity at any rate up to the full payload capacity can be assigned to independent data for the delivery of business data, paging still pictures graphics, etc. under conditional access control if desired	Capacity at any rate up to the full payload capacity can be assigned to independent data for the delivery of business data, paging still pictures graphics, etc. under conditional access control if desired

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TABLE 1 (*end*)

Characteristics from Recommendation ITU-R BS.774 (condensed wording)	**Digital System A**	**Digital System F**	**Digital System C**	**Digital System "G"**
Receiver low-cost manufacturing	Allows for mass-production manufacturing and low-cost consumer receivers. Typical receivers have been integrated in two chips. One chip manufacturer has integrated the full receiver circuitry into one chip	The system was specifically optimized to enable an initial low complexity vehicular receiver deployment. Standardization group has been established to achieve low cost receivers based on large scale integration (LSI) mass production techniques	The system was specifically optimized to enable an initial low complexity vehicular receiver deployment	Allows for mass-production manufacturing and low-cost consumer receivers.

(1) Additional information about the HD Codec (HDC) can be found at www.ibiquity.com.

(2) The modes implemented in the in-band on-channel (IBOC) chipset (Digital System C) do not support vehicular operation at frequencies above 230 MHz.

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Annex 1

Summaries of Digital Systems

1 Summary of Digital System A

Digital System A, also known as the Eureka 147 digital audio broadcasting (DAB) system, has been developed for both satellite and terrestrial broadcasting applications in order to allow a common low-cost receiver to be used. The system has been designed to provide vehicular, portable and fixed reception with low gain omni-directional receive antennas located at 1.5 m above ground. Digital System A allows for complementary use of satellite and terrestrial broadcast transmitters resulting in better spectrum efficiency and higher service availability in all receiving situations. It especially offers improved performance in multipath and shadowing environments which are typical of urban reception conditions by the use of on-channel terrestrial repeaters to serve as gap-fillers. Digital System A is capable of offering various levels of sound quality up to high quality sound comparable to that obtained from consumer digital recorded media. It can also offer various data services and different levels of conditional access and the capability of dynamically re-arranging the various services contained in the multiplex.

2 Summary of Digital System F

Digital System F, also known as the ISDB-TSB system, is designed to provide high-quality sound and data broadcasting with high reliability even in mobile reception. The system is also designed to provide flexibility, expandability, and commonality for multimedia broadcasting using terrestrial networks. The system is a rugged system which uses OFDM modulation, two-dimensional frequency-time interleaving and concatenated error correction codes. The OFDM modulation used in the system is called band segmented transmission (BST)-OFDM. The system has commonality with the ISDB-T system for digital terrestrial television broadcasting in the physical layer. The system has a wide variety of transmission parameters such as carrier modulation scheme, coding rates of the inner error correction code, and length of time interleaving. Some of the carriers are assigned to TMCC carriers which transmit the information on the transmission parameters for receiver control. Digital System F can use high compression audio coding methods such as MPEG-2 AAC. And also, the system adopts MPEG-2 systems. It has commonality and interoperability with many other systems which adopt MPEG-2 systems such as ISDB-S, ISDB-T, DVB-S and DVB-T.

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3 Summary of Digital System C

Digital System C, also known as the IBOC DSB system, is a fully developed system. The system was designed to provide vehicular[1], portable and fixed reception using terrestrial transmitters. Although Digital System C can be implemented in unoccupied spectrum, a significant feature of the system is its ability to offer simul-casting of analogue and digital signals in the existing FM broadcasting band. This system feature would allow for a rational transition for existing FM broadcasters seeking to transition from analogue to digital broadcasting. The system offers improved performance in multipath environments resulting in greater reliability than is offered by existing analogue FM operations. Digital System C offers enhanced audio quality comparable to that obtained from consumer digital recorded media. Moreover, the system incorporates flexibility for broadcasters to offer new data-casting services in addition to the enhanced audio programming. In addition, the system allows for allocation of bits between audio and data-casting capacity to maximize the data-casting capabilities.

4 Summary of Digital System "G"

Digital System "G", also known as the Digital Radio Mondiale (DRM) system, has been developed for terrestrial broadcasting applications in all the frequency bands allocated worldwide for analogue sound broadcasting. It respects the ITU defined spectrum masks allowing a smooth transition from analogue to digital broadcasting. In the bands above 30 MHz, it defines Robustness Mode E (also known as DRM+) to offer audio quality comparable to that obtained from consumer digital recorded media. In addition, Digital System "G" also offers various data services, including images and electronic programme guides, and the capability of dynamically re-arranging the various services contained in the multiplex without loss of audio.

[Secretarial Note: There is no change proposed to Annexes 2, 3 and 4.]

1 The modes implemented in the IBOC chipset (Digital System C) do not support vehicular operation at frequencies above 230 MHz.

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Annex 5

Digital System "G"

1 Introduction

Digital system "G", also known as the DRM system, is designed to be used at any frequency below 174 MHz, with variable channelization constraints and propagation conditions throughout these bands. In order to satisfy these operating constraints, different transmission modes are available. A transmission mode is defined by transmission parameters classified in two types:

- signal bandwidth related parameters;
- transmission efficiency related parameters.

The first type of parameters defines the total amount of frequency bandwidth for one transmission. Efficiency related parameters allow a trade-off between capacity (useful bit rate) and ruggedness to noise, multipath and Doppler.

Digital system "G" is standardised at ETSI as ES 201 980V3.1.1 (2009.08) "Digital Radio Mondiale (DRM); System specification".

Digital system "G" has a number of robustness modes, each designed for different bands and propagation conditions, as illustrated in Table 1.

TABLE 1

Robustness mode uses

Robustness mode	Typical propagation conditions
A	Gaussian channels, with minor fading
B	Time and frequency selective channels, with longer delay spread
C	As robustness mode B, but with higher Doppler spread
D	As robustness mode B, but with severe delay and Doppler spread
E	Time and frequency selective channels

DRM30 consists of robustness modes A to D and is designed for the long, medium and short wavebands and is the subject of Rec. ITU-R 1514.

DRM+ consists of robustness mode E and is designed for the low VHF bands and is the subject of this recommendation as digital system "G".

2 System architecture

Figure 1 describes the general flow of different classes of information (audio, data, etc.) and does not differentiate between different services that may be conveyed within one or more classes of information.

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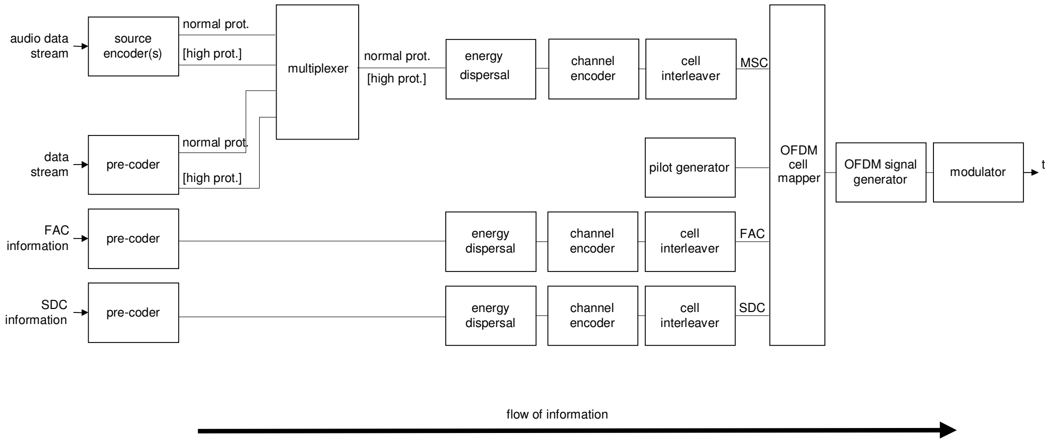
FIGURE 1



The figure describes the general flow of the different classes of information (audio, data, …) from encoding on the left to a transmitter on the right. Although a receiver diagram is not included it would represent the inverse of the process shown in this diagram.

- On the left are two classes of input information: the encoded audio and data that are combined in the main service multiplexer, and the information channels that bypass the multiplexer that are known as the FAC and SDC.
- The audio source encoder and the data pre-coders ensure the adaptation of the input streams onto an appropriate digital format. Their output may comprise two parts requiring two different levels of protection within the subsequent channel encoder.
- The multiplexer combines the protection levels of all data and audio services.
- The energy dispersal provides a deterministic, selective complementing of bits in order to reduce the possibility that systematic patterns result in unwanted regularity in the transmitted signal.
- The channel encoder adds redundant information as a means for error correction and defines the mapping of the digitally encoded information into QAM cells. The system has the capability, if a broadcaster desires, to convey two categories of "bits", with one category more heavily protected than the other.
- Cell interleaving spreads consecutive QAM cells onto a sequence of cells, quasi-randomly separated in time and frequency, in order to provide an additional element of robustness in the transmission of the audio in time-frequency dispersive channels.
- The pilot generator injects information that permits a receiver to derive channel-equalisation information, thereby allowing for coherent demodulation of the signal.
- The OFDM cell mapper collects the different classes of cells and places them on a time-frequency grid.
- The OFDM signal generator transforms each ensemble of cells with the same time index to a time domain representation of the signal, containing a plurality of carriers. The complete

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time-domain OFDM symbol is then obtained from this time domain representation by inserting a guard interval — a cyclic repetition of a portion of the signal.

- The modulator converts the digital representation of the OFDM signal into the analogue signal that will be transmitted via a transmitter/antenna over the air. This operation involves frequency up-conversion, digital-to-analogue conversion, and filtering so that the emitted signal complies with ITU-R spectral requirements.

3 Audio coding, text messages and packet data

3.1 Audio

Within the constraints of broadcasting regulations in broadcasting channels between 30 MHz and 174 MHz and the parameters of the coding and modulation scheme applied, the bit rate available for audio coding is in the range from 35 kbit/s to 185 kbit/s.

In order to offer optimum quality at a given bit rate, the system offers different audio coding schemes:

- A subset of MPEG-4 AAC (Advanced Audio Coding) including error robustness tools for generic mono and stereo audio broadcasting.
- Spectral Band Replication (SBR), an audio coding enhancement tool that allows the full audio bandwidth to be achieved at low bit rates.
- Parametric Stereo (PS), an audio coding enhancement tool relevant to SBR that allows for stereo coding at low bit rates.
- MPEG Surround (MPS), an audio coding enhancement tool that allows for multichannel coding at low bit rates.

AAC is highly optimized in terms of coding efficiency and according to information theory this should lead to the fact that the entropy of the bits is nearly equal. If this assumption is true, then the channel coding must be optimized such that the total amount of residual errors usually referred to as Bit Error Rate (BER) is minimized. This criterion can be fulfilled by a channel coding method called Equal Error Protection (EEP), were all information bits are protected with the same amount of redundancy.

However, the audible effects of errors are not independent of the part of the bitstream that was hit by the error. The optimized solution to cope with this unequal error sensitivity is called Unequal Error Protection (UEP). In such a system, higher protection is assigned to the more sensitive information, whereas lower protection is assigned to the less sensitive part of the bitstream.

To accommodate UEP channel coding, it is necessary to have frames with a constant length and a UEP profile that is constant as well for a given bit rate. Since AAC is a coding scheme with a variable length, digital system "G" groups several coded frames together to build one audio super frame. The bit rate of the audio super frame is constant. Since the channel coding is based on audio super frames, the audio super frames themselves consist of two parts: a higher protected part and a lower protected part. Therefore, the coded audio frames itself have to be split into these two parts.

The bit-stream transport format of MPEG AAC has been modified to meet the requirements of digital system "G" (audio superframing). Unequal Error Protection (UEP) can be applied to improve the system behaviour in error prone channels.

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3.2 Text message application

Text messages can provide a highly valuable additional element to an audio service without consuming much data capacity. The text message is a basic part of digital system "G" and consumes only 320 bits/s. This capacity can be saved if the service provider does not use text messaging.

3.3 Packet data mode

Data services generally consist of either streams of information, in either synchronous or asynchronous form, or files of information. Digital system "G" provides a generalized packet delivery system which allows the delivery of asynchronous streams and files for various services in the same data stream and allows the bit rate of the (synchronous) data stream to be shared on a frame-by-frame basis between the various services. The data stream may be provided with additional error control by the addition of forward error correction. Services can be carried by a series of single packets or as a series of data units. A data unit is a series of packets that are considered as one entity with regard to error handling - one received errored packet within a data unit causes the whole data unit to be rejected. This mechanism can be used to transfer files and also to allow simpler synchronization of asynchronous streams. The packet data mode of digital system "G" is configuarble by the broadcaster to allow optimised use of any capacity: both the packet length and strength of the Forward Error Protection may be varied and signalled to receivers.

4 Multiplex, including special channels

Receivers must be easy to use. Digital system "G" provides signalling data to allow the listener to access the service he wants with a simple button press, and to allow the radio to track the broadcast to find the best frequency at all times and so leave the listener free to enjoy the programme.

DRM uses a combination of techniques to provide user friendliness. First, the total data capacity is divided into a multiplex of three sub-channels:

- the Fast Access Channel (FAC);
- the Service Description Channel (SDC);
- the Main Service Channel (MSC).

The FAC contains useful information to allow the receiver to find services of interest to the listener quickly. For example, the receiver can scan the bands looking for services with a particular Programme Type or in a particular language. It also contains information about the broadcast mode to allow further decoding of the signal.

The SDC contains further information about the service (or multiplex of services - up to four) to enhance user friendliness. This includes a label of up to 16 characters (the UTF-8 coding standard is used so all characters are available, not just Latin based ones) and how to find alternative sources of the same data, and gives attributes to the services within the multiplex. The size of the SDC varies according to the mode.

Alternative frequency checking may be achieved, without loss of service, by keeping the data carried in the SDC quasi-static. Therefore, the data in the SDC frames has to be carefully managed.

The MSC contains the audio and/or data services. The overall frame structure is designed to allow a receiver to jump to an alternate frequency and back without losing any data from the MSC. This means that when a number of frequencies are needed to provide the service, the receiver can always be checking for the best frequency and re-tune when necessary without any interruption to the audio. The SDC provides the list of frequencies and can also give a frequency schedule to allow for services that need different frequencies during the day and week.

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By using these features, receivers can present services in a friendly way to the listener, who no longer has to be dependent on knowing the frequency or frequency schedule, and gets a positive confirmation from the displayed label that he is tuned to the service he wants.

The Main Service Channel (MSC) contains the data for all the services contained in the multiplex. The multiplex may contain between one and four services, and each service may be either audio or data. The gross bit rate of the MSC is dependent upon the selected transmission parameters.

The MSC contains between one and four streams. Each stream is divided into logical frames. Audio streams comprise compressed audio and optionally they can carry text messages. Data streams may be composed of data packets, carrying information for up to four "sub-streams". An audio service comprises one audio stream and optionally one data stream or one data sub-stream. A data service comprises one data stream or one data sub-stream.

Each logical frame generally consists of two parts, each with its own protection level. The lengths of the two parts are independently assigned. Unequal error protection for a stream is provided by setting different protection levels to the two parts.

The logical frames are each 100 ms long. If the stream carries audio, the logical frame carries the data for either the first or the second part of one audio super frame containing the audio information for 200 ms duration. Since, in general, the stream may be assigned two protection levels, the logical frames carry precisely half of the bytes from each protection level.

The logical frames from all the streams are mapped together to form multiplex frames of the same duration, which are passed to the channel coder.

The multiplex configuration is signalled using the SDC. The multiplex may be reconfigured at transmission super frame boundaries. A reconfiguration of the multiplex occurs when the channel parameters in the FAC are changed, or when the services in the multiplex are reorganized. The new configuration is signalled ahead of time in the SDC and the timing is indicated by the reconfiguration index in the FAC.

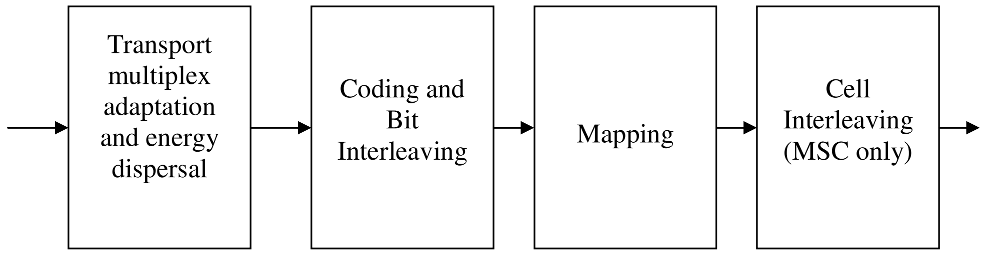
5 Channel coding and modulation

5.1 Introduction

Because of the different needs of the three sub-channels, the MSC, SDC and FAC, these sub-channels apply different coding and mapping schemes. An overview of the encoding process is shown in Figure 2.

FIGURE 2

Functional block diagram of the coding and interleaving



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The coding is based on a multilevel coding scheme. Due to different error protection needs within one service or for different services within one multiplex different mapping schemes and combinations of code rates are applicable: Unequal Error Protection (UEP) and Equal Error Protection (EEP) are available. Equal error protection uses a single code rate to protect all the data in a channel. EEP is mandatory for the FAC and SDC. Instead of EEP, unequal error protection can be used with two code rates to allow the data in the Main Service Channel to be assigned to the higher protected part and the lower protected part.

5.2 Multilevel coding

The channel encoding process is based on a multilevel coding scheme. The principle of multilevel coding is the joint optimization of coding and modulation to reach the best transmission performance. This denotes that more error prone bit positions in the QAM mapping get a higher protection. The different levels of protection are reached with different component codes which are realized with punctured convolutional codes, derived from the same mother code.

The decoding in the receiver can be done either straightforwardly or through an iterative process. Consequently the performance of the decoder with errored data can be increased with the number of iterations and hence is dependent on the decoder implementation.

5.3 Coding the MSC

The MSC may use either 4-QAM or 16-QAM mapping: the lower constellation provides a more robust error performance whereas the higher constellation provides high spectral efficiency.

In each case, a range of code rates is available to provide the most appropriate level of error correction for a given transmission. The available combinations of constellation and code rate provide a large degree of flexibility over a wide range of transmission channels. Unequal error protection can be used to provide two levels of protection for the MSC.

Two protection levels within one multiplex frame are possible resulting in the use of two overall code rates. The overall code rates and code rates for each level are defined in Tables 2 and 3. The protection level is signalled in the multiplex description data entity of the SDC.

TABLE 2

Code rates for the MSC with 4-QAM

Protection level	**R_{all}**	**R_0**
0	0,25	1/4
1	0,33	1/3
2	0,4	2/5
3	0,5	1/2

TABLE 3

Code rate combinations for the MSC with 16-QAM

Protection level	**R_{all}**	**R_0**	**R_1**	**RY_{lcm}**
0	0,33	1/6	1/2	6
1	0,41	1/4	4/7	28
2	0,5	1/3	2/3	3

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3	0,62	1/2	3/4	4

One or two overall code rates shall be applied to one multiplex frame. When using two overall code rates, both shall belong to the same constellation.

5.4 Coding the SDC

The SDC uses 4-QAM mapping with code rate 0,5 or 0,25: a choice is available between greater capacity and a more robust error performance.

The constellation and code rate should be chosen with respect to the MSC parameters to provide more robustness for the SDC than for the MSC.

5.5 Coding the FAC

The FAC shall use 4-QAM mapping with code rate 0,25.

6 Transmission structure

The transmitted signal is organized in transmission super frames which consist of four transmission frames.

Each transmission frame has duration T_f, and consists of N_s OFDM symbols.

Each OFDM symbol is constituted by a set of K carriers and transmitted with a duration T_s.

The spacing between adjacent carriers is $1/T_u$.

The symbol duration is the sum of two parts:

- a useful part with duration T_u;
- a guard interval with duration T_g.

The guard interval consists in a cyclic continuation of the useful part, T_u, and is inserted before it.

The OFDM symbols in a transmission frame are numbered from 0 to N_s - 1.

All symbols contain data and reference information.

Since the OFDM signal comprises many separately modulated carriers, each symbol can in turn be considered to be divided into cells, each cell corresponding to the modulation carried on one carrier during one symbol.

An OFDM frame contains:

- pilot cells;
- control cells;
- data cells.

The pilot cells can be used for frame, frequency and time synchronization, channel estimation, and robustness mode identification.

The transmitted signal is described by the following expression:

$$x(t) = \mathrm{Re}\left\{ e^{j2\pi f_R t} \sum_{r=0}^{\infty} \sum_{s=0}^{N_s-1} \sum_{k=K_{\min}}^{K_{\max}} c_{r,s,k}\, \psi_{r,s,k}(t)_{\,k} \right\}$$

where:

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$$\psi_{r,s,k}(t)=\begin{cases} e^{j2\pi\frac{k}{T_u}(t-T_g-sT_s-N_s rT_s)} & (s+N_s\ r)T_s\le t\le(s+N_s\ r+1)T_s \\ 0 & otherwise \end{cases}$$

and:

N_s number of OFDM symbols per transmission frame

k denotes the carrier number (= K_{min}, … , K_{max})

s denotes the OFDM symbol number (= 0 to N_s - 1)

r denotes the transmission frame number (= 0 to infinity)

K is the number of transmitted carriers (≤ K_{max} - K_{min})

T_s is the symbol duration

T_u is the duration of the useful part of a symbol

T_g is the duration of the guard interval

f_R is the reference frequency of the RF signal

$c_{r,s,k}$ complex cell value for carrier k in symbol s of frame number r.

The $c_{r,s,k}$ values depend on the type of cell, as defined below.

For data/control cells (MSC, SDC, FAC), $c_{r,s,k} = z$ where z is the constellation point for each cell as given by the mappings defined in clause 7.

For each reference cell, a defined phase and amplitude is transmitted, , where:

$a_{s,k}$ is the amplitude, which always takes one of the values $\{1,\sqrt{2},2\}$, and

$U_{s,k} = e^{j2\pi\vartheta_{s,k}}$ is a unit-amplitude term of phase $\vartheta_{s,k}$.

7 Transmit diversity

Digital system "G" is designed for various transmission environments with different delay spread and Doppler spread. Multipath environments with short and strong echoes, which typically occur in urban canyons, lead to a huge coherence bandwidth so that the channel can be described as flat instead of frequency-selective. Systems with a bandwidth smaller than the coherence bandwidth can accordingly suffer from flat fading. Time interleaving applied to digital system "G" improves the performance of moving receivers in such circumstances.

A further method to overcome flat fading is antenna diversity, which means the application of more than one antenna at the receiver or transmitter. Antenna diversity at the receiver is effective but difficult to implement in small receiver boxes. For broadcast systems the use of transmit diversity is a good alternative or addition to receive diversity.

In the development of digital system "G", different methods, such as space time coding and delay diversity, were evaluated. This work showed that delay diversity is the preferred choice because space time coding requires more overhead in the signal for channel estimation and it is more sensitive against the time-incoherence for fast fading channels.

The idea of delay diversity is quite simple. In addition to the original signal a delayed version of the same signal is transmitted from another spatially separated antenna. This method increases the channel delay spread by an additional echo with a comparable effect as with single frequency

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networks. The application of transmit delay diversity does not require any modifications at the receiver.

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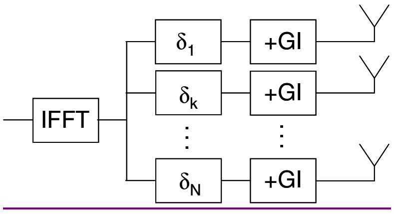
Figure 3 shows how delay diversity can be implemented at the transmitter for an arbitrary number of antennas. After the OFDM modulation with an IFFT the signal path is split according to the number of antennas. Each signal path will be delayed by a chosen value δ_k before insertion of the guard interval.

FIGURE 3

Transmit delay diversity scheme



The only parameters which have to be chosen are the delay δ_k of each path. Two requirements have to be considered:

- the delay δ_k should be large enough to increase the frequency selectivity of the composed channel that is the superposition of the channels for the transmit antennas;
- and it should be much less than the guard interval duration T_g in order to avoid inter-symbol-interference.

According to the first requirement the value δ should be at least 10 µs for a two antenna system in digital system "G". This value also fulfils the second requirement because only 4 % of the guard interval duration gets lost. For further optimisation the appropriate scientific literature is recommended.

The improvements which can be obtained with transmit delay diversity depend on the actual transmission channel. Simulations show that the SNR gain at a BER of 10^{-4} for the Terrain Obstructed profile at 60 km/h receiver speed is around 1 dB, for the Typical Urban profile at 60 km/h around 2 dB and the Typical Urban profile at 5 km/h more than 4 dB.

8 Combined transmission of digital and analogue signals

A close placement of a digital system "G" signal to an analogue FM signal is possible and can be flexibly configured depending on the existing use of spectrum. In this way, digital system "G" may be introduced into the FM frequency bands.

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FIGURE 4

Example configuration for digital system "G" (DRM+, left) and FM signal (right)

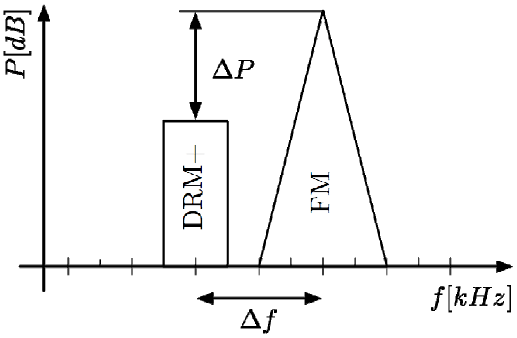

Figure 4 shows that the digital system "G" signal can be placed closely to the left or right of the existing FM signal. To guarantee the respective protection levels and audio quality of the FM signal, the carrier frequency distance (Δf) and the power level difference (ΔP) of the FM and the digital system "G" signals can be planned accordingly. Δf can be chosen according to a 50 kHz channel raster. Δf>=150 kHz is recommended. ΔP can be varied flexibly; however, a ΔP>20 dB is recommended for the minimum Δf=150 kHz.

Two transmission configurations are possible: the analogue and digital signals can be combined and transmitted via the same antenna; or the two signals can be transmitted from different antennas.

Different configurations for the digital system "G" signal are possible. The digital system "G" signal can have the same program as the FM service, a different program or the same program as well as additional programs. If the same program is available via digital system "G" and FM, the alternative frequency switching (AFS) flag should be sent in the service description channel (SDC) of the transmission multiplex allowing for a support of heterogeneous networks.

Figure 5 shows some example configurations.

FIGURE 5

Example configuration with digital system "G" (left) and 2 FM stations (right)

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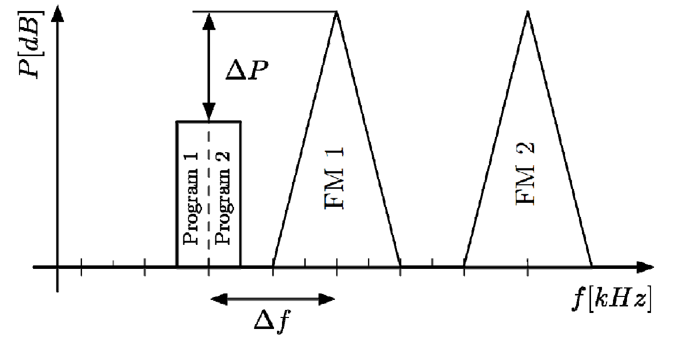

9 Simulated system performance

The radio wave propagation in VHF bands I and II is characterized by diffraction, scattering and reflection of the electromagnetic waves on their way between the transmitter and the receiver. Typically the waves arrive at different times at the receiver (multipath propagation) resulting in more or less strong frequency-selective fading (dependent on system bandwidth). In addition movements of the receiver or surrounding objects cause a time variation of the channel characteristic (Doppler effect). In contrast to sky wave propagation e.g. at short waves, ionospheric variations play no role for channel modelling for bands I and II.

The approach is to use stochastic time-varying models with a stationary statistics and define models for good, moderate and bad conditions by taking appropriate parameter values of the general model. One of those models with adaptable parameters is the Wide Sense Stationary Uncorrelated Scattering model (WSSUS model). The justification for the stationary approach with different parameter sets is that results on real channels lead to BER curves between best and worst cases found in the simulation.

Additional variations of the short-term average power (slow or lognormal fading) caused by changing environment (e.g. building structure) or phenomena like sporadic E layer propagation are not incorporated in the WSSUS model. Their effects, as well as the influence of disturbances like man-made noise, are normally integrated in the computation of the coverage probability during the network planning process.

The channel models have been generated from the following equations where e(t) and s(t) are the complex envelopes of the input and output signals respectively:

$$s(t) = \sum_{k=1}^{n} \rho_k c_k(t) e(t - \Delta_k) \qquad \text{(B.5)}$$

This is a tapped delay-line where:

ρ_k is the attenuation of the path number k - listed in table B.2.

Δ_k is the relative delay of the path number k - listed in table B.2

the time-variant tap weights $\{c_k(t)\}$ are zero mean complex-valued stationary Gaussian random processes. The magnitudes $|c_k(t)|$ are Rayleigh- or Ricean-distributed (dependent on the availability of line-of-sight (LOS) between transmitter and receiver) distributed and the phases Φ (t) are uniformly distributed.

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For each weight $\{c_k(t)\}$ there is one stochastic process, characterized by its variance and its power density spectrum $P_k(f)$. The variance is a measure for the average signal power which is received via this path and is defined by the value of ρ_k. $P_k(f)$ determines the average speed of variation in time, i.e. describes the influence of the Doppler effect on the waves arriving at delay time Δ_k. Therefore $P_k(f)$ is also known as Doppler spectrum.

For the description of the channel models the following definitions for the Doppler spectra are used:

A basic parameter is the maximum Doppler frequency

$$f_d = \frac{v}{\lambda} \tag{B.6}$$

With:

v the velocity of the receiver or surrounding objects; and

λ the wavelength of the transmitted signal.

In case that all waves are arriving from all directions at the receiving antenna with approximately the same power the real Doppler spectrum can be approximated by:

$$P_k(f) = \frac{A}{\sqrt{1-\left(\frac{f}{f_d}\right)^2}} \quad \text{for } f \in \left]-f_d, f_d\right[\tag{B.7}$$

This spectrum is also known as classical Jakes' spectrum and will be denoted as "**Classical**" in the following models.

In the LOS case an additional deterministic component with a distinct Doppler shift has to be added to the Doppler spectrum for the stochastic component. The resultant spectrum denoted as "**Rice**" is defined by the following equation:

$$P_k(f) = \frac{A}{\sqrt{1-\left(\frac{f}{f_d}\right)^2}} + B \times \delta\left(f - f_D\right) \quad \text{for } f \in \left]-f_d, f_d\right[\tag{B.8}$$

with $\delta(f)$ the Dirac pulse and $-f_d \le f_D \le f_d$. For a propagation path with a Rice Doppler spectrum the so-called Rice factor is given by $B/(\pi f_d A)$. It describes the power ratio between the LOS and the stochastic component.

Further spectra are defined with the help of the Gaussian function $G(f, A, f_1, f_2)$:

$$G(f, A, f_1, f_2) = A \exp\left(\frac{(f-f_1)^2}{2f_2^2}\right) \tag{B.9}$$

The spectra denoted by "Gauss1" and "Gauss2" consist of a single Gaussian function and are defined as

$$P_k(f) = G\left(f, A, \pm 0{,}7 \times f_d, 0{,}1 \times f_d\right) \tag{B.10}$$

where the "+" sign is valid for "**Gauss1**" and the "-" sign for "**Gauss2**".

The Gaussian spectra are used in channel profiles for propagation paths with large delay times.

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TABLE 4

Set of channels

Channel no 7: AWGN			
Velocity: 0 km/h (no time variation)			
Path no, *k*	**Delay (µs)**	**Rel. power (dB)**	**Path type**
1	0,0	0,0	Rice (A = 0, B = 1, f_D = 0 Hz)

Channel no 8: Urban			
Velocities: 2 and 60 km/h (pedestrian and vehicle speed)			
Path no, *k*	**Delay (µs)**	**Rel. power (dB)**	**Path type**
1	0,0	-2,0	Classical
2	0,2	0,0	Classical
3	0,5	-3,0	Classical
4	0,9	-4,0	Classical
5	1,2	-2,0	Classical
6	1,4	0,0	Classical
7	2,0	-3,0	Classical
8	2,4	-5,0	Classical
9	3,0	-10,0	Classical

Channel no 9: Rural			
Velocity: 150 km/h (vehicle speed on highways)			
Path no, *k*	**Delay (µs)**	**Rel. power (dB)**	**Path type**
1	0,0	-4,0	Classical
2	0,3	-8,0	Classical
3	0,5	0,0	Classical
4	0,9	-5,0	Classical
5	1,2	-16,0	Classical
6	1,9	-18,0	Classical
7	2,1	-14,0	Classical
8	2,5	-20,0	Classical
9	3,0	-25,0	Classical

Channel no 10: Terrain obstructed			
Velocity: 60 km/h (speed within built-up areas)			
Path no, *k*	**Delay (µs)**	**Rel. power (dB)**	**Path type**
1	0,0	-8,0	Classical
2	1,0	-2,0	Classical
3	2,5	0,0	Classical
4	3,5	-1,0	Classical
5	5,0	-2,0	Classical
6	8,0	-3,0	Classical
7	12,0	0,0	Classical
8	14,0	-6,0	Classical
9	16,0	-3,0	Classical

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Channel no 11: Hilly terrain			
Velocity: 100 km/h (vehicle speed along country roads)			
Path no, *k*	**Delay (µs)**	**Rel. power (dB)**	**Path type**
1	0,0	0,0	Classical
2	0,5	-5,7	Classical
3	1,3	-12,7	Classical
4	1,9	-20,6	Classical
5	30,0	-3,1	Gauss1
6	31,3	-5,4	Gauss1
7	34,9	-11,6	Gauss1
8	37,2	-15,9	Gauss1
9	39,1	-18,9	Gauss1
10	40,0	-25,7	Gauss1
11	80,0	-4,5	Gauss2
12	82,7	-11,5	Gauss2

Channel no 12: SFN			
Velocity: 150 km/h (vehicle speed on highways)			
Path no, *k*	**Delay (µs)**	**Rel. power (dB)**	**Path type**
1	0,0	0,0	Classical
2	100,0	-13,0	Gauss1
3	220,0	-18,0	Gauss2
4	290,0	-22,0	Gauss1
5	385,0	-26,0	Gauss2
6	480,0	-31,0	Gauss1
7	600,0	-32,0	Gauss2

Simulated system performance anticipating perfect channel estimation, ideal synchronization and the absence of phase noise and quantization effects has been performed. The signal power includes pilots and the guard interval. Channel decoding is assumed to be done with single stage Viterbi decoding for 4-QAM modulation and with a multistage decoder with two iterations for 16-QAM modulation.

The results in Table 5 are given for six of the channels of clause B.2, whereby the associated robustness mode is E. The code rate is R=0,33 and the modulation is 4-QAM.

TABLE 5

Required C/N for a transmission to achieve a BER = 1×10^{-4} after the channel decoder for the MSC (Mode E)

Channel model	**C/N**
Channel 7 (AWGN)	1,3 dB
Channel 8 (Urban) at 60 km/h	7,3 dB
Channel 9 (Rural)	5,6 dB
Channel 10 (Terrain obstructed)	5,4 dB
Channel 11 (Hilly terrain)	5,5 dB
Channel 12 (SFN)	5,4 dB

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The results in Table 6 are given for six of the channels of clause B.2, whereby the associated robustness mode is E. The code rate is R=0,5 and the modulation is 16-QAM.

TABLE 6

Required C/N for a transmission to achieve a BER = 1×10^{-4} after the channel decoder for the MSC (Mode E)

Channel model	C/N
Channel 7 (AWGN)	7,9 dB
Channel 8 (Urban) at 60 km/h	15,4 dB
Channel 9 (Rural)	13,1 dB
Channel 10 (Terrain obstructed)	12,6 dB
Channel 11 (Hilly terrain)	12,8 dB
Channel 12 (SFN)	12,3 dB

10 Field trials

Two sites in Germany have been conducting field trials of digital system "G" in band II and demonstrations have been successfully given in Paris in Band I.

In Hannover, digital system "G" has been operating on 95.2 MHz in the FM band. The work involves the University of Hannover, Fraunhofer, Bosch and others and has used experimental equipment developed by the partners. The trials have confirmed the technical parameters and comparisons of coverage area have been performed between FM and digital system "G".

In Kaiserslautern, similar work has taken place led by the University in partnership with many other companies. In addition, protection ratio measurements have been performed and planning models have been used to predict coverage. Further information is available from http://www.fh-kl.de/~drm.

The results from both German sites show that digital system "G" works well. Wherever FM works, digital system, "G" works too and at lower transmit power. Digital system "G" has much greater coverage from the same transmit power; alternatively, the same coverage area can be achieved with a significant power reduction. In the congested European spectrum, however, coverage in the FM band is almost always limited by interference rather than by the sensitivity of the receiver (noise limited). Therefore until the high power FM transmissions are removed, digital system "G" will not be able to operate at the lowest planned power in order to counter interference from high power co-channel FM services. Further studies and coverage trails with fully documented results are planned for 2009/10.

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