International Digital Audio Broadcasting Standards: Voice Coding and Amateur Radio Applications

Here's a digital-voice standard for broadcast and Amateur Radio—read all about it!

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This paper presents a new digital system for use in terrestrial audio broadcasting in the frequency bands below 30 MHz and its potential use for Amateur Radio. The system is based upon a COFDM modem; it was elaborated as a derivation of the DRM modem. DRM (Digital Radio Mondiale) is a worldwide consortium proposing through ITU a new standard for digital radio broadcasting for frequencies below 30 MHz. It is now the only digital standard adopted by ITU for HF digital radio broadcasting. A specially adapted version that fits inside a 3-kHz channel was derived for Amateur Radio. This paper presents the existing demonstrator to be tested on an across-the-Atlantic link, together with the associated modem and voice coding techniques.

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A new coherent COFDM (coded orthogonal frequency-division multiplexed) modem for use in high-quality audio broadcasting in the frequency bands below 30 MHz is proposed by DRM as a candidate to replace the current AM modulation that is notoriously perturbed by multipath, Doppler shift and fading. The main characteristics of this system are to be compatible with existing channelization within these bands and with current transmitter technology (5-kHz spacing). Therefore, it should be a better alternative than satellite broadcasting at higher frequencies because it is cheaper, offers better propagation and indoor coverage. It also opens the door for much better quality by offering net bit rates compatible with up-to-date audio compression techniques like AAC-MPEG 4 (near FM quality).

Such a broadcasting scheme may even be implemented using existing power transmitters in a transmission mode named *Simulcast*. This mode consists in transmitting a program in a given frequency channel by using an AM compatible analogue waveform and the new digital signal simultaneously.

This broadcasting system was initially elaborated within the NADIB consortium (European Eureka 1559 project: *NArrow Band DIgital Broadcasting*) and was adopted and enhanced within the DRM consortium as a candidate for international normalization at ITU. Real transmission on a standard 100-kW, shortwave AM transmitter was demonstrated at the

1997 IBC conference in Amsterdam. Since then, numerous field trials have been performed. The latest series (May/June 2002) involved transmissions from Canada, the Caribbean and Europe to Australia and New Zealand.

Thales, with its two affiliates Thales Communications SA and Thales Broadcast and Multimedia SA, was within the initial group of companies to work on the standard.[1] Going for digital broadcasting allows the introduction of new services such as Program Associated Data (PAD), RDS type alternative frequency switch (AFS) management, single frequency network (SFN), simultaneous data channels for various services such as picture transmission, and so forth. This paper describes work performed by Thales Communications France to adapt this standard and the associated modem to 3-kHz Amateur Radio needs.

DRM Standards

DRM stands for Digital Radio Mondiale, where the spelling of the last word is on purpose to show its international nature. DRM is a worldwide consortium established in Geneva, Switzerland, to promote a unique LF, MF and HF audio-broadcasting standard. See **www.drm.org** for more details. This site contains two reference ITU papers on the subject.[2, 3]

DRM's members include: broadcasters and broadcasting associations; network operators; research institutes; component, receiver and transmitter manufacturers; regulatory and standardization authorities.

There are presently more than 80 members in DRM, corresponding to 27 countries. In the USA for example, members include: Harris Broadcast, the International Broadcasting Bureau, Continental Electronics Corporation, Sangean America Inc, Technology for Communications International. Some other active members include: the British Broadcasting Corporation (BBC), Sony, Bosch, Thales, RFI, DW, JVC, Telefunken and so on.

The standardization process involves mainly the ITU (International Telecommunication Union) based in Geneva, the European body ETSI (European Telecommunications Standard Institute) and the ISO (International Standard Office). Other bodies such as IEC and ARIB are also involved.

Key features of the standard include:

- A worldwide standard to allow for unique replacement of the AM format
- Compatibility with existing channelization: The DRM signal is designed to fit in with the existing AM broadcast band plan, based on signals with 9 or 10-kHz channelization. It has modes requiring as little as 4.5- or 5-kHz bandwidth, plus modes that can take advantage of wider bandwidths, such as 18 or 20 kHz.
- Better audio quality: The aim is to obtain near-FM quality within a much narrower frequency bandwidth. The improvement upon analogue AM is immediately noticeable. DRM can be used for a range of audio content, including multi-lingual speech and music.
- Simple-to-use receivers, especially when it comes to HF programming (frequent frequency changes due to propagation conditions)
- Low-cost equipment to quickly reach the mass market
- Text messaging similar to RDS at FM for simple PAD (Program Associated Data) transmission
- Data applications: an open path for new applications of this medium with a large geographical coverage
- Future enhancements: a clear path for new ideas with an open standard

The DRM standard has been designed taking into account a number of technical constraints, among which are:

- Short access time for the receiver: The listener shall not wait for more than a few seconds before getting access to the desired program, and shall obtain radio broadcasting information even faster.
- Maximum quality (objective and subjective): In the allowed transmission bandwidth, a maximum useful bit rate must be conveyed. This implies a high-spectral-efficiency modulation scheme—more than 2 bit/s/Hz.
- Robustness against distortions (multipath, Doppler, noise): This is mandatory, especially in the shortwave bands (which are often severely affected by propagation disturbances and interference) or in medium waves during the transition between day and night.
- Flexibility: According to the current broadcasting frequency band, the frequency separation of different transmissions, the bandwidth of the transmitter and the total available bandwidth can be adjusted to the needs of the broadcasters. In the same way, the required protection level is not the same in LW, MW and SW and in a given band. It can vary according to the time of day. Moreover, the system should include operating modes that can be used in the transition phase, where simultaneous broadcasting (*simulcast*) of compatible AM is required. Notice that a change in any parameter needs no intervention from the listener, since the receiver is remote controlled by the transmitter.
- Minimum disturbance of AM users on the same band: This leads to the design of a signal which, as seen by an AM receiver, must be as noise-like as possible. The frequency spectrum of the transmitted signal must also be as compact as possible to minimize jamming in the adjacent channels.
- Low complexity: This is essential for having receivers of low complexity and low power consumption, especially in countries where batteries are rare and expensive.
- Graceful degradation: If desired, graceful degradation may be obtained by the use of hierarchical coding. This option, although compatible with the system design, is not described in this paper.

DRM benefits for the listeners are the following:

- FM-like sound quality with the AM reach
- Improved reception quality
- Flexible use of radio, whenever and wherever you want it
- No change to existing listening habits: same frequencies, same listening conditions (fixed, portable and mobile radio), same listening environment (indoors, in cities, in dense forests)
- Low-cost receiver, low energy consumption
- Easy tuning with selection by frequency, station name or programming
- More diverse program content, using the full capabilities of new digital features
- Wide receiver range with more and better features

Radios that will give you programs with associated text information, station name, record title, singer's name.

DRM System Description

With the limited bit rate available, it is important to strike the right balance between flexibility and efficiency while protecting each bit of information to an appropriate degree. A distinction is therefore made between main payload data and the various types of data that the receiver needs to help it find and decode the desired program.

The main payload is called the *Main Service Channel (MSC)*. Two subsidiary channels are also provided namely the *Fast Access Channel (FAC)* and the *Service Description Channel (SDC)*. These two are key to ensure simplicity of receiver operation and are therefore designed to be reliably received in adverse conditions, with different forward error-correction schemes from the MSC.

The FAC is intended to be decoded quickly by the receiver on first acquiring the signal (at switch-on, or during scanning). It carries a minimum of

[1]Notes appear on page 56.

constantly repeated data that might be essential at this stage: informing the receiver what bandwidth option is in use, what modulation is used for the SDC and MSC, which length of interleaver is used for the MSC, and so forth.

The FAC is naturally concentrated in the narrowest frequency bandwidth available (namely 4.5 kHz) so that the receiver needs only to receive and decode this bandwidth to know exactly what is transmitted. (See Fig 1.)

The SDC contains more data, also sent repeatedly but in a longer cycle to maintain efficiency. It contains an identification of the services available in the MSC, together with further information to instruct the receiver how to decode each service. Here, lists of alternative frequencies and frequency schedules would be transmitted if appropriate.

Finally, the bulk of the signal conveys the MSC. With the limited bitrate available within one 9- or 10-kHz channel, this would normally be used to carry one audio program, together with a modest stream of data. Nevertheless, there is a degree of flexibility, so the MSC may contain between one and four streams of data. Streams and services are distinguished as follows.

An audio service consists of one stream carrying audio, and optionally one stream carrying data. A data service consists of one stream carrying data.

The proposed system is based upon a multi-carrier modulation. It can be seen as a regular juxtaposition in the frequency domain of k elementary narrow-band subcarriers, each conveying a bit rate of d/k if d is the bit rate of the overall system. This choice comes from the robustness of such a waveform when a high spectral efficiency is necessary, while at the same time severe propagation conditions must be endured. Similar choices were made for higher-frequency terrestrial radio broadcasting (DAB, see Note 2) and TV broadcasting (DVB-T, see Note 3) in Europe, or more recently for the indoor high-data-rate wireless communications standard IEEE 802.11a at 5 GHz.

This leads to an optimum occupancy of the available bandwidth since the frequency spectrum of the signal is (almost) rectangular. This is possible because the signal that is conveyed by each carrier is orthogonal to the others. This property enables the signals conveyed by the different carriers to be separable, even if the narrow-band carrier spectra overlap. The adjustment of the system bandwidth is obtained via the modification of the number k of subcarriers, without any other change, especially at the receiver hardware and software level.

These definitions are used hereafter: The system conveys symbols that are located at known instants and frequencies. A carrier is the set of symbols that are located at the same frequency. A COFDM symbol is the set of symbols that are synchronously transmitted on all the used carriers. Hence, the number of symbols in a COFDM symbol is the number of used carriers. The COFDM symbols are grouped in a complete transmission frame that appears periodically with the same format. That is, the same repartition of reference and useful symbols. The notion of a transmission super-frame is also used to denote the group of transmission frames that starts with the SDC special group of COFDM symbols and contains exactly three transmission frames. As decoding the SDC is necessary to start decoding the whole bit stream, one can see that a standard delay to start reception is on the order of a super-frame.

Short access time is obtained by means of a few dedicated carriers that the receiver looks for in a first step for fast frequency synchronization. In addition, another set of carriers always contains the same information at a given instant—this corresponds to a constant known waveform/pattern which is used for pattern synchronization.

Maximum quality is obtained by the

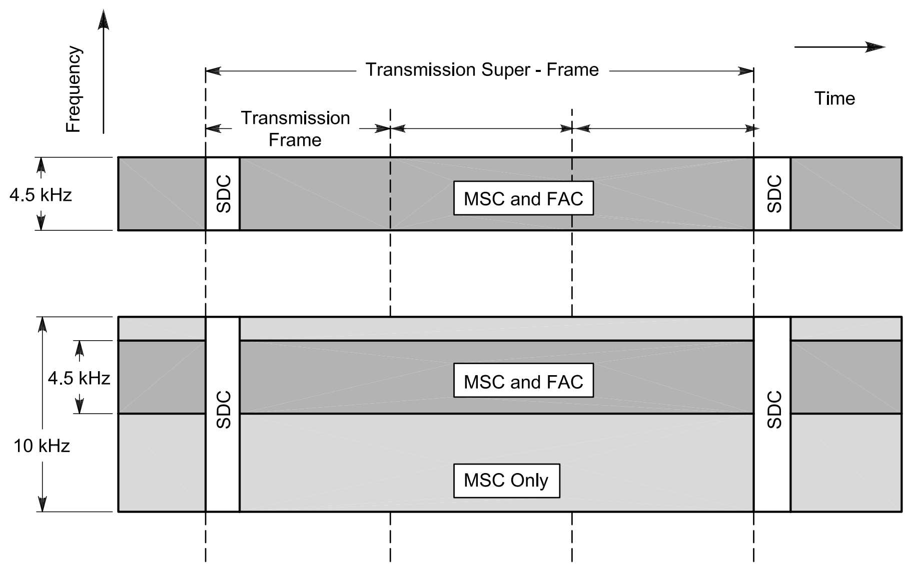

Fig 1—DRM time-frequency structure showing the various channel positions for 4.5 and 10-kHz bandwidth.

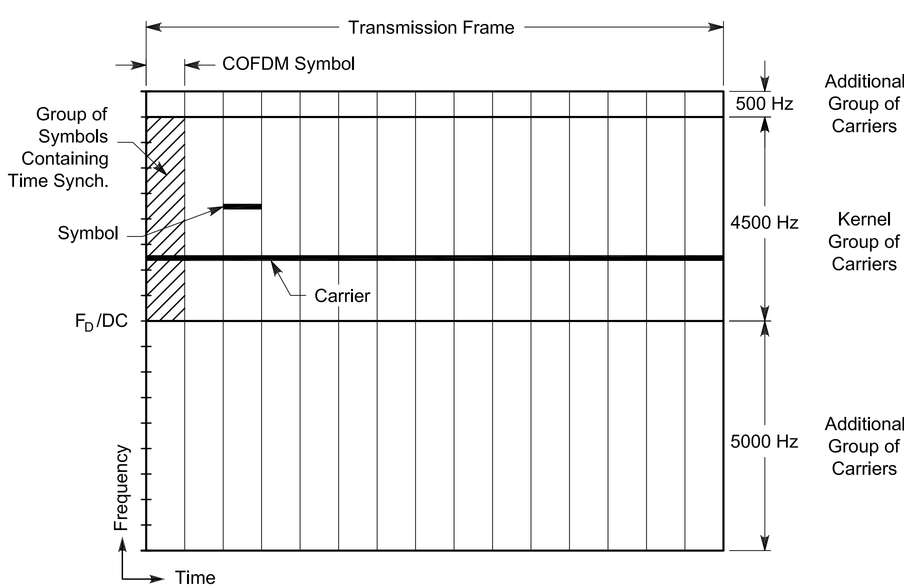

Fig 2—Symbol, frame and carrier definitions (example of 10-kHz overall bandwidth).

use of multilevel quadrature amplitude modulation (QAM). The proposed QAMs have levels of 4, 8, 16 or 64, the number of states being chosen as a function of the desired level of robustness. For this type of modulation, coherent demodulation is required; at each instant and at each frequency, it is necessary to estimate the complex gain of the transmission channel. To evaluate the channel response, some of the symbols—at predefined frequencies and instants—are sent with predefined amplitude and phase references so that the gain of the channel can be evaluated at any instant (using time interpolation) and any frequency (using frequency interpolation).

Robustness is achieved by the use of multilevel coding (MLC). An association of convolutional encoding and interleaving leading to an optimization of the transmission efficiency.[4, 5] In conjunction with coding, time and frequency interleaving are used, which have the effect of spreading perturbations (frequency-selective fading, flat fading, interference) on distant symbols, so that decoding is more efficient.

"Parametrability" is obtained by an incremental design of the broadcast signal. The signal contains a standard kernel group of carriers (occupying 4.5 kHz) that is common to all versions. The required bandwidth/bit rate is obtained by adding additional groups of subcarriers on either side of the kernel group. (See Fig 3.)

During the analog-to-digital transition phase, a small number of additional groups may be used. The unused spectrum will be occupied by an AM-compatible waveform, which can be received by classical AM receivers without any modification.

There is flexibility because the bit stream is divided into a main stream for conveying standard audio or audio plus still pictures or data and a system data stream with a much lower bit rate conveying additional data, such as program-associated data (PAD similar to RDS). With up-to-date audio encoders/decoders, the ratio of audio to pictures may be instantaneously variable in the main stream, and the significance of the data bit stream can vary as desired.

There is minimum disturbance to traditional AM users because the digital waveform has a flat spectrum so that, as received by an AM receiver, the digital signal sounds almost like white Gaussian noise. A pulse shaping of the transmitted symbols (that is, time windowing of the COFDM symbol) and/or additional output signal filtering can further decrease the disturbance of AM receivers in the adjacent channels.

Low complexity is inherent in the system. Since the signal can be seen as a number of elementary carriers uniformly spaced in frequency, the main digital signal processing is done by means of several fast Fourier transforms or inverse fast Fourier transforms (FFTs, IFFTs) which are known for their very efficient implementations. At the receiver, the FFT is equivalent to a large filter bank, each filter selecting only one subcarrier. The complexity level is only proportional to the occupied bandwidth, and it is independent of the channel quality.

Finally, ease of use of the system is obtained by automatic remote control of the receiver. Dedicated, highly protected symbols convey all the necessary configuration parameters: Any change in the transmission characteristics (bandwidth, coding, interleaving) is automatically taken into account by the receiver. These symbols are grouped in the FAC and the SDC channels.

DRM currently contains 4 modes:

- Mode A (ground wave): Gaussian channels, with minor fading adapted to LW and MW during daytime.
- Mode B (sky wave): Time and frequency-selective channels, with longer delay spread for SW and MW nighttime.
- Mode C (robust): Time and frequency-selective channels, with greater Doppler spread for bad SW channels.
- Mode D (extreme): Very robust mode, but with a reduced net bit rate (approximately half of the bit rate of mode A).

The first three modes are expected to cover most applications.

Mode A has a guard interval of $2\frac{2}{3}$ ms, together with a carrier spacing of $41\frac{2}{3}$ Hz. This is described as intended for "Gaussian channels, with minor fading," and is thus particularly suitable for local or national coverage at LF/MF, although it may also be useful in some longer-distance applications. The guard interval is sufficient for SFN operation.

Mode B can be described as intended for "Time and frequency selective channels, with longer delay spread." It has a guard interval of $5\frac{1}{3}$ ms, with a carrier spacing of $46\frac{7}{8}$ Hz and a higher density of pilots. The longer guard interval is intended to cope with greater multipath spread, as can be caused during multimode, multihop sky-wave propagation, while the greater carrier spacing and pilot density give greater tolerance to Doppler spread.

Mode C is devoted to "Fast varying time and frequency selective channels, with longer delay spread." It has a guard interval of $5\frac{1}{3}$ ms, with a carrier spacing of $68\frac{2}{11}$ Hz and the highest density of pilots. The long guard-interval is intended to cope with greater multipath spread, as can be caused during multimode, multihop sky-wave propagation, while the greatest carrier spacing and pilot density gives maximum tolerance to Doppler spread.

Each symbol is modulated independently from its neighbors by choosing a given point in a predefined constellation according to the value on an information word (2-6 bits). The reference symbols are transmitted with a predefined amplitude and phase. A reference symbol is used either for synchronization purposes or for channel complex-response estimation. A useful symbol must be demodulated and decoded in order to recover the original information it conveys.

The subcarriers are located at offsets from the center frequency that are multiples of $\Delta f = 1/T_u$, that is $41\frac{2}{3}$ Hz (system for ground wave) or $46\frac{7}{8}$ Hz (system for sky wave) or $68\frac{2}{11}$ (robust). By convention, in the complex baseband representation, the kth carrier is at an

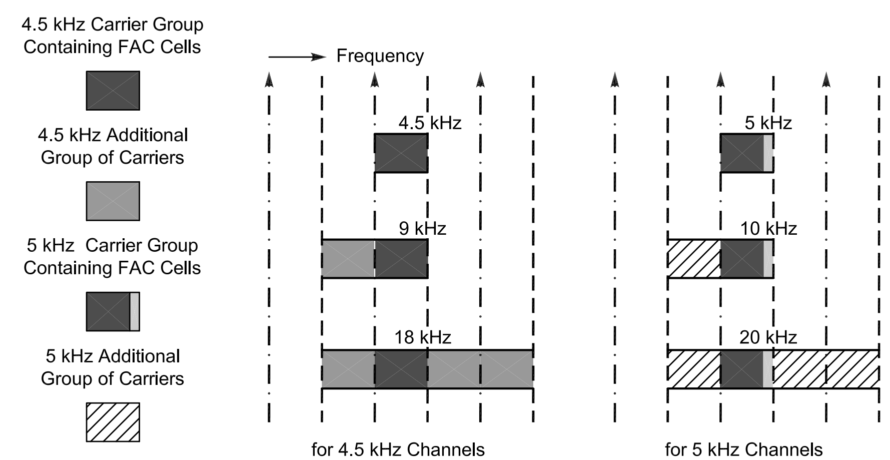

Fig 3—Various bandwidths possible.

offset of $k\Delta f$ from the dc component; k can be positive (to the "right" of the reference carrier) or negative (to the "left" of the reference carrier). The 0th carrier is dc. In the first two parameter sets the transmission frame contains 15 COFDM symbols; there are 20 COFDM symbols in the last one, leading to a constant common duration of 400 ms.

As already mentioned above, the carriers are grouped in a kernel group of carriers, which is common to all transmission modes. The group conveys all the reference symbols necessary for time and frequency synchronization, as well as the FAC symbols that describe the current mode. The kernel group is immediately above the carrier at frequency F_0 and does not contain it. Its bandwidth is exactly 4.5 kHz between the zeros of its frequency spectrum.

Additional groups of subcarriers (possibly none, in the 4.5-kHz version), the number of which is defined according to the desired bit rate and available bandwidth. An additional group does not contain any synchronization or FAC symbol. If there are additional groups below the carrier, their number is always such that the frequency spectrum of the signal is symmetrical around the carrier. The bandwidth of each additional group is exactly 1.5 kHz between the zeros of its frequency spectrum.

The source coding requirements are directly derived from the channel capacity. The capacity available for audio within a single 9- or 10-kHz channel is distinctly limited—at 20 to 25 kbps and perhaps as little as 10 kbps for some extremely unfavorable HF paths. This clearly represents a serious source-coding challenge for DRM, which expects to deliver good audio quality for both speech and music.

DRM has incorporated work and key technologies already done in the development of source coding elsewhere and fine-tuned it to this particular application. For coding most broadcast program material, an audio coder is needed to cope with the arbitrary mix of speech, music and incidental background sounds. For this purpose, DRM uses advanced audio coding (AAC) from the ISO MPEG-4 standard, supplemented by spectral band replication (SBR).

The SBR technique synthesizes the sounds that fall within the highest frequency octave. Sounds in this range are usually either:

1. noise-like (sibilance, percussion instruments such as shakers, brushed cymbals and so on), or

2. periodic and related to what

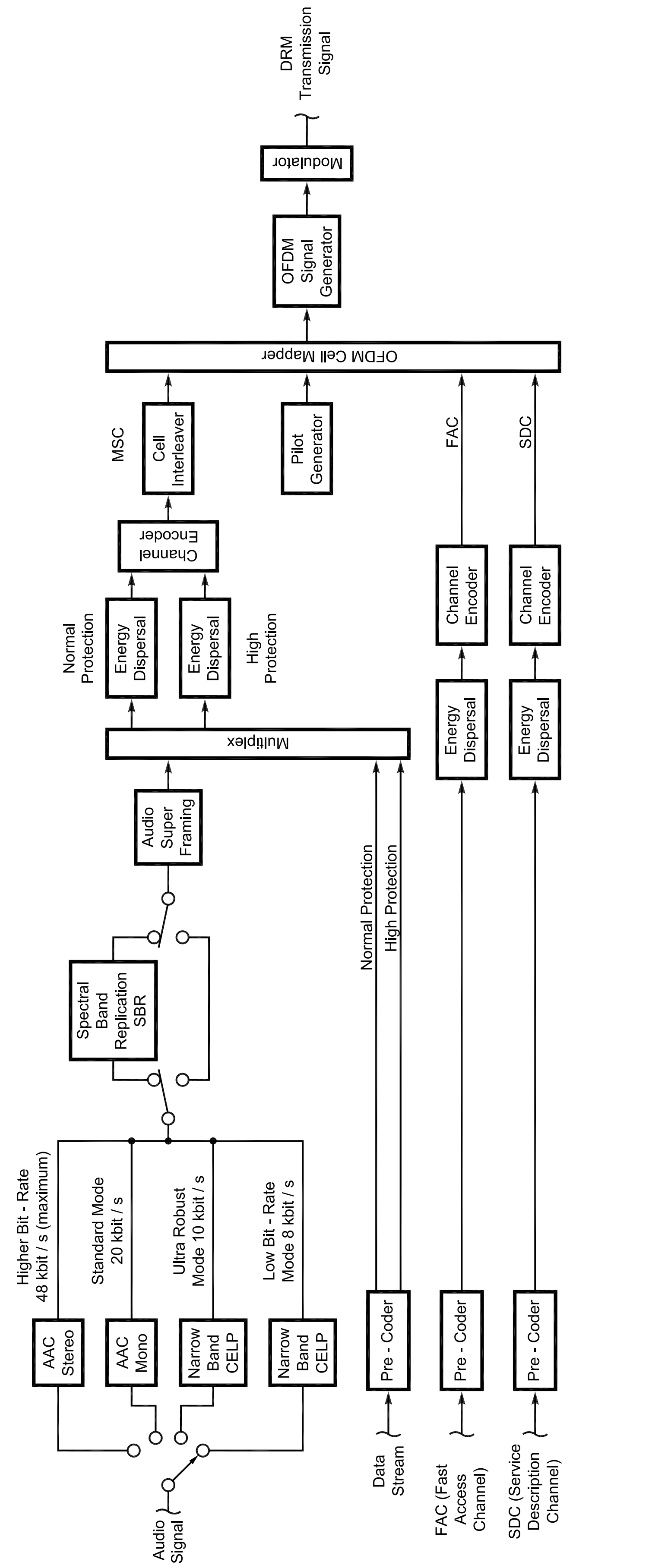

Fig 4—Complete DRM architecture for transmit side.

appears lower in the spectrum (overtones of instruments or voiced sounds).

At the sender, the highest-frequency band of the audio signal is examined to determine the spectral distribution and whether it falls into category 1 or 2 above. A small amount of side information is then prepared for transmission to help the decoder. The highest-frequency band is then removed before the remaining main band of the audio signal is passed to the AAC coder, which codes it in the conventional way.

At the receiver, the AAC decoder first decodes the main band of the audio signal. The SBR decoder then adds the synthetic upper band, helped by the instructions sent in the side information. Overtones are derived from the output of the AAC decoder, while noise-like sounds are synthesized using a noise generator with suitable spectral shaping.

An alternative for speech-only programming is to use a coder designed expressly for speech, in which case the bit-rate can be reduced much more than with a waveform audio coder, while retaining the same speech quality. Although this offers broadcasters further flexibility, there is, however, some doubt whether this approach would be used much in practice (apart from multilingual news broadcasting). Even speech-only broadcast material contains jingles, background sounds in interviews and so on, all of which can cause serious problems to speech coders. Figure 3 summarizes the various basic channel-content possibilities. Thus the complete system is given in Fig 4 (transmit side).

Adaptation to Amateur Radio Use

From the beginning, Thales has proposed to limit the kernel group of carriers to 3 kHz, thus enabling easily a 3-kHz mode for use in radio amateur HF or military applications.[6] So far, this mode has not been retained by the DRM consortium to keep the basic set of modes as limited as possible, thus lowering the development cost of the first version of the receiver chipsets.

Sensing the importance of such a mode, a demonstrator was developed anyway. Further modifications of the DRM system were applied while keeping the same overall structure:

• To limit the kernel group of carriers to less than 3 kHz.

• To adapt to push-to-talk mode, and other features of amateur transceivers.

• To include lower bit-rate vocoders (waveform coders are not usable at such low bit rates) imposed by the further reduction in available bandwidth (1200, 2400, 3200 bits/s voice coders are implemented in the demonstrator).

• To simplify the multiplex scheme and its associated description to reduce the bit rate for signals and have a minimum transmission delay (see above).

• To simplify intellectual-property issues.

The main similarities between the proposed system and the original DRM standard are the following.

Unmodified Features

• Symbol duration and guard times

• The positions of the three unequally spaced unmodulated carriers (pure tones), which help find rapidly the actual frequency shift of the received signal.

• The repartition and positions of the gain-reference symbols

• The modulation schemes (constellations)

• The interleaving principles (short and long interleaving)

• The convolutional codes—in practice, we use only the simplest one, for both vocoded audio and free user data. Its performance is, of course, the same as with other more complex encoding schemes.

Modified Features

• The SDC stream has been completely suppressed since it is no longer necessary. All the configuration information is located in the FAC only, so that the super-frame of DRM (three frames) now only contains one frame (latency reduction, software simplification).

• The FAC contains only 40 symbols (instead of 65), 26 useful bits (instead of 64) and is more protected (code rate 0.5 instead of 0.6). It contains the full set of parameters of the current transmission, as well as other information like the frame contents: A flag indicating the number of vocoder frames in the transmission frame, or the end-of-message (EOM) flag.

• The positions of the FAC cells and some frame-synchronization cells have been modified to fit in the reduced bandwidth.

• The whole original signal is shifted towards dc to fit in a bandwidth of 300-3000 Hz. The 3-kHz bound is always respected. The 300-Hz bound is only approximately respected because of the spacing between tones (333 Hz, 281 Hz and 273 Hz for modes A, B and C).

• Finally, each vocoder frame and each free data block is completed by an eight-bit CRC (parity check) to detect errors and minimize their impact on the audio quality.

In practice, the signal-processing software for transmission and reception are essentially the same as for the DRM system. The only changes are software simplifications and different primary data (constants or data arrays). The demonstrator was developed to show the highest quality achievable with current voice-compression techniques and the possibility to add new digital data services.

The voice codecs used are part of the Thales portfolio of coders (the HSX family for Harmonic Stochastic eXcitation). Thales Communications has been active in low-bit-rate voice codecs since 1968. In particular, the 800 bits/s Stanag (Standard NATO Agreement) and the ETSI Tetra coders (for Professional Mobile Radio, the European equivalent of mode-25 radios) came out of these laboratories.

The advantage of having coders from the same family is that they share a lot of common parts, allowing for simple switching of coders, reduced code size and thus simple maintenance or even hierarchical error-correcting schemes. Such schemes correspond to transmis-

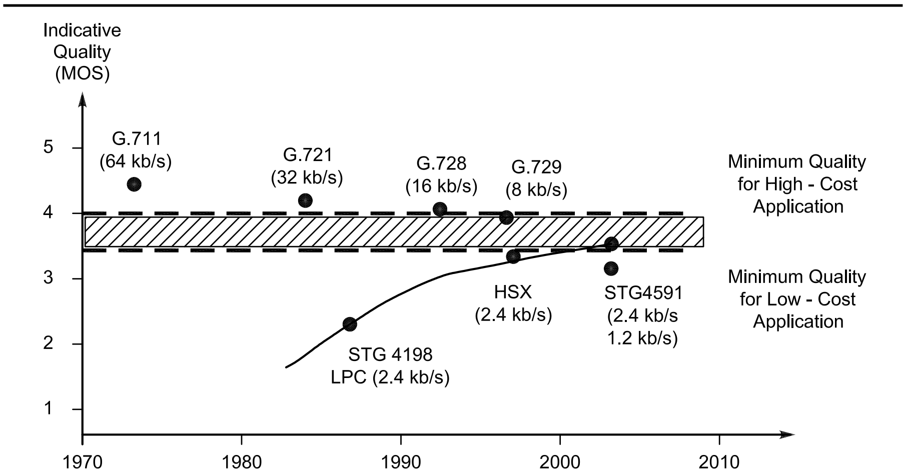

Fig 5—Evolution of voice coders quality through time.

sions where the additional information necessary to go from a given quality to a better one is less protected. So quality at reception varies depending on the ability of the modem to decode part or all the information, thus insuring at least the minimum quality and at best, the best quality when the channel permits it. Such a scheme is not implemented yet, but it could be added in a future version of the system.

Fig 5 shows the evolution through time of the quality of voice coders. The middle-range quality lying between MOS (mean opinion scores) of 3.5 and 4 are considered as the range acceptable for low- to medium-quality commercial applications. The latest 2400-bits/s coders have equivalent or better quality than older 4800-bits/s ACELP coders (adaptive code-excited linear prediction), while-1200 bit/s coders become of acceptable quality. New research in further rate reduction allows hope for 600 bits/s in a few years. Competition for a high-quality 4-kbits/s codec has been ongoing for several years without anything yet satisfying all the very stringent constraints (such as a very low delay). All the G.xxx codecs are standardized by the ITU, while the STG coders are military NATO standards—also named Stanags.

Other potential coders could have been:

• DVSI: proprietary format (IMBE and AMBE: 2 to 9.6 kbits/s) mainly chosen for satellite communications systems (Inmarsat), Iridium and APCO 25.

• MELP and EMELP: basis for the STG 4591 Stanag

The main reason for not choosing them is that the first is a proprietary, not easily available format, and that both come with IPR licensing fees and conditions, without any important gain in quality.

Our demonstrator software is 100% PC based for simple integration with standard off-the-shelf transceivers. A simple and intuitive man/machine interface (MMI) based upon the use of *LabWindows* software has been developed. The demonstrator has been fully integrated with Ten-Tec off-the-shelf transceivers. Transatlantic tests with ARRL are underway. A screen dump or the MMI in receive mode is given in Fig 6.

The six sub-windows show (left to right, top row first): the time values of the received signal, the spectrum of the received signal (filtered within 3 kHz), the reception level (estimation of the signal-to-noise ratio, SNR), the audio output signal, the received constellation (after time and frequency synchronization) and the estimated channel impulse response (here two paths are present with around 90° shift between them).

Other information are given such as the text message received, the mode detected, the estimation of the Doppler shift, the estimated instantaneous SNR.

The demonstrator is able to provide various transmission applications:

• Real-time voice transmissions, using a microphone/speaker and a sound card in the PC

• Transmit digital data (hand-written text or even complete files). The maximum available digital data rate is directly related to the vocoder bit rate and the current error correcting code and modulation.

• Test the modem performance by sending a test sequence for bit-error-rate estimation at the receiver

To test the demonstrator in a controlled environment, a real-time SW propagation simulator was included to show the insensitivity of the digital part of the system to multipath, whereas at the same time, an analogue AM equivalent is seriously perturbed. The simulator may generate a maximum of four paths. All the characteristics (amplitude, average power, frequency offset, Doppler spread, time spread) are adjustable in real time using a natural man/machine interface. Narrow-band interference may be added for system tests. All these impairments are added at the transmit side.

The DRM consortium normalized laboratory test conditions to get a fair comparison between various proponents. The channel model selected is described hereafter.

The approach is to use stochastic time-varying models with stationary statistics and define models for good, moderate and bad conditions by taking appropriate parameter values of the general model with stationary statistics. 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.

A tapped-delay-line model is then used for multipath generation.[7, 8] The time-variant tap weights are zero-mean, complex-valued stationary Gaussian random processes. The magnitudes are Rayleigh-distributed and the phases are uniformly distributed. For each weight, there is one stochastic process, characterized by its variance and power density spectrum (PDS). The variance is a measure for the average signal power, which can be received via this path and the PDS determines the average speed of variation in time. This type of channel model is known as the "Watterson" model.

A number quantifies the width of the PDS and this quantity itself (the width) is commonly referred to as the

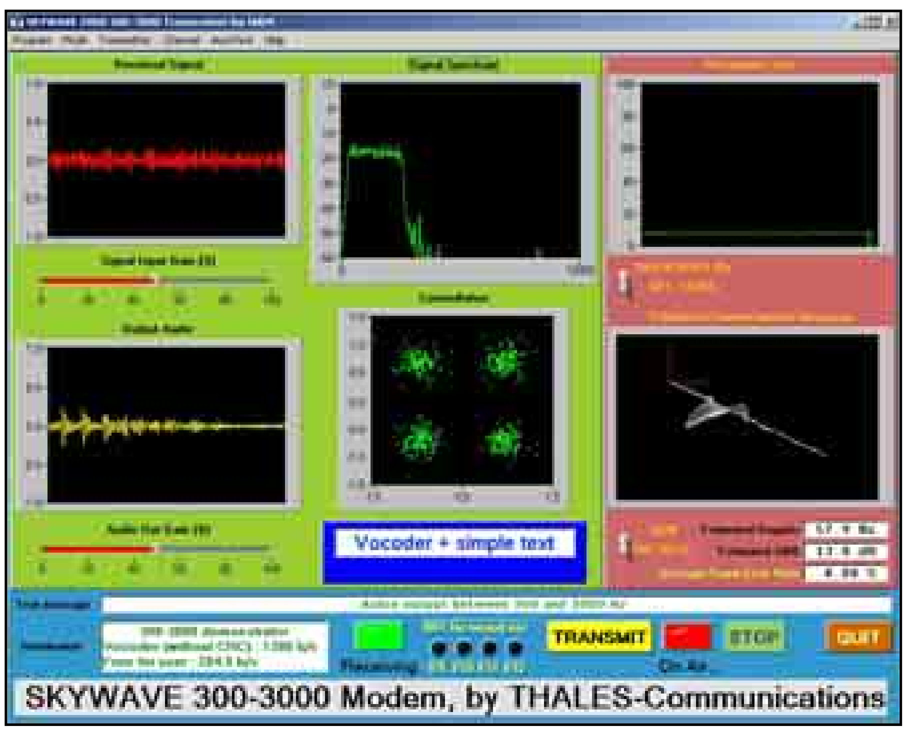

Fig 6—Demonstrator MMI while in receive mode.

Doppler spread of that path. There might be also a non-zero center frequency of the PDS, which can be interpreted as an average frequency shift (or Doppler shift).

It is common to assume a Gaussian amplitude statistic for the process, which is a reasonable assumption for an ionospheric channel. The stochastic processes belonging to every individual path then become Rayleigh processes. WSSUS does not define the shape of the PDS. For ionospheric channels, Watterson has shown a Gaussian shape to be a good assumption.[9, 10] The one-sided Doppler spread is then defined as the standard deviation (*s*) of the shape of the PDS.

Five channels are currently used by DRM:

- AWGN (additive white Gaussian noise): one path of constant amplitude (for ground-wave propagation).
- Ricean with delay: one constant path of unit amplitude, a second with half amplitude delayed 1 ms, with 0.1-Hz Doppler spread (for flat fading at MW and SW).
- US Consortium: four paths of amplitudes (1, 0.7, 0.5, 0.25), delay (0, 0.7 ms, 1.5 ms, 2.2 ms) Doppler spread (0.1 Hz, 0.5 Hz, 1 Hz, 2 Hz) and frequency shifts (0.1 Hz, 0.2 Hz, 0.5 Hz, 1 Hz).
- CCIR poor: two paths of equal amplitude delayed 2 ms and equal 1-Hz Doppler spread (SW propagation).
- Similar to channel 4, but with 4-ms delay and 2-Hz Doppler spread (bad SW propagation).

Conclusion

This paper presents a complete system proposal for Amateur Radio digital voice transmission derived from the DRM standard. It shows the current state of the art in COFDM modem technology. COFDM was chosen because of its very good robustness when a high spectral efficiency is desired, while at the same time severe propagation conditions must be endured. Such a system should start wide interest in digital voice, as the necessary computing power required is well within current PC technology reach (or DSP for future integration inside transceivers).

This system proposal is currently being tested (real-time demonstrator measurements) using a transatlantic link.

GLOSSARY

AAC	Advanced Audio Coding (MPEG-2/4)
ACELP	Adaptive Code Excited Linear Prediction
AFS	Alternative Frequency Switching
AM	Amplitude Modulation
ARIB	Association of Radio Industries and Businesses
AWGN	Additive White Gaussian Noise
BER	Bit-Error Rate
COFDM	Coded Orthogonal Frequency Division Multiplex
DAB	Digital Audio Broadcasting
DC	Direct Current
DRM	Digital Radio Mondiale
DVB	Digital Video Broadcasting
DVB-T	DVB - Terrestrial
EBU	European Broadcasting Union
ETSI	European Telecommunications Standard Institute
FAC	Fast Access Channel
FFT	Fast Fourier Transform
FM	Frequency Modulation
HF	High-Frequency
IBC	International Broadcasting Conference
IEC	International Electrotechnical Commission
IFFT	Inverse Fast Fourier Transforms
ISO	International Standard Office
ITU	International Telecommunication Union
ITU-R	ITU - Radiocommunication Sector
LF	Low-Frequency
LW	Long-Wave
MF	Medium Frequency
MLC	Multi-Level Coding
MMI	Man Machine Interface
MOS	Mean Opinion Scores
MPEG	Moving Picture Experts Group
MSC	Main Service Channel
MW	Medium Wave
NADIB	Narrow Band DIgital Broadcasting
OFDM	Orthogonal Frequency Division Multiplex
PAD	Program Associated Data
PDS	Power Density Spectrum
QAM	Quadrature Amplitude Modulation
RDS	Radio Data System
RF	Radio Fequency
SBR	Spectral Band Replication
SDC	Service Description Channel
SFN	Single Frequency Network
SNR	Signal-to-Noise Ratio
SSB	Single Side Band
SW	Short Wave
UEP	Unequal Error Protection

Notes

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