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Abstract

The year 2003 sees the official launch of Digital Radio Mondiale (DRM), the new broadcast system for short-wave, medium-wave and long-wave. As with many digital broadcasting systems that replace their analogue predecessor, DRM offers a long list of enhancements and new features that broadcasters should be aware of.

The paper presents DRM from the perspective of broadcasters who are considering introducing DRM services or switching their analogue transmissions to digital technology. What is the trade-off between audio quality and signal robustness? What is the impact on the production of radio programmes in the studio? What issues need to be considered on the transmitter side? What features are available to increase the radio listening experience? These and other questions will be considered in the paper.

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Digital Radio Mondiale: Features and Requirements from a Broadcaster's Perspective

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ABSTRACT

The year 2003 sees the official launch of Digital Radio Mondiale (DRM), the new broadcast system for short-wave, medium-wave and long-wave. As with many digital broadcasting systems that replace their analogue predecessor, DRM offers a long list of enhancements and new features that broadcasters should be aware of.

The paper presents DRM from the perspective of broadcasters who are considering introducing DRM services or switching their analogue transmissions to digital technology. What is the trade-off between audio quality and signal robustness? What is the impact on the production of radio programmes in the studio? What issues need to be considered on the transmitter side? What features are available to increase the radio listening experience? These and other questions will be considered in the paper.

INTRODUCTION

In 1998, the DRM consortium was formed to define a new digital radio system aimed to replace AM radio on long-wave, medium-wave and short-wave. Three years later, the International Telecommunications Union (ITU) recommended the system for worldwide use. As soon as the first prototype modulators were available at the beginning of the year 2000, a series of field trials was started, which, partitioned into several phases, aimed at

- evaluating the developing standard by assessing its capabilities and limits
- demonstrating the system
- exploring new possibilities, such as the transmission of multimedia content and electronic programme guides.

Some of the experience gained during these tests is summarised in the following sections of this paper. As with any kind of new broadcast system, the introduction of DRM is not a trivial exercise. From a broadcaster's point of view it is beneficial to consider potential impacts on the whole production process of a radio programme. Recording raw material in- or outside the studio, editing the material to transform it into a final programme, distributing material to transmitter sites followed by the transmission itself, and, last but not least, monitoring of the transmitted signal in the coverage area.

PROGRAMME CONSIDERATIONS

DRM will provide a wide range of opportunities for broadcasters, both those concerned with national and international broadcasting. Additionally there will be opportunities for new services from both current and new broadcasters.

The question is, will the services provided just be "more of the same" or should broadcasters be considering what DRM has to offer and tailor their DRM offer accordingly.

For example, the BBC, as an international broadcaster, could just take its existing network output and feed it to the DRM transmitters. Although in the short term this may be the most attractive way of getting a service on the air, the longer term needs for DRM broadcasts should be considered – many of the changes required will need integration into the existing infrastructures and will not be done overnight. So the need to start planning what will be required starts now.

Let us consider the evolution of international broadcasters such as the BBC over the last few decades. Twenty years ago the majority of the international output of the BBC was fed to listeners via conventional shortwave services along with some medium wave outlets. Quality was and still is poor over shortwave – that is one of the fundamental needs for developments such as DRM. On the plus side – geographical coverage was wide, a number of countries could be covered with a single transmission and indeed this advantage stays with us in the DRM era.

Poor audio quality brought with it a certain style of presentation and programme content. For example news was read in a slow and considered manner, music output was low because over shortwave and with a narrow bandwidth a lot was lost. Widespread geographical coverage also meant that news content was presented from the global or at least the continental perspective. All of this lead to a "one size fits all" approach to programming and led to complaints from the listeners from one area that they didn't want to hear about news from another area.

With the advent of deregulation in the FM spectrum considerable opportunities were presented. Local FM for cities were available, programming could become more targeted and became more competitive against local FM operators.

Without careful consideration of what programmes are carried on DRM and what are covered on FM, there is a potential that the service may decline from being inappropriately targeted at audiences and a unique position may be lost.

RECORDING AND EDITING

After having been planned and commissioned, a programme needs to be recorded and edited. For this purpose, broadcasters make increasing use of computer-based, non-linear editing systems. For cost reasons, the latter keep hard-disk space to a minimum and therefore compress the sound using a variety of compression systems. Bearing in mind that the final programme will be compressed again in order to fit the corresponding DRM signal into a long-wave, medium-wave or short-wave channel, it is vitally important to use a low compression rate (i.e. high bit rate) for the editing processes (for example 384 kbit/s MPEG Layer-3). This reduces the detrimental effects of cascaded audio coding.

DISTRIBUTION

The same considerations as described for the recording and editing process are true for the signal distribution. The effects of cascaded coding should be avoided, or at least reduced.

The DRM Multiplex Distribution Interface (MDI) offers the possibility to encode the programme at the studio output, thus producing a bit stream containing the compressed audio (using one of the audio encoders built into the DRM system) as well as information describing the service such as the station label, 128-character text message and a list of alternative frequencies. Note that a DRM multiplex can contain up to four different audio and/or data services, under the condition of course that they share the available bit rate.

The extra information as well as the additional protection add about 20%-25% to the useful programme bit rate, with the result that a typical MDI stream will fit into a transmission channel offering a capacity in the range of 32 kbit/s to 64 kbit/s (the exact value obviously depends on the various DRM audio encoder and modulator settings).

However, this distribution method assumes that all transmitters that are fed with the same MDI stream transmit the same content, using the same modulator settings. If a broadcaster intends to transmit the same audio programme from different transmitters, but not necessarily using the same signal robustness, the "mother" signal needs to be distributed at a much higher bit rate to avoid tandem coding artefacts. Coding Technologies, the inventor of the bandwidth extension technique SBR (which is part of the DRM specification) suggest the following bit rate ranges for commonly used audio codecs (see Table 1). Note that the values are only estimates and probably need further research into transcoding effects.

Table 1 Distribution Bit Rate Ranges For Mono Signals

Codec	**Bit rate in kbit/s**
MPEG Layer-2	96…128
MPEG Layer-3	64…96
MPEG-2 AAC	48…64
aacPlus	32…48

TRANSMISSION

In difference to conventional AM transmissions, DRM requires the transmitter operator to specify a large number of modulation parameters, thus shaping the signal to specific needs:

- The RF bandwidth can be chosen between 4.5, 5, 9, 10, 18 and 20 kHz. This leaves enough flexibility to simulcast alongside an analogue SSB signal in a single channel, or, in the other extreme, to use two adjacent 9 or 10 kHz RF channels together.
- Four different robustness modes adapt the signal to fundamentally different propagation scenarios: medium-wave (mode A), benign short-wave (mode B) and challenging short-wave (as found with spread-F conditions after sunset in areas served by tropical broadcasting), for which modes C or D would be used.
- Six different combinations of code rates and constellations are allowed. They determine the robustness with respect to interference and low

signal strength. In descending order of robustness, the combinations are: 16-QAM with code rate 0.5, 16-QAM/0.62, 64-QAM/0.5, 64-QAM/0.6, 64-QAM/0.71 and 64-QAM/0.78

The choice of the parameters depicted above determines the useful bit rate. As a result, the latter ranges from 4.8 kbit/s (bandwidth 4.5 kHz, mode B, constellation 16-QAM, code rate 0.5) up to 72 kbit/s (bandwidth 20 kHz, mode A, constellation 64-QAM, code rate 0.78). Neither of these extreme values is likely to be used very often though. Typical bit rates that are currently being employed for DRM tests are between 14 kbit/s and 34 kbit/s.

Naturally, the available bit rate has an important impact on the achievable audio quality. The broadcaster needs to decide which audio quality is acceptable for a given content. If the programme only contains speech, then one of the two DRM speech encoders (HVXC or CELP) might be suitable to do the necessary compression. As soon as music is present, AAC or aacPlus (the latter extends AAC by applying bandwidth extension through a technique called 'Spectral Band Replication', or SBR) should be the coder of choice. Table 2 intends to give an idea about the bit rates that are covered by the individual audio coders. Note that the recommended bit rate ranges overlap, thus further stressing the point that the choice is not straight-forward and will require each broadcaster to assess individually which codec and bit rate fits a given programme best.

Table 2 DRM Audio Codecs

Codec	Content	Possible bit rates	Authors' recommended bit rates
HVXC	Speech	2-4 kbit/s	2-4 kbit/s
CELP	Speech	4-24 kbit/s	8-16 kbit/s
AAC	Speech/Music	8-20 kbit/s	12-20 kbit/s
aacPlus	Speech/Music	14-72 kbit/s	20-72 kbit/s

In the end, the choice of bit rate will always be a trade-off between signal robustness and audio quality. The signal robustness (and therefore usually the coverage area) can be increased at the expense of audio quality. Normally, a broadcaster will want to achieve the highest possible coverage at the lowest acceptable audio quality. But care needs to be taken to make sure that this choice does not compromise one of the big advantages of DRM over AM: the superior audio quality.

The modulator settings given in Table 3 seem to be a good starting point (the 'Constellation' column designates the constellation used for the Main Service Channel and the bit rates are given in bits per second).

For those broadcasters who are able to use several transmitters simultaneously, DRM opens up another exciting field: the synchronised transmission of the same programme from different transmitters. Depending on the choice of frequencies, this transmission technique is referred to as Single Frequency Network (SFN) or Multiple Frequency Network (MFN). Both networks can be set-up from a single transmission site or, alternatively, from different transmission sites, thus suiting a multitude of needs.

Both network options should increase the reception reliability, either by reducing the risk of flat-fading over the entire channel bandwidth (SFNs), or by offering the same signal on a different frequency which is subject to different fading statistics (MFNs). DRM receivers supporting the Alternative Frequency Switching (AFS) mechanism will then have the opportunity to tune themselves automatically to the strongest frequency and reduce the occurrence of audio dropouts considerably. For 2003, a series of MFN/SFN tests is planned and the results are hoped to give broadcasters guidance as to which combination of transmission sites and frequencies give the best reliable coverage for a given target area.

MONITORING

Monitoring plays an important role in the world of AM broadcasting and this remains true for DRM. There is a big difference though: whereas monitoring of AM transmissions usually requires the ear of a professionally trained person, the errors of a received digital audio stream can be evaluated automatically by calculating audio checksums.

The audio stream is transmitted in the form of audio frames. Each audio frame contains a checksum that allows the associated information to be checked for correctness. Single audio frame errors are usually concealed, but the audio decoder is forced to mute the audio output as soon as too many audio frames are lost (approximately the equivalent of 400 milliseconds or more). Automatic evaluation software can easily count the number of corrupted audio frames and compile an audio dropout distribution for each reception hour.

Assuming that several monitoring receivers are placed into the target area, several dropout distributions will be available. Current research is looking into combining these distributions into a compact format (ideally a single number) that makes it easy to assess how acceptable a transmission was.

In case that a transmission is found to be unacceptable, the automatic evaluation software should be able to decide what the cause of the unacceptable dropouts was. There are several potential causes:

- Low signal strength
- High levels of RF noise
- Co-channel or adjacent channel interference
- High fading rates (Doppler spread) and/or multipath (high Delay spread)

The signal strength is relatively easy to measure, but requires a calibrated receiver. If fed with a signal received on a calibrated antenna, it is even possible to derive the field strength (in dBµV/m), thus allowing for a comparison with coverage predictions.

DRM receivers are also able to estimate the signal-to-noise ratio on the reception frequency, which, in combination with the field strength measurements, can reveal high noise levels at the reception site. Note that the choice of the monitoring locations has a strong influence on the results, and it is therefore crucial to select these sites carefully in order to reflect typical reception conditions.

AM interference can also be assessed automatically and tests are being carried out to evaluate different ideas, most of them looking at the degradation of the signal-to-noise ratio of specific carriers in the multi-carrier signal. Each DRM receiver also needs to estimate the channel impulse response, which can be used to get an indication on Doppler and Delay spreads.All of the necessary information is provided by the current generation of field trial receivers at an update rate of 400 milliseconds. The corresponding data format was agreed within DRM and it is envisaged to standardise it through the International Telecommunications Union (ITU).

In January 2002, DRM members started to transmit several hours of daily DRM programmes. Since then, an increasing network of DRM receivers (16 in December 2002) collected a large amount of reception data, which is regularly sent to a central database via email (typically 100 kbyte per day and receiver). The data is processed automatically and daily reception reports are generated and sent to a list of interested people. Figure 1 shows an extract of such a report for a transmission from Jülich (Germany) to receivers in England, Germany, Norway and The Netherlands.

The experience gained with these long-term tests is very likely to help DRM broadcasters to set-up their own monitoring network (or, more often, share a monitoring network with other broadcasters) in the near future. It will allow transmitter operators and broadcasters to take a daily decision about which modulation parameters should be chosen to serve the target in the best possible way. It would even be imaginable to take the decision on an hourly basis by feeding back the reception data in real-time. Research is currently underway to investigate this concept (the EU-funded QoSAM project for example).

SUMMARY

As a new digital radio system, DRM offers many possibilities to transmit audio and data in a form that significantly improves upon the listener's experience with AM radio. Alongside the increased attractiveness of radio reception over long distances for the listener comes the broadcaster's and network operator's challenge to produce, distribute and transmit radio programmes suitable for the DRM medium in a way that reaches the listeners in an optimum way.

Extra care needs to be taken to ensure that the effects of cascaded audio compression are minimised throughout the production chain, and the right signal distribution method needs to be chosen in order to achieve high reception reliability. This could involve multiple transmission sites operating on the same or on different frequencies and/or continuous feedback from monitoring receivers allowing the modulator parameters to be adjusted dynamically.

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QoSAM: http://www.ist-qosam.com

BBC: http://www.bbc.co.uk

Table 3 Useful Modulator settings to start from

	Bandwidth	Mode	Constellation	Code rate	Interleaver	Bitrate	Audio coder
MF	9 kHz	A	64-QAM	0.6	Short	23620 bit/s	aacPlus mono
MF	10 kHz	A	64-QAM	0.6	Short	27880 bit/s	aacPlus mono aacPlus stereo
MF	18 kHz	A	16-QAM	0.62	Short	34100 bit/s	aacPlus stereo
MF	20 kHz	A	16-QAM	0.5	Short	30540 bit/s	aacPlus stereo
HF	10 kHz	B	64-QAM	0.6	Long	20960 bit/s	aacPlus mono
HF	20 kHz	B	16-QAM	0.62	Long	29800 bit/s	aacPlus stereo

Figure 1 Daily Field Trials Report

```
----------------------------------------------------------------------------------------------------
13:00-14:00  2002-12-18  Mode:B  MSC:64   SDC:16  Prot:A0/B1  17760 bps   5975 kHz "Showcase!"
----------------------------------------------------------------------------------------------------
RxName     Quality[%]   AudioQ>0%: MER  AudioQ=100%: MER   Signal [dBuV]    Delay90[ms]    Doppler[Hz]
          Slot  Audio      10% [dB]     10% Med 90% [dB]   10% Med 90%      10% Med 90%    10% Med 90%

BBC1      99.9   99.9         20          21  22  23         25  34  47      0.5 1.0 3.3    0.4 0.5 0.8
DW1       95.6   95.7         14          18  21  23         42  48  62      0.2 2.9 4.9    0.3 0.8 2.0
FhG1     100.0  100.0         20          20  22  23         27  40  46      0.2 0.2 2.4    0.3 0.7 1.3
NOR1      99.4   99.4         18          20  22  23         34  40  56      0.6 2.3 5.2    0.4 0.7 1.7
NOR5      99.5   99.4         16          19  21  23         34  40  54      0.2 1.9 5.1    0.2 0.7 1.7
RNW2     100.0  100.0         19          19  22  22         45  58  72
RNW3     100.0  100.0         21          21  22  23         41  55  64      0.1 0.2 3.0    0.4 0.8 1.3

Evolution of Audio Quality Q in 1-Minute Steps. L is the accumulated dropout length in seconds.
0%<=Q<10%: _                  10%<=Q<90%: -                 Q=100%: `
5<L<=6: 5      4<L<=5: 4      3<L<=4: 3      2<L<=3: 2      1<L<=2: 1      0<L<1: *

BBC1    ``*``````3```````````*```````````````*``````````````
DW1     4````****````````14--*5*`*`1`*`***----1**1**311--*-1
FhG1    ``````````````````````````````````````````````*`````
NOR1    `````***1`*****`*`******2**```****21``***1*1***``*``
NOR5    ```******```**``*``***212**```****11`*`1***1*`*`````
RNW2    ```````````````````````````````````*````````````````
RNW3    ````````````````````````````````````````````````````
```