COFDM:
The modulation system for digital radio

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1. Introduction

This paper explains the background of COFDM, the new modulation system used for digital audio broadcasting. Together with other papers in this session of the 4th International DAB Symposium, it gives an introduction to the technology behind the new Eureka 147 system for Digital Audio Broadcasting.

2. The aims of digital broadcasting

Broadcasting radio is about programmes: music, news, drama, entertainment. It is not about technology. Listeners do not care about the technology of broadcasting. They are interested in the programme material.

Digital broadcasting is not therefore the prime goal. But it is the means to achieve the prime goal.

The history of broadcasting has been one of continued expansion. Even in the early days of broadcasting in the 1920s, there was a shortage of spectrum for broadcasters. As time has gone by, we have seen more and more newcomers to the broadcasting scene, and the appetite of the public has risen. Even though television was predicted to kill radio dead, we have seen radio focus its programmes to suit the needs of the public. Now it provides entertainment and information to people who are active or on the move, as well as to the original audience, the home listener. The old home receiver was large, with valves, using a large outdoor antenna and driven by mains electricity. This has now given way to transistor portables, and car radios. These use smaller antennas.

As the public has become more interested in listening to radio (in my country the average radio listener listens to over 20 hours of radio a week), more and more broadcasters have sought access to the FM band. This is now so full that, in many countries, quality is severely limited by interference. The reception was not improved by the widespread adoption of antennas that were unable to discriminate between wanted and unwanted signals.

The public therefore were demanding choice, and they wanted their choice to be available in reasonable quality, even on portable radios at home, or when moving around in the car.

These factors were translated into the engineering equivalents:

- mobility
- spectrum efficiency
- quality

The question then came — how can we satisfy that demand?

3. The problems to be overcome

The early attempts to find a solution did not work. Digital systems are often associated with the global quality of ruggedness. However, most of the early experience was based on simple QPSK systems which had been used for telecommunications. Such telecommunications systems nearly always used large directional antennas, which provide significant reduction in the level of any interference or unwanted signals arriving from different directions.

In our new broadcasting environment, large antennas are not acceptable. Small antennas do not give us the ability to discriminate between the wanted signal and all the likely unwanted signals. This means that a normal radio is often plagued with multipath reception. Signals arrive from a range of different directions, having been reflected from buildings, hills and mountains, vehicles etc. The characteristic of this type of reception is frequency selective fading.

We have to rely on the modulation system to overcome frequency selective fading. FM is not good in this respect, which is one of the reasons we are seeking a new modulation system. The first trials of digital signals for broadcast radio showed that simple systems were not able to deliver rugged reception if a simple antenna is used (the system went on to become NICAM 728, the digital system for television — which normally uses reasonably directive antennas). Part of the problem is that the bandwidth of the signal was narrow, and often the whole signal was lost during fades. Other techniques were then tried, including equalisation of QPSK and spread spectrum systems.

Equalisation of QPSK is an attractive proposition, but no one has yet made a system that will track the rapid variation of reception conditions found in a car. It is most unlikely that a narrowband QPSK system will ever be made to work.

Spread spectrum systems appeared to offer some hope, but only if the broadcasters took control of the majority of the VHF spectrum. This is of course impractical.

Something new was needed. Fortunately COFDM was available.

4. COFDM: The new Technology

Coded Orthogonal Frequency Division Multiplex (or COFDM to give it its more usual title) is an evolution of ideas whose time arrived in the mid 1980s. At this point, Daniel Pommier's team of engineers at CCETT[1,2] linked the early papers of Wienstein and Ebert[3], which described the concept of a rugged modulation system, with the new DSP technology that was just becoming powerful enough to deliver a working system.

This new system avoided many of the problems of the multipath environment, and provided a very rugged broadcast channel.

4.1 FDM

The problem with receiving signals which are subject to multipath propagation is that the channel is often distorted by frequency selective fading. Part of the signal spectrum is lost. In conventional modulation systems this leads to severe distortion which in turn leads to severe errors in the receiver.

The main characteristic of COFDM is that it is a frequency division multiplex (the FDM in the name). There are many carriers which are broadcast, each carrying part of the transmitted data stream at a very low bit rate. Inevitably, some of the signal will be distorted, but the majority of the carriers are received without too much distortion. Multipath distortion is less disturbing in low bit rate channels than in high bit rate channels. Thus, using a multicarrier system using many low bit rate channels means that a lot of the original data can be received accurately.

We can then use forward error correction to recover any data that may have been lost.

A key parameter in an FDM system is the bandwidth of the signal. If we adopt a bandwidth that is relatively narrow, the whole of the signal can be lost in a selective fade, and we need to broadcast more power to compensate. What is needed is a signal that is sufficiently broadband to provide useful power, even if there are frequency-selective fades. The minimum bandwidth appears to be around 1 MHz in most real environments. There are benefits to be obtained as the bandwidth increases to around 2 MHz, with the law of diminishing returns starting to apply after that. The Eureka system uses a bandwidth of 1.536 MHz which has delivered good quality reception in every country that has tried it.

The number of carriers used depends on several features. The more carriers there are, the lower is the bit rate on each carrier. This makes the system more

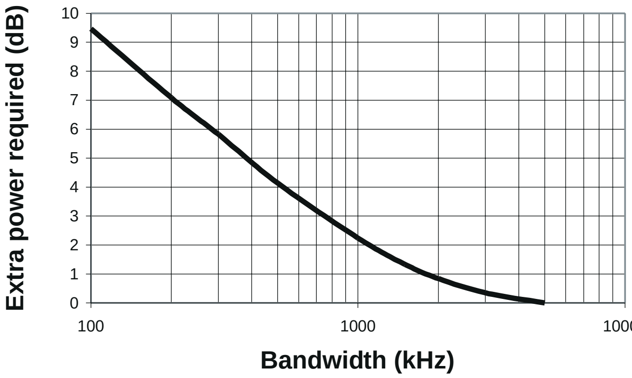

Figure 1 - Extra power required to compensate for reduced bandwidth of an OFDM signal (dB w.r.t. 5 MHz).*

* Source, ITU-R Special Publication on Digital Sound Broadcasting.

immune to multipath, but more sensitive to Doppler shift if the receiver is moving. The Eureka system offers four different modes which make it suitable for a wide range of applications and frequency bands.

4.2 Coding

The use of a frequency multiplex alone gives a significant improvement in performance, but it is not sufficient for mobile radio. During fading, some bits will be received in error while others are received correctly.

By using an error-correcting code it is possible to correct many or all of the bits which might otherwise be incorrectly received. In the Eureka system, we first ensure that any errors are largely random, and then use a convolutional code to remove the majority of the errors. To make the errors random, the transmitted data are spread across all the carriers, and interleaved it in time. Any transmitted bit may be separated from the next piece of data by up to 1.5 MHz and 340 ms.

Five levels of protection are offered in the DAB standard. Convolutional code rates* from about 1/3 to 3/4 are available, but the usual option for broadcast purposes gives an average rate of around 1/2.

Not all errors are removed by such a system, and when it does go wrong, there can be a burst of errors in the output. Many transmission standards use a Reed Solomon block code to improve on the error performance. In the Eureka system, this is not used because it makes very little difference to the performance of a mobile audio channel, but does use up a significant amount of capacity. We have adopted a strategy which provides more protection to the more important bits in the audio frame. This unequal error protection together with a cyclic redundancy check, enables us to identify errors and to carry out an error concealment strategy. This leads to an acceptably smooth degradation of quality at the edge of the service area.

If the system is used to carry data, instead of audio, then a there are specific choices of convolutional code available. If necessary, the data application can offer its own error protection code to ensure higher reliability.

4.3 Orthogonality

The early FDM systems, spaced the carriers apart in such a way that the signals could be received using a large number of separate demodulators, each with its own filter. In such receivers, guard bands have to be introduced between the different carriers, and the introduction of these guard bands in the frequency domain results in a lowering of the spectrum efficiency.

* The code rate is the number of useful bits to transmitted bits.

It is possible, however, to arrange the carriers in an FDM signal so that the sidebands of the individual carriers overlap and the signals can still be received without adjacent carrier interference. In order to do this the carriers must be mathematically orthogonal.

The carriers are linearly independent (i.e. orthogonal) if the carrier spacing is a multiple of $1/\tau$ (τ is the symbol period).

In practice, the signal is generated using an inverse fast Fourier transform (iFFT). This guarantees the orthogonality of the signal.

4.4 The Use of the FFT in COFDM

The implementation of COFDM, both at the transmitter and at the receiver uses the fast Fourier transform (FFT) algorithm to perform the modulation/demodulation process on the many carriers.

The signal is defined in the frequency domain. Each COFDM carrier is defined as one element of a discrete Fourier spectrum. The modulation that is used is a version of QPSK, so the amplitude is fixed (at unity) but the phase varies with the coded data.

In order to transmit the signal, the frequency domain representation is transformed, using an inverse FFT, into the time domain. This signal is then transmitted, and at the receiver, the reverse process is applied to recover the data.

The Eureka DAB system became feasible when VLSI became available which was capable of carrying out the required iFFT at the transmitter and the corresponding FFT at the receiver. What was state of the art technology at the beginning of the project is now easily achievable. This means that receivers can be made at prices the public can afford.

4.5 Synchronisation

One of the crucial problems in the receiver is to sample the incoming signal correctly. If the wrong set of samples is processed, the Fourier transform will not correctly decode the data on the carriers. It is especially difficult when the receiver is first switched on. There is therefore a need for acquiring timing lock.

In DAB, coarse synchronisation is provided by a simple analogue sync pulse. All the carriers are switched off on a regular basis. By using a simple amplitude detection circuit to detect this null symbol it is possible to generate a course approximation to the timing. However, the timing will not be perfect, and all of the samples could be displaced by a fixed time offset.

If the signal were truly time repetitive, as formally required for the FFT to be applied, then the effect of

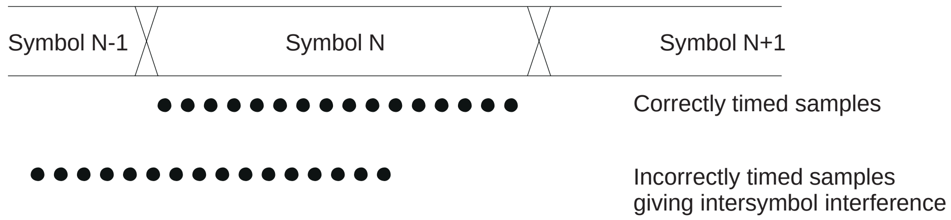

Figure 2 - The effect of incorrect synchronisation on a system with no guard interval.

this time displacement is to modify the phase of all the carriers by a known amount. This is determined by the time shift theorem in conventional transform theory.

However, the signal is not truly repetitive, we have cheated and performed the mathematical transform as if it were truly repetitive, but then chosen different symbols and transmitted them one after the other. The effect of the time shift would then be not only to add the phase shift referred to above, but also to add some interference from the transmitted data of the adjacent symbols. This interference could severely degrade reception.

To avoid these problems, more than one complete set of time samples is transmitted. The assumed waveform in transform theory is a continuously repetitive sequence. The minimum information is one cycle of this pattern. To increase the tolerance in timing, more than one complete symbol is transmitted. The additional data is called the guard interval.

With the addition of the guard interval, the initial timing accuracy only needs to ensure that the samples to be subject to the FFT are derived from one symbol. The longer the guard interval, the more rugged the system, but at a penalty of the power needed to transmit the guard interval. In practice, it is convenient to think of the transmitted symbol in two parts, the guard interval precedes the active symbol period which is so called because in a correctly aligned receiver, the FFT window is in that time slot. This gives ruggedness in the presence of echoes.

Once coarse synchronisation has been obtained, there is then the question on how to improve it. In DAB, there is a reference signal transmitted after the null symbol. Correlation of the reference symbol with the known transmitted signal provides the impulse response of the channel. From this, a much more accurate timing can be obtained. Similarly, the same signal also provides a measure of the frequency error of both the accuracy of the sampling frequency and the actual frequency of the signal (and hence allows accurate automatic frequency control to be implemented).

4.6 The guard interval: echoes and the Single Frequency Network

The guard interval provides an important property of the DAB system. It reduces impairment from echoes. If the delay of the echo is short compared with the symbol period, then energy in the echo from one symbol only corrupts the first part of the next. That part is the guard interval. In a correctly synchronised system, only data in the active symbol period is processed. The impairment to the this active symbol period comes from reflections derived from the same symbol. This is not inter-symbol interference, but a form of linear distortion. In the receiver the calculated phase of the symbol for each spectral component is distorted by the multipath signal, but every symbol is affected by the same perturbation. As we use differential data coding, the receiver looks at the difference in phase from one symbol to the next, and the errors cancel out.

Thus, provided that the echo has a delay less than the additional guard interval, there is no degradation to reception of the active symbol period. In practice, the length of the guard interval can be chosen to suit the broadcast, and so it is most unusual to find echoes with such a long delay that there is some real corruption of the signal.

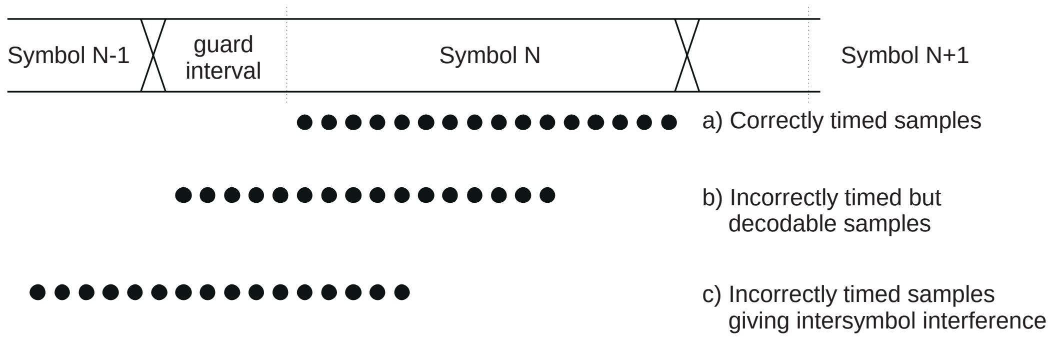

Figure 3 - Adding a guard interval to the signal.
With a guard interval included in the signal, the tolerance on timing the samples is considerably more relaxed.

The Single Frequency Network (SFN) is a major advantage of the Eureka 147 system. Because echoes from hills or buildings are indistinguishable from broadcast signals from another transmitter we can use the same frequency for all the transmitters in a network. Provided that the transmitters radiate the same signal at the same time, then the receiver will use all the available signals. Reception will be good, even in the overlap zones between the transmitters. This ability to re-use spectrum is a major saving over conventional systems. These require different frequencies in adjacent service areas. COFDM is, for a single channel, only as efficient as the underlying modulation system. However when planning a complete service, the Single Frequency Network (SFN) operation is a major advantage[4]. The spectral re-use leads to major spectrum efficiencies.

5. Frame structure

This is not the place to give a detailed description of the data formats in DAB. However, a few words about the framing are desirable to give an overview of the system.

The structure of the multiplex contains four main elements:

- the null symbol — an analogue synchronisation signal which gives receivers a method of rapid synchronisation
- the TFPR — a reference symbol which gives accurate frequency and timing signals
- the FIC — the fast information channel which gives the receiver all the data it needs to select services. To provide rapid access to this information, the data is not interleaved. This means that it is more vulnerable to errors. It is protected with a very strong rate 1/3 convolution code, and errors can be rectified because the information is repeated frequently.
- The main service channel — this is the main transport system in DAB. It contains all the audio services, and the majority of the data services. Its interleaving and error protection make it very rugged.

6. Practical experience

The Eureka 147 Digital Audio Broadcasting System is not just a laboratory system. The system has been proven by extensive testing[5] and a series of public demonstrations around the world. Single Frequency Network operation has been demonstrated from the outset.

The specification is openly available: it was agreed and ratified by the European Telecommunications Standards Institute [6], and adopted by the ITU as Digital System A.

It has been shown to work over a wide range of frequencies and for satellite as well as terrestrial transmission.

Its technology is now mature. This means that it is possible to make receivers which work, at a price a price which is attractive.

The proof of the technology can be seen in the UK and Sweden where there are national services already available. Many other countries in Europe have extensive experience of transmissions and reception. Outside Europe, all other countries who have tested the Eureka system have found that it works well.

Broadcasters and the public can therefore adopt the system with confidence.

7. Conclusion

In this short paper, it has only been possible to give an outline of COFDM*, the modulation system used for digital audio broadcasting. This modulation system has been proven and is being increasingly used for digital transmission in environments where multipath propagation can cause significant signal distortions. Its application for carrying a range of digital services will be described in the other papers in this conference.

8. Acknowledgements

The author would like to thank the many engineers in the Eureka 147 project who have contributed to his understanding of the system, and to the BBC for permission to publish this paper.

9. References

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* Further details can be found in reference [7].

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Also available as BBC Research Department Report 1996/8.

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