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Research and Development Report

THE COFDM MODULATION SYSTEM

The heart of Digital Audio Broadcasting

P. Shelswell, M.A., C.Eng., M.I.E.E.

Research and Development Department
Policy and Planning Directorate
THE BRITISH BROADCASTING CORPORATION

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Summary

Digital audio broadcasting offers the potential to give every radio in Europe the sound quality of a compact disc. To accomplish this, it requires a rugged method of transmission. The coded orthogonal frequency division multiplexing (COFDM) modulation system was developed to meet this need. This Report describes the reasons why a new modulation process was needed, and explains how the COFDM system has been optimised to meet the requirements.

Note that, in addition to the references embedded in the main text, further useful material can be obtained from other BBC R&D Report documents which are listed in the Appendix to this Report.

* This Report uses core material taken from the Institution of Electrical Engineers (IEE) publication: *Electronics & Communication Engineering Journal,* June 1995, and is published with permission. The article was written by the same author, and was entitled: *The COFDM system: The heart of digital audio broadcasting.*

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1. INTRODUCTION

Sound broadcasting history has recorded evidence of sustained technical improvement. The quality of sound reproduction has increased dramatically, receivers have become more convenient in size, and offer better performance. In the early years of broadcasting it was difficult to consider portable or mobile reception because receivers at that time were large, and were either mains powered or required sizeable batteries. People listened to programmes using amplitude modulation in the long, medium and short wavebands.

In the UK, FM broadcasting was introduced in VHF Band II during the 1950s, with stereo transmissions being added in the 1960s. This new broadcasting system was originally planned to serve household receivers using fixed, directional rooftop antennas; it was not planned for mobile reception. But nowadays, there is a large diversity of receivers in regular use – such as standard transistor portables and ghetto blasters; to which are added car and personal radios.

Thus there is a demand for something that was not originally part of the broadcast plan: mobile reception. The consequence of this demand for a wider-ranging usage, has been that receivers now have to cope with weaker signals than is desirable; the sound reproduction consequently bears the characteristic degradations caused by multipath propagation. Programmes are therefore liable to suffer from unwanted noises, typically characterised in car radios, for example, by sharp crackles and drop-outs (selective fading) and distorted sound (as a weak signal is interfered with by another signal). The latter problem is worsened by the ever-increasing demand for more services in an already-congested frequency band, leaving little room for expansion or improvement.

One solution to these problems would be to change to a digital transmission standard. But this is not of itself, enough. Simple digital signals, while being capable of producing very high-quality sound can be badly affected by multipath propagation. Merely applying error-protection is not sufficient; a major rethink of the digital transmission system itself is needed.

COFDM*, a new digital system which can provide rugged reception, even in the fading channel, has been developed to provide a suitable transmission standard for Digital Audio Broadcasting (DAB). The work on this system was initiated by CCETT[1,2]** in France, and was developed into a major new broadcasting standard by a European collaborative project, Eureka 147. The new DAB system provides good reception in a range of difficult conditions and is ideally suited to the application. Reference 3 provides an introductory review of the DAB system. The BBC started a new service of Digital Audio Broadcasting on 27 September 1995 in London, and is now (1996) expanding the service to cover other parts of the United Kingdom.

COFDM can be used for the transmission of any digital information, and variants of it are now being used for trial digital television transmissions. Consequently, although this Report uses the Eureka 147 system as a model, it will be sufficiently general to allow the theory to be applied to other systems. Only the modulation system itself will be dealt with in this Report, not the sound coding and multiplexing of the DAB signal.

2. THE PROBLEMS OF MULTIPATH PROPAGATION

The main problem that occurs with the reception of normal radio signals is that of fading caused by multipath propagation. This is not an isolated problem. Delayed signals are the result of reflections from fixed terrain features such as hills, trees or buildings, and moving objects like vehicles, aircraft and even people. Some of these reflections can be avoided by using a good antenna, but there is now a trend towards simple antennas on all radios, so there is no longer a realistic solution from this quarter – commercial markets require attractive, saleable products.

A characteristic of frequency selective fading is that some frequencies are enhanced whereas others are attenuated (Fig. 1 *(overleaf)*). When the receiver, and all the objects giving rise to the reflections, remain stationary, the effective frequency response of the channel from the transmitter to the receiver will be substantially fixed.

If the wanted signal is relatively narrowband, and falls into part of the frequency band with significant attenuation, then there will be flat fading and reception will be degraded.

* **C**oded **O**rthogonal **F**requency **D**ivision **M**ultiplex.

** Please note that further useful references have also been listed in an Appendix to this Report.

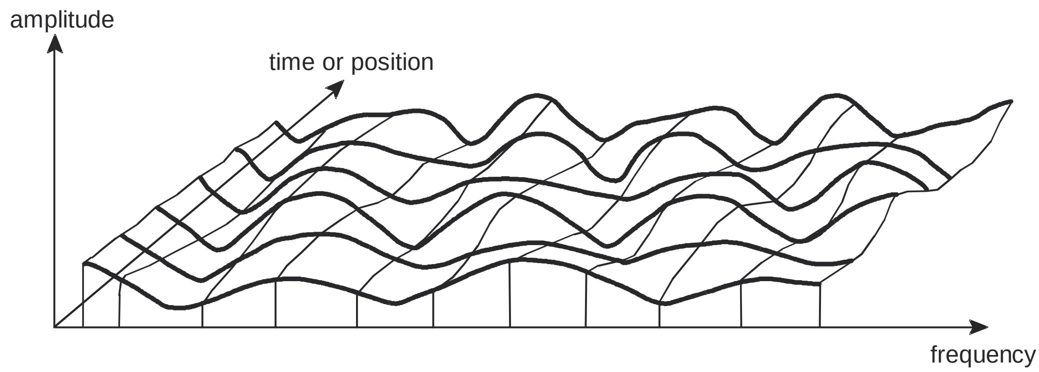

Fig. 1 - Typical frequency response of a channel suffering from multipath propagation.
The frequency response will vary with both time and position. Frequency selective attenuation is clearly present.

On the other hand, if there is some movement, either of the vehicle containing the receiver, or of any of the surroundings, then the relative lengths and attenuations of the various reception paths will change with time. A narrowband signal will vary in quality as the peaks and troughs of the frequency response move around in frequency. There will also be a noticeable variation in phase response, which will affect all systems using phase as a means of signalling. For FM reception, these channel distortions give rise to distortion of the sound, as well as the more obvious drop-outs as the signal level falls below the receiver threshold.

Now consider a signal which is of greater bandwidth. Some parts of the signal may suffer from constructive interference and be enhanced in level, whereas others may suffer from destructive interference and be attenuated, sometimes to the point of extinction. In general, frequency components close together will suffer variations in signal strength which are well correlated. Others that are further apart will be less well correlated. The correlation bandwidth is often used as a measure of this phenomenon. There is no standard definition of the correlation bandwidth; the alternative term ‘coherence bandwidth’ is sometimes used. One typical definition is the frequency separation of signals which are correlated by a factor of 0.9 or better. For a narrowband signal, distortion is usually minimised if the bandwidth is less than the correlation bandwidth of the channel. There is, however, a significant chance that the signal will be subject to severe attenuation on some occasions. A signal which occupies a wider bandwidth, greater than the correlation bandwidth, will be subject to more distortion, but will suffer less variation in total received power, even if it is subject to significant levels of multipath propagation.

Looking at the temporal response of the channel, a number of echoes may be seen to be present. There are many different types of echo environment which typify different geographical areas. In cities, echoes come from reflections from buildings; these can produce many separately identifiable echoes, with a large range of delays observable. However, in the countryside, the echoes are usually less distinct and have a smaller range of delay, especially if there are no nearby hills, although the presence of large hills and mountains can increase the range of observed delay[4].

This range of delay can be measured statistically. Different studies use the total range of delay, or the average delay. Whichever is chosen, the inverse of this leads to a good approximation for the correlation bandwidth[5].

As would be expected, therefore, measurements by the BBC have shown that the correlation bandwidth depends very much on the particular surroundings of the receiver site. Typical results in a built-up city show correlation bandwidths (using the 90% definition) of about 0.25 MHz in the VHF band. Changing the definition of the correlation bandwidth to reflect a correlation of 0.5 or more gives a bandwidth of over 1 MHz.

COFDM is a wideband modulation scheme which is specifically designed to cope with the problems of multipath reception. It achieves this by transmitting a large number of narrowband digital signals over a wide bandwidth.

3. A QUALITATIVE DESCRIPTION OF COFDM

3.1 The importance of Frequency Division Multiplex (FDM)

In COFDM, the data is divided between a large number of closely-spaced carriers. This accounts for the ‘frequency division multiplex’ part of the name COFDM. Only a small amount of the data is carried on each carrier, and this significantly reduces the influence of intersymbol interference (Fig. 2). In principle, many modulation schemes could be used to modulate the data at a low bit-rate onto each carrier[6]. In DAB, quadrature phase shift keying (QPSK)* is used, with differential encoding of the data at the transmitter and differential demodulation at the receiver.

* It is a minor detail, but the modulation system is actually $\pi/4$ D-QPSK. In this, the phases of the reference signals are increased by 45° each symbol period. In the rest of this Report, for simplicity, normal QPSK will be assumed.

It is an important part of the system design that the bandwidth occupied is greater than the correlation bandwidth of the fading channel. From the measurements that have been made, this means that the bandwidth occupied is ideally more than about 1 MHz. Then, although some of the carriers are degraded by multipath fading, the majority of the carriers should still be adequately received. A good understanding of the propagation statistics is needed to ensure that this condition is met.

3.2 The importance of coding

The distribution of the data over the many carriers means that selective fading will cause some bits to be received in error while others are received correctly. By using an error-correcting code, which adds extra data bits at the transmitter, it is possible to correct many or all of the bits which were incorrectly received. The information carried by one of the degraded carriers is corrected because other information, which is related to it by the error correction code, is transmitted in a different part of the multiplex (and, it is hoped, would not suffer the same deep fade). This accounts for the 'Coded' part of the name COFDM.

There are many types of error correcting code that could be used[2,7]. In the DAB system, the main channel code is a convolutional code, with a Viterbi receiver. As there is always a chance of residual errors after a Viterbi decoder, some of the higher priority data is pre-coded with a block code for additional security. These coding options have been specifically tailored to the audio and signalling data that is being broadcast, and would probably not be quite the same if COFDM were used for other than DAB purposes.

3.3 The importance of orthogonality

Finally, the 'Orthogonal' part of the COFDM name indicates that there is a precise mathematical relationship between the frequencies of the carriers in the system. In a normal FDM system, the many carriers are spaced apart in such a way that the signals can be received using conventional filters and demodulators. In such receivers, guard bands have to be introduced

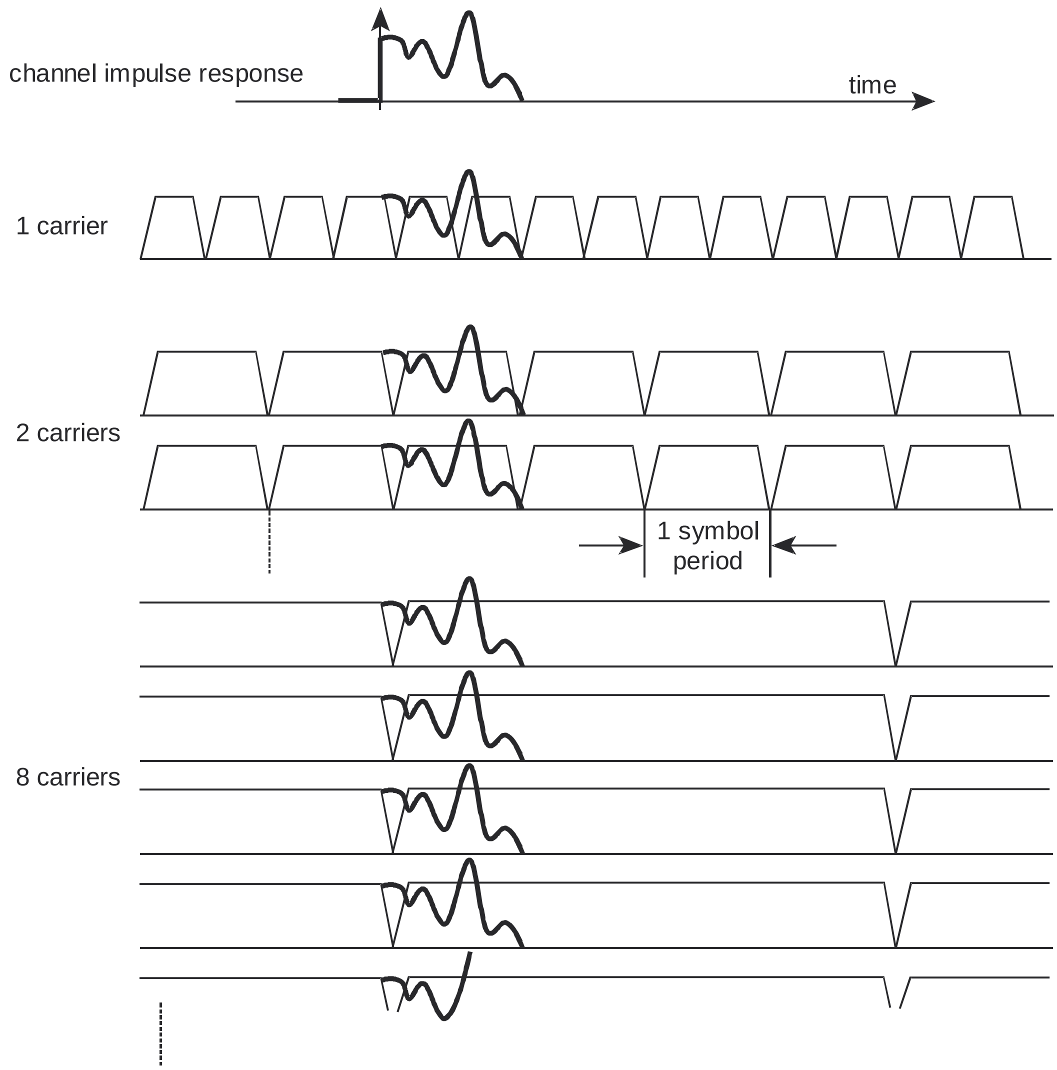

Fig. 2 - The effect of adopting a multicarrier system.
The intersymbol intereference affects a smaller percentage of each symbol as the number of carriers (and hence the symbol period) increases.

between the different carriers, and the introduction of these guard bands in the frequency domain results in a lowering of the spectrum efficiency.

It is possible, however, to arrange the carriers in a COFDM 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 receiver acts as a bank of demodulators, translating each carrier down to dc, the resulting signal then being integrated over a symbol period to recover the raw data. If the other carriers all beat down to frequencies which, in the time domain, have a whole number of cycles in the symbol period (τ), then the integration process results in zero contribution from all these other carriers. Thus, the carriers are linearly independent (i.e. orthogonal) if the carrier spacing is a multiple of $1/\tau$. This does mean, though, that all hardware must have linear characteristics. Any non-linearity causes inter-carrier interference (ICI) and therefore damages orthogonality.

Mathematically, suppose there is a set of signals ψ, where ψ_p is the pth element in the set. The signals are orthogonal if :

$$\begin{aligned} \int_a^b \Psi_p(t)\Psi_q^*(t)dt &= K \quad \text{for } p = q \\ &= 0 \quad \text{for } p \neq q \end{aligned} \qquad (1)$$

where the * indicates the complex conjugate. It is fairly simple to show, for example, that the series $\sin(mx)$ for $m = 1,2,....$ is orthogonal over the interval $-\pi$ to π.

Much of transform theory makes use of orthogonal series. Fourier series are a well known example of an orthogonal series, although they are by no means the only example.

3.4 Other features of COFDM

In the Eureka 147 system, some further refinements are adopted to ensure the maximum robustness:

- the data is interleaved, both in frequency and time. The error correction process works best if the errors in the incoming data are random. So the effects of burst errors must be minimised. To ensure this happens, the transmitted data is interleaved over all the carriers and over a range of time;

- the addition of a guard interval allows the system to cope with echoes of moderate duration, and with small inaccuracies in the receiver (for example, small timing errors). It is discussed in more detail below. See also, Reference 8 for a detailed analysis of OFDM frequency errors.

3.5 The sound coding associated with COFDM

When transmitting digitally-coded sound signals, it is particularly important to ensure high efficiency in the use of spectrum; bit-rate reduction of the sound is therefore used[9]. The choice of source coding for DAB is essentially independent of the choice of COFDM for the modulation scheme.

The Eureka 147 group has developed a system called MUSICAM, which has now been adopted by the International Standards Organisation in their ISO 11172-3 Layer II standard. This offers a variety of options, in which bit-rate can be traded for quality. For high quality stereo signals, a bit-rate between 192 and 256 kbit/s is needed, but the standard offers a range starting at 32 kbit/s and rising to 384 kbit/s.

In order to occupy sufficient bandwidth, to gain the advantages of the COFDM system, it is necessary to group a number of programmes together to form a wideband system.

4. MATHEMATICAL DESCRIPTION OF COFDM

After the qualitative description of the system it is valuable to discuss the mathematical definition of the modulation system. This shows how the signal is generated and how the receiver must operate, and also provides a tool for understanding the effects of imperfections in the transmission channel.

As noted above, COFDM transmits a large number of narrowband carriers, closely spaced in the frequency domain. In order to avoid a large number of modulators and filters at the transmitter, and complementary filters and demodulators at the receiver, it is desirable to be able to use modern digital signal processing techniques.

Mathematically, each carrier can be described as a complex wave:

$$s_c(t) = A_c(t)e^{j[\omega_c t + \phi_c(t)]} \qquad (2)$$

The real signal is the real part of $s_c(t)$. Both $A_c(t)$ and $\phi_c(t)$, the amplitude and phase of the carrier, can vary on a symbol by symbol basis. For QPSK, the amplitude is nominally unity and the phase takes one of the four quadrature phases of the conventional QPSK modulation system. For the pth symbol, over the time period $(p-1)\tau < t < p\tau$, $\phi_c(t)$ would take a value from the set 0°, 90°, 180°, 270°.

In COFDM there are many carriers. So the complex signal $s_s(t)$ (Figs. 3 and 4) is represented by:

$$s_s(t) = \frac{1}{N}\sum_{n=0}^{N-1} A_n(t)e^{j[\omega_n t + \phi_n(t)]} \qquad (3)$$

where:

$$\omega_n = \omega_0 + n\Delta\omega \qquad (4)$$

This is, of course, a continuous signal. If the waveforms of each component of the signal over one symbol period are considered, then the variables $A_c(t)$ and $\phi_c(t)$ take on fixed values which depend on the frequency of that particular carrier, and so can be rewritten:

$$\phi_c(t) \Rightarrow \phi_n \qquad (5)$$
$$A_c(t) \Rightarrow A_n$$

If the signal is sampled using a sampling frequency of $1/T$, then the resulting signal is represented by:

$$s_s(kT) = \frac{1}{N}\sum_{n=0}^{N-1} A_n e^{j[(\omega_0 + n\Delta\omega)kT + \phi_n]} \qquad (6)$$

At this point, the time over which the signal to N samples is analysed, has been restricted. It is convenient to sample over the period of one data symbol. Thus there is a relationship:

$$\tau = NT \qquad (7)$$

If Eqn. 6 is now simplified without loss of generality, by letting $\omega_0 = 0$, then the signal becomes:

$$s_s(kT) = \frac{1}{N}\sum_{n=0}^{N-1} A_n e^{j\phi_n} e^{j(n\Delta\omega)kT} \qquad (8)$$

Now Eqn. 8 can be compared with the general form of the inverse Fourier transform:

$$g(kT) = \frac{1}{N}\sum_{n=0}^{N-1} G\left(\frac{n}{NT}\right) e^{j2\pi nk/N} \qquad (9)$$

In Eqn. 8, the function $A_n e^{j\phi_n}$ is no more than a definition of the signal in the sampled frequency domain, and $s(kT)$ is the time domain representation. Eqns. 8 and 9 are equivalent if:

$$\Delta f = \frac{1}{NT} = \frac{1}{\tau} \qquad (10)$$

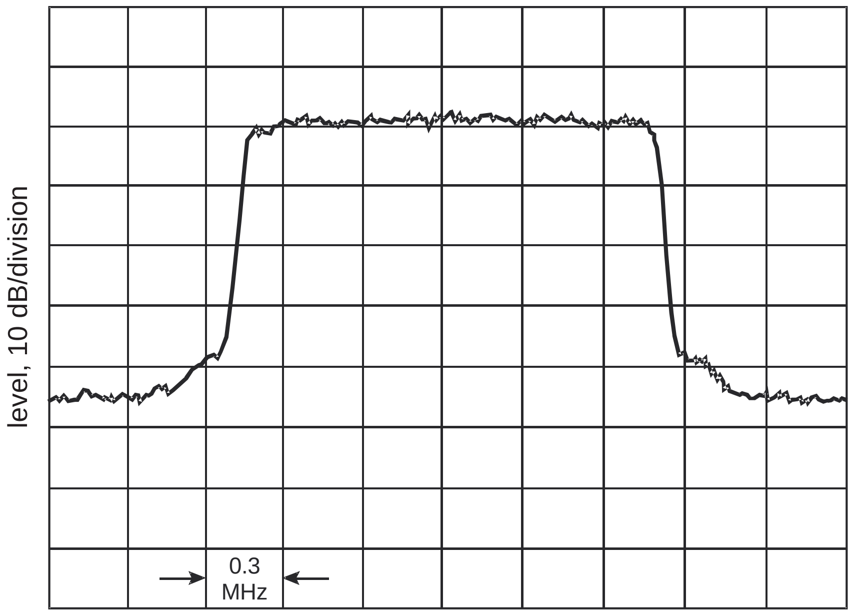

Fig. 3 - COFDM signal spectrum showing the full bandwidth.

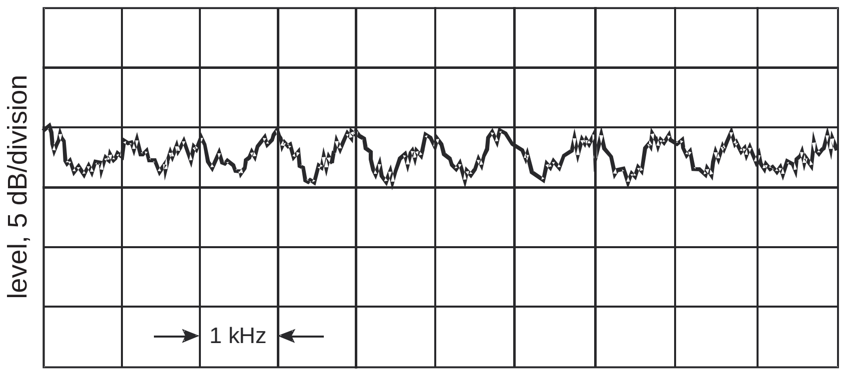

Fig. 4 - Detail of a COFDM signal.
The narrow spectrum of the individual QSPK carriers can be seen. Each signal has a spectrum which overlaps that of the nearby carriers.

This is the same condition that was required for orthogonality (see Section 3.3). Therefore, one consequence of maintaining orthogonality is that the COFDM signal can be defined by using Fourier transform procedures.

4.1 The Fourier transform

The Fourier transform relates events in the time domain to events in the frequency domain. There are several versions of the Fourier transform, and the choice of which one to use depends on the particular circumstances of the work.

The conventional transform relates continuous signals which are not limited in either the time or frequency domains. However, signal processing is made easier if the signals are sampled. Sampling of signals with an infinite spectrum causes a variety of problems because the sampling process leads to aliasing, and processing of signals that are not time limited can lead to problems with signal storage.

To avoid this, the majority of signal processing uses a version of the discrete Fourier transform (DFT). The DFT is a variant on the normal transform, where the signals are sampled in both the time and the frequency domains. By definition, the time waveform must repeat continually; this leads to a frequency spectrum which repeats continually in the frequency domain[10].

The fast Fourier transform (FFT) is merely a rapid mathematical method for calculating the DFT. It is the availability of this technique and technology that allows it to be implemented on integrated circuits at a reasonable price, that has permitted COFDM to be developed as far as it has. The rapid progress in the development of FFT chip sets has led to many exciting opportunities in this field.

The process of transforming from the time domain representation to the frequency domain uses the Fourier transform itself, whereas the reverse process uses the inverse Fourier transform.

4.2 The use of the Fast Fourier Transform (FFT) in COFDM

The main reason that the COFDM technique has taken so long to come to prominence has been a practical one. It has been difficult to generate such a signal, and even harder to receive and demodulate the signal. The *hardware* solution which makes use of multiple modulators and demodulators in parallel was somewhat impractical for use in a domestic system.

Now, the ability to define the signal in the frequency domain in *software*, and to generate the signal using the inverse Fourier transform, is the key to its current popularity. The use of the reverse process in the receiver is essential if cheap and reliable receivers are to be readily available. Although the original proposals were made some time ago[11], it has taken a period of time for technology to catch up*.

* Under the co-operative R&D programme JESSI (**J**oint **E**uropean **S**ubmicron **S**ilicon **I**nitiative).

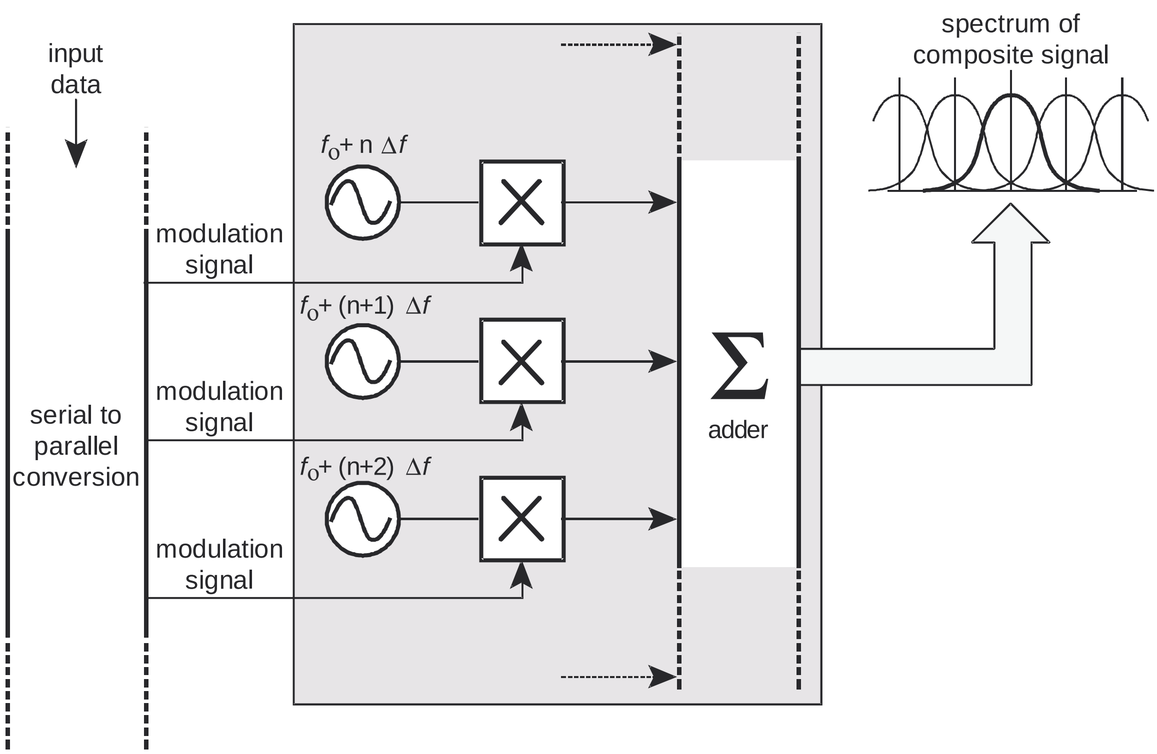

Fig. 5 - Generation of an OFDM signal.
Generation of the OFDM signal could use a large bank of oscillators and multipliers. In reality, the functions of the system in the shaded area are replaced by an inverse FFT chip.

At the transmitter, the signal is defined in software in the frequency domain. It is a sampled digital signal, and is defined in such a manner that the Fourier spectrum exists only at discrete frequencies. Each COFDM carrier corresponds to one element of this discrete Fourier spectrum. The amplitudes and phases of the carriers depend on the data to be transmitted. As each carrier is QPSK in this example, all the carrier amplitudes are unity, but this is not necessary in the more general case*. The phase of each carrier is defined for each transmitted symbol. All the carriers have their data transitions synchronised, and can be processed together, symbol by symbol. Fig. 5 is a schematic diagram of the transmitter signal processing.

Using VLSI, it is possible to carry out the inverse FFT. This provides a series of digital samples which are the time domain representation of the signal. These samples can be applied to a conventional digital-to-analogue converter (DAC) to give the real electrical signal.

To enable the signal to be generated using an inverse FFT, it is preferable that the number of carriers considered in the calculation is a power of two. In practice, it is not always desirable to have the number of real carriers restricted in this way. However, it is convenient to make up the actual number chosen to a power of two by setting the amplitudes of those not wanted to zero. This feature also simplifies the design of the anti alias filter after the DAC.

In the receiver, the reverse process is applied. Assuming the receiver has some means of synchronisation, then the signal is converted from analogue format to a sampled digital representation. The samples corresponding to each symbol are then Fourier transformed to the frequency domain. This gives the amplitude and phase of each transmitted carrier.

In the Eureka 147 system, it is the change in phase of each carrier from one symbol to the next which communicates the information.

4.3 Orthogonality

A natural consequence of this method is that it generates carriers that are orthogonal. The members of an orthogonal set are linearly independent.

Consider that the set of transmitted carriers, ψ, is an orthogonal set, such that:

$$\Psi_k(t) = \exp(j\omega_k t) \qquad (11)$$

$$\omega_k = \omega_0 + 2\pi \frac{k}{t}$$

* Digital terrestrial television uses **Q**uadrature ***A**mplitude* **M**odulation (QAM), for example

If this is truly orthogonal, then the orthogonality relationship in Eqn. 1 should hold, that is:

$$\int_a^b \Psi_p(t)\Psi_q^*(t)dt = \int_a^b e^{j[2\pi(p-q)t/\tau]}dt$$

$$= (b-a) \text{ for } p = q$$

$$= \frac{e^{j[2\pi(p-q)b/\tau]} - e^{j[2\pi(p-q)/\tau]}}{j2\pi(p-q)/\tau}$$

$$= \frac{e^{j[2\pi(p-q)b/\tau]}\left[1 - e^{j[2\pi(p-q)(a-b)/\tau]}\right]}{j2\pi(p-q)/\tau}$$

$$= 0 \quad \text{for } p \neq q \text{ and } (b-a) = \tau$$

$$\text{(remember that } p \text{ and } q \text{ are integers)} \qquad (12)$$

The carriers, consequently separated in frequency by $1/\tau$, meet the requirements of orthogonality provided that they are correlated over a period τ. This is the formal derivation of the result quoted earlier.

If the analysis is extended to include the phase of each carrier, then this is recovered by the process defined in Eqn. 12. This is exactly the computation that is needed in the receiver.

The method can be extended to show the effect of frequency and timing errors in the system, as shown in Reference 8.

4.4 Synchronisation and the guard interval

In the receiver, it is necessary to sample the incoming signal and perform a transform to recover the carriers. If the system is fully locked up, and the sampling frequency is correct and in the right phase, there is no problem. However, this ideal case will be difficult to achieve, especially when the receiver is first switched on. There is therefore a need for acquiring a timing lock.

In the DAB system, coarse synchronisation is provided by a simple technique: all the carriers are switched off on a regular basis. By using a simple amplitude detection circuit, it is possible to generate an approximation to the timing. However, the timing will not be perfect, and all of the samples could be displaced by a fixed time offset, giving rise to intersymbol interference[8].

In DFT transform theory, the assumed waveform is a continuously repetitive sequence, rather like wallpaper patterns repeating across a wall. The minimum information required is one cycle of this pattern. To avoid the timing problems, more than one complete symbol is transmitted, the part of the symbol that is repeated has been named the ‘guard interval’.

With the guard interval, the initial timing accuracy only needs to ensure that the samples are taken from one symbol. The longer the guard interval, the more rugged the system, but with the penalty of needing more power to transmit a longer guard interval. There is obviously a compromise to be reached; this will be dealt with later in the Report. In practice, it is convenient to think of the transmitted symbol in two parts:

- the guard interval,
- the following active symbol

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 matter of how to improve on it. After the null symbol, a reference signal is transmitted. This a known digital sequence. Correlation of the received version of this 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 reference signal can be processed to provide a measure of the frequency errors in both the local oscillator and the sampling clock circuits. Reference 8 gives an indication of the effects of imperfections in frequency acquisition.

The other important purpose of the guard interval, is to minimise the effect of echoes. If the echo is short compared with the total symbol period, then any energy conveyed from one symbol to the next by the echo only degrades the guard interval. The active symbol period contains direct energy and reflections whose delay is less than the guard interval, and is therefore derived from the same symbol period. 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. If the channel is not changing rapidly then successive symbols of any one carrier will be perturbed in a *similar* manner. If differential coding is used, the receiver looks at the *difference* in phase from one symbol to the next, and the errors cancel out.

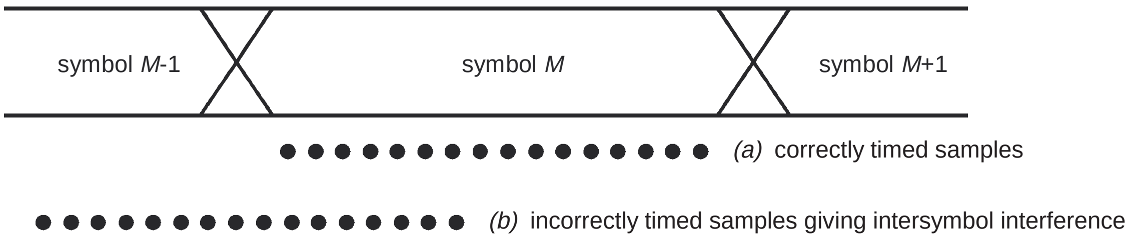

Fig. 6 - The effect of timing errors on reception of a signal.
The signal must be decoded by sampling within the symbol period and then performing a Fast Fourier Transform to calculate the phases of the individual carriers. With a timing error, there can be considerable intersymbol interference.

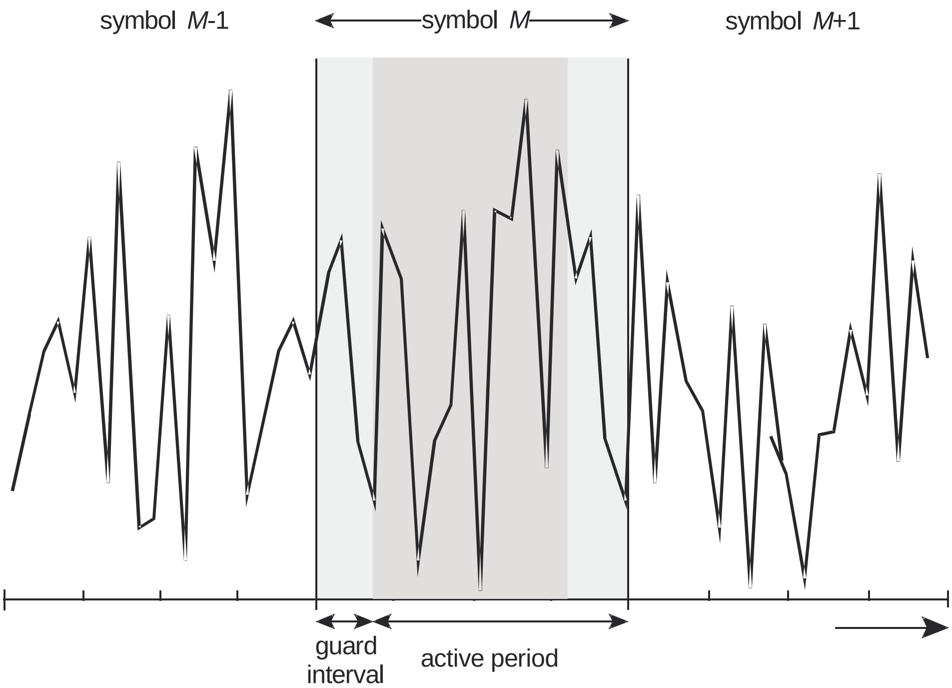

Fig. 7 - Example of the guard interval.
Each symbol is made up of two parts. The whole signal is contained in the active symbol. The last part of the active symbol (shown light grey) is repeated at the start of the symbol (also shown as light grey) and is called the guard interval.

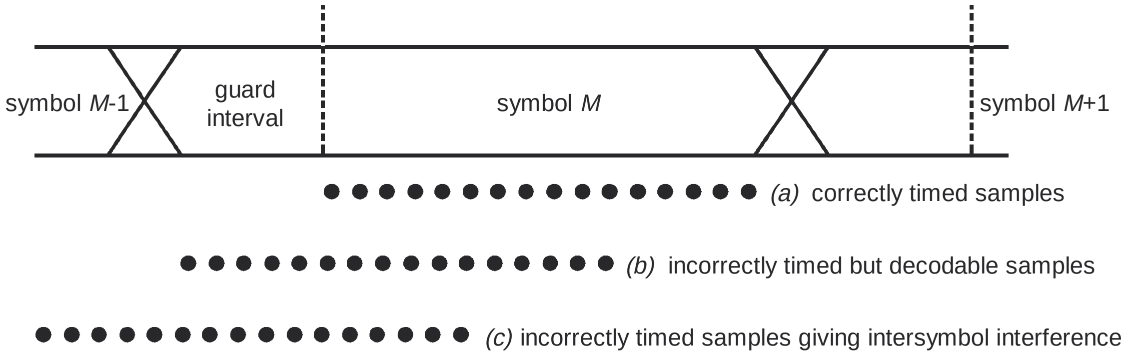

Fig. 8 - The effect of adding a guard interval on the timing tolerance.
With a guard interval included in the signal, the tolerance on timing the samples is considerably more relaxed.

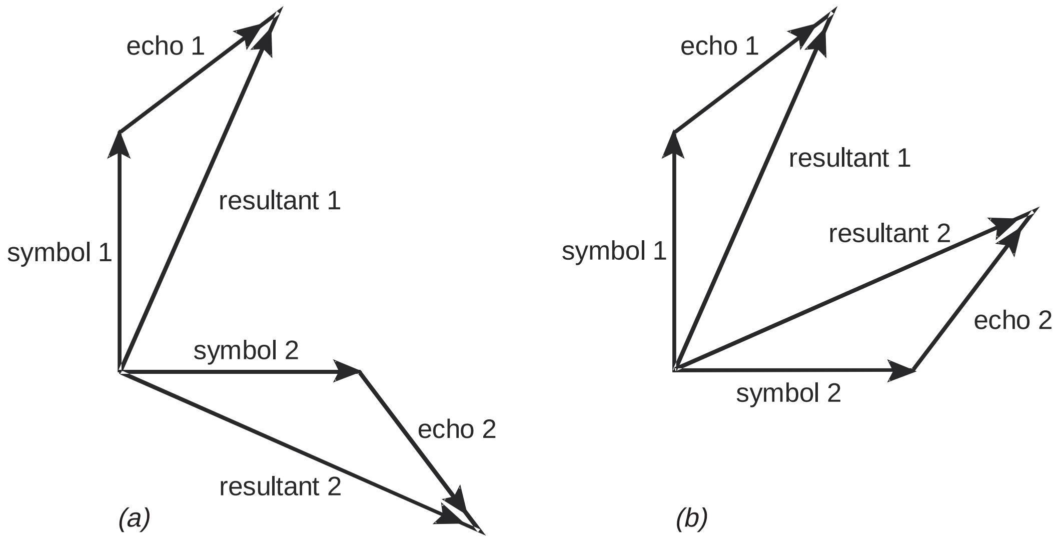

Fig. 9 - Vector diagram of the distortion caused to a signal in the presence of a single echo.
Here, two adjacent symbols are depicted, both affected by a single echo. Arbitrarily, the QPSK signal has been coded such that there is a 90° phase shift from one symbol to the next. When there are echoes present, the received phase is distorted. In the first example, the symbol and the echo have a fixed relationship. This occurs when the echo contains energy which comes exclusively from the same symbol period. An example, would be if the samples for the signal are timed as in Fig. 8(a) and (b). The phase relationship between the two resultant symbols is therefore maintained. As differential decoding is used, the constant phase shift is of no consequence.When echoes are delayed by more than the guard interval, as is the case in Fig. 8(c), matters are different. The phase of the signal in the previous symbol does not have a fixed relationship with the current symbol, and so the relationship between the phases of the two resultant symbols changes, and this distortion could degrade the signal.

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.

It does not matter whether the echo is passive, such as one from a hill or building, or active, such as one from another transmitter. This is a major feature which allows DAB to use the same radio frequency from all of its transmitters in the network – a Single Frequency Network (SFN). Provided that the transmitters radiate the *same signal* at the *same time*, then reception will be good, even in the overlap zones between the transmitters. This re-use of the spectrum produces a major saving over conventional systems which would need different frequencies in adjacent service areas. COFDM is, for a single channel, only as efficient as the underlying modulation system. However, when the planning of a complete service is provided, the SFN operation is a major advantage[12]; the spectral re-use leads to major spectrum efficiencies. See Section 5 for details.

5. THE APPLICATION OF COFDM TO DAB

Let the Eureka 147 DAB system be used as an example of a COFDM system and consider the choice of parameters:

Firstly, there is the question of what bandwidth to use for the COFDM system. The wider the bandwidth, the more likely that the system exceeds the correlation bandwidth of the channel. Short delay echoes are the main problem to overcome, and as these are always present, there is no hard bound; but the narrower the bandwidth, the more likely it is that the whole signal will be affected. There is a trade-off between bandwidth and transmitter power.

The original experiments were carried out with a bandwidth of about 7 MHz, and showed few problems[13]. Then the bandwidth was successively reduced to below 2 MHz; at which point, there is a degradation equivalent to about 1 dB in performance. This is not a large amount, but the degradation starts to increase quite rapidly when the bandwidth is reduced below 1.5 MHz. If the bandwidth is reduced to the 200 kHz used for FM sound, then the margin required would be an additional 6 dB or so. So a figure of about 1.5 MHz for the bandwidth of the system is a good compromise for the type of propagation conditions that apply to mobile and portable radio reception.

One of the parameters that is directly affected by the bandwidth is the available bit-rate. The modulation system on each carrier is QPSK. The carriers are approximately separated in frequency by the inverse of their symbol period. Consequently, the maximum bit-rate available is 2 bit/s/Hz of bandwidth. This figure is reduced by the inefficiency (signal redundancy) of the guard interval, the null symbol and the error coding. For DAB, this brings the useful bit-rate down to about 1 bit/s per Hertz of bandwidth.

Therefore, a DAB system will provide just under 1.5 Mbit/s of useful data. This is considerably more than the 256 kbit/s that is needed for a high-quality stereophonic programme, so the implication is that *several* broadcast programmes will share the same multiplex.

Secondly, consider the number of carriers. The more there are, the greater is the resolution of the diversity offered by the system. There is, however, a relationship between the symbol period and the carrier spacing. The carrier spacing is $1/\tau$. For the differential demodulation to work properly, the multipath environment must change slowly from symbol to symbol. Thus, there is a limit to the symbol period and hence the number of carriers. For static reception, this is not a major problem. But for mobile reception, the motion of the vehicle leads to changes in the multipath environment. Over a symbol period, a vehicle moving at a velocity v m/s will travel $v\tau \times f/c$ wavelengths. This is $f_d\tau$ wavelengths, where f_d is the maximum Doppler shift. If this is to introduce negligible phase distortion, then the function $f_d\tau$ must be small. A figure of $f_d\tau < 0.02$ has been proposed as suitable for general use[14]. The Eureka 147 system has been specified so that reception is not significantly impaired unless vehicles are travelling in excess of 100 mph (160 km/h). However, to achieve this, it has been necessary to adopt three different modes of operation, each mode being suited to a different part of the VHF and UHF frequency bands. The main difference between the modes is the symbol period and, as a direct consequence, the number of carriers.

If it were only a question of single transmitters, then the significant echoes would all be relatively short. Surveys indicate that a guard interval of the order of 10 µs would be satisfactory for the majority of locations, in the UK at least.

The use of several transmitters puts a limit on the minimum guard interval that should be used. The transmissions from areas that are some distance away can reach quite high levels on occasions of anomalous propagation. On the days when the weather forecast states that television reception may be subject to co-channel interference, there is also a strong likelihood that similar interference from remote DAB transmitters may also cause a problem. It is possible to plan a service without worrying too much about this if there is sufficient signal from the local transmitters. Then it becomes a simple matter of deciding the spacings of the main transmitters. This has to be a compromise between a small number of high power transmitters spaced by about 50 km, or a much larger number of lower-power transmitters. Because the first option is likely to be cheapest, the guard interval is set to about 250 µs – equivalent to a maximum difference of about 80 km in transmission distance at the receiver.

The symbol period *need not* be directly related to the guard interval. It is just a question of how much of the symbol period is repeated in the guard interval. This is purely a matter of efficiency, as the power transmitted in the guard interval does not form a useful part of the information in the receiver unless there are substantial echoes. To minimise the power loss by this mechanism, it is desirable to keep the guard interval to as low a percentage as possible of the symbol period. In practice, a guard interval of the order of 25% of the symbol period has been found to be a good compromise.

This leads to a symbol period of 1 ms, and, using Eqn. 10, leads to a carrier spacing of about 1 kHz. It means that approximately 1500 carriers are accommodated in the minimum bandwidth that would be desirable for one COFDM transmission.

In Section 4.4, it is stated that the DAB system is spectrum efficient. Within the multiplex, about one programme per 250 kHz of bandwidth is being achieved. It is possible to transmit the same signal throughout the full extent of any area requiring the same set of programmes, thus offering four programmes per megahertz in that area. To permit separate regional or national requirements, by the map-colouring theorem, four frequency blocks should be adequate to provide one single-frequency network (SFN) in each area. This leads to an overall spectrum requirement of 1 MHz per stereo programme to provide flexibility of programmes worldwide.

By comparison, at least 2.2 MHz is needed to provide VHF FM programmes in the UK. As the UK is surrounded by water, the planning problem is somewhat easier than in continental Europe, where the land-locked countries need a spectrum closer to 3.3 MHz. Optimistic estimates to provide separate services of conventional *digital* radio, for example using the NICAM system used for television sound, indicate that the minimum required spectrum would be

3.4 MHz per programme. In practice, a figure closer to 10 MHz per programme is thought to be more appropriate for NICAM. So it becomes apparent that the SFN approach using COFDM offers significant economies in spectrum.

When the signal is generated, it is defined in the frequency domain, and is then transformed into its time domain representation. Sampled digital signals of course suffer from aliasing. In an operational system, the spectral repeats are not wanted, so they must be filtered off. To make the filtering easier, the spectrum is not defined such that all the available carriers are used. The outer carriers are set to zero. This leaves a gap in the spectrum that can be used to minimise the complexity of the output filter.

When deciding exactly how many carriers are to be used and how many processed, the decision is guided by the fact that the FFT works most effectively if the number of samples in either domain is a power of two. So for DAB, the number of carriers defined is 1536, and the processing is based on 2048 carriers in the system. When DAB was initially being developed, the need to include a 2048 point FFT in the receiver was seen as a disadvantage. Now, however, technology has advanced to the point where higher order FFTs are being proposed for digital television applications.

6. OTHER SYSTEMS

The DAB system discussed here, as defined by the Eureka 147 partners, is not the only COFDM system.

The Eureka group members have other options; these are tailored for use with DAB systems operating in the top end of the UHF range, and have transmission by satellite as a major objective.

If the COFDM system is used for fixed reception, as would usually be the case with *television* for example, then many of the problems with multipath propagation are less severe. There is less variation in the channel, both because the antenna is capable of being selective, and because the channel conditions are naturally not as variable as would be the case in a moving vehicle. Consequently, it is possible to use shorter guard intervals, or even no guard interval at all. However, if an SFN is to be considered, an appropriately large guard interval would clearly be necessary.

The system used to modulate the individual carriers does not need to be QPSK . It could be a higher order system (as is required for television transmissions, for example) such as 16, 64 or even 256 QAM. Indeed, a mixture of modulation systems is possible, so long as they are orthogonal. This restriction is usually met if the symbol periods are the same. The use of coherent demodulation will give better performance if the channel is not varying too much. For mobile reception of DAB, the channel response may vary rapidly in phase, and so the potential benefits of coherent demodulation are lost in the implementation. For the more static channel, these benefits can be realised, and so it is useful to adopt the technique. The main challenge of using these higher order systems is to be able to cope with the changing environment, so synchronisation and equalisation strategies play a key part in their design.

7. CONCLUSIONS

Now that the Eureka 147 DAB system has been fully defined and accepted as a European Standard,[15] many broadcasters are preparing to start services. The BBC started a public DAB service in September 1995 in the London area. This is expanding to cover the major UK cities and motorways, providing coverage of 60% of the population by Spring 1998. Although receivers are somewhat more expensive than would be desirable in an established market, the rapid pace of development means that their availability will greatly improve, with the price being expected to drop rapidly in the near future.

The system has proved to give extremely rugged reception, in a wide range of differing environments. It really does offer the listener a marked improvement over FM in the portable and mobile environments.

8. ACKNOWLEDGMENTS

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APPENDIX

Reports on DAB published by BBC Research & Development Department

Report Title and Author(s)	Reference No.
• Digital Audio Broadcasting: The first UK field trial *(Peter Shelswell, Colin Bell, Jon Stott, Sue Wakeling, Mark Maddocks, John Moore, Peter Durrant, Richard Rudd)*	**BBC RD 1991/2**
• Building penetration loss, measurements for DAB signals at 211 MHz *(John Green)*	**BBC RD 1992/14**
• Digital sound: Subjective tests on commentary-quality codecs *(Neil Gilchrist, A. Oxenham)*	**BBC RD 1993/2**
• Coverage aspects of a single frequency network designed for digital audio broadcasting *(Colin Bell, Bill Williams)*	**BBC RD 1993/3**
• Dynamic range control of audio signals by digital signal processing *(Neil Gilchrist)*	**BBC RD 1993/7**
• Digital Audio Broadcasting: Comparison of coverage at different frequencies and with bandwidths *(Mark Maddocks, Ian Pullen)*	**BBC RD 1993/11**
• DAB: Multiplex and system support features *(John Riley)*	**BBC RD 1994/9**
• DRACULA: Dynamic range control for broadcasting and other applications *(Neil Gilchrist)*	**BBC RD 1994/13**
• Digital Audio Broadcasting: Comparison of coverage at Band II and Band III *(Ian Pullen, Phil Doherty, Mark Maddocks)*	**BBC RD 1994/16**
• Eureka 147: Tests of the error performance of the DAB system *(Neil Gilchrist)*	**BBC RD 1994/18**
• Digital Audio Broadcasting: Measuring techniques and coverage performance for a medium power VHF single frequency network *(Mark Maddocks, Ian Pullen, John Green)*	**BBC RD 1995/2**
• Experimental satellite broadcast of Eureka 147 DAB from Solidaridad 2 *(John Zubrzycki, Richard Evans, Pete Shelswell, M.A. Gama, M. Gutierrez)*	**BBC RD 1996/5**

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