Digital Radio Guide

WBU Technical Committee

World Broadcasting Unions

FOREWORD

The purpose of the Digital Radio Guide is to help engineers and managers in the radio broadcast community understand various aspects of digital radio systems that are available in 2006. The guide covers those systems used for transmission in different media, but not in the production chain. The in-depth technical descriptions of the systems are available from the proponent organisations and their websites listed in the appendices. The choice of the appropriate system remains the responsibility of the broadcaster who should take into account the various technical, commercial and legal factors relevant to the application.

It is my sincere hope that the publication will be a useful tool for radio broadcasters to evaluate the various options available to them.

I would like to thank the editorial team for the excellent job they did in preparing this revised edition of the Digital Radio Guide. The team was chaired by Wayne Heads, ABU Technical Director, and consisted of Franc Kozamernik and David Wood, EBU, and Mike Starling, NABA.

We are grateful to the many organisations and consortia whose systems and services are featured in the guide for providing the updates for this latest edition. In particular, our thanks go to the following organisations:

- European Broadcasting Union
- North American Broadcasters Association
- Digital Radio Mondiale
- iBiquity Digital
- WorldDAB Forum
- WorldSpace Inc

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TABLE OF CONTENTS

1 Introduction

Digital technology has steadily transformed the way in which programmes are made and distributed in recent years. Already many broadcasters have invested in digital systems for contribution and production and now the switch from analogue to digital is moving along the broadcasting chain into transmission. At the same time, digital developments are drawing together the broadcasting, telecommunications and computer industries in a process of convergence. For all broadcasters, this is leading to a new and challenging business environment in which they are searching for a clear ‘multimedia’ role.

Although similar changes are happening in both radio and television, this guide deals with radio. It is designed to help managers, including those in developing countries, identify the technical and business forces that are driving the analogue to digital conversion process. There are many benefits that radio broadcasters stand to gain by adopting digital technology and the current interest in digital television should help and encourage the switch from analogue to digital in radio broadcasting. The issue is likely to be brought into sharper focus if and when individual countries or regional groups set timetables for phasing out existing analogue services.

This updated Digital Radio Guide focuses primarily on the various digital radio systems in operation today and their associated standards. The guide visits not only terrestrially based digital system but also overviews the services now available via satellite radio.

The important development seen in this updated guide is the significant changes to digital radio development compared to the original guide published in 1998. The first guide presented many options for the US-based studies into digital radio as well as satellite radio. These systems have now matured to the level that there is unlikely to be changes in the choice for digital standards for many years. The only development planned at present is that by the DRM Consortium with its DRM120 project.

This guide is a compilation of inputs provided by WBU members for the benefit of the world broadcasting community. Note that references to relevant worldwide websites and a glossary of acronyms are provided in Appendices B and C at the end of this guide.

2 What is Digital Radio?

Since the early days of broadcasting, analogue systems have been used to carry programmes from the studios to the listeners. Now, due to the growing number of broadcasters and programme services, the frequency bands allocated to AM and FM radio in many regions of the world are full. The resulting congestion in the radio spectrum has led to a decline in reception quality and is a real constraint to further growth. Furthermore, in densely populated areas, FM reception on car radios and portables can be very poor. This is due to the effect of severe multipath propagation caused by signal reflections and shadowing due to high buildings.

Digital transmission technology can offer much improved coverage and availability. It is expected to replace analogue transmissions in many areas, but as digital systems are incompatible with current AM and FM broadcasting systems, new receivers will be needed.

In basic form, digital radio is an application of the technology in which sound is processed and transmitted as a stream of binary digits. The principle of using digital technology for audio transmission is not new, but early systems used for terrestrial television sound (such as NICAM 728) need considerable bandwidth and use the RF spectrum inefficiently, by comparison with today's digital systems.

The development of digital radio has been helped by the rapid progress that has been made in digital coding techniques used in RF and audio systems. This has led to improved spectrum efficiency, more channel capacity, or a combination of these benefits. Digital compression techniques used in audio systems have improved sound quality at low bit rates to the extent that radio broadcasts can be made on location and then transmitted to the broadcaster's production studios over telephone circuits in high quality.

Ideally, to reach the widest range of listeners, a genuinely universal digital radio system should be capable of being transmitted via terrestrial, satellite and cable systems.

There are new digital radio systems in operation. The list is set out in Table 2.1.

The table illustrates the wide spread of operational systems throughout the world.

The great strength of the present analogue transmission systems is the world-wide standardisation on just two systems (FM and AM). This enables listeners to use one radio to receive programmes at any location. But in the development of digital systems, it is now clear that similar standardisation will not be so easily achieved. Differing market requirements are driving digital systems to be more specialised and tailored to meet regional, national, or application-oriented needs. Furthermore, the complexity of digital systems compared to existing analogue techniques fosters this differentiation.

Table 2.1. Digital Radio Systems

	AVAILABILITY	
SYSTEM	**Terrestrial in service date**	**Satellite in service date**
Eureka 147 (ITU-R Digital System A)	1995 (for the UK, Norway, Denmark and Sweden)	---
DRM - Digital Radio Mondiale ETSI ES 201 980 V1.2.2 (2003-4) International consortium	Transmissions tests successfully since 2000; regular broadcasting from July 2003. For use in all broadcasting bands below 30 MHz	---
DRM - Digital Radio Mondiale	2010 DRM+	
HD Radio (iBiquity Digital) (FCC Docket 99-325, NRSC-5 Standard) in the HF and MF Bands	Now rolling out in US	---
WorldSpace (ITU-R Digital System D)		1998
XM Radio		2001 (North America)
Sirius Satellite Radio		2000 (North America)
Digital Radio Broadcasting ISDB-TSB (Japan)	(1)	---

Notes:

--- Not applicable

(1) System under trial development. No date set for a service.

3 Why Digital Radio?

The existing AM and FM analogue systems suffer from inherent short-comings and neither can offer uniform reception quality throughout the coverage area. AM radio reception is constrained by bandwidth limitations, which restrict the audio quality and by interference from other co-channel and adjacent channel transmissions. This is particularly troublesome during the hours of darkness. The start of FM services in the 1950's improved the audio bandwidth and overcame the night-time interference, but the broadcasts were designed to be received using fixed receivers with external antennas. When listened to in vehicles or on portables, reception suffered from the effects of reflected signals (multipath) and other forms of interference, particularly in suburban and city areas.

Another aspect of AM and FM analogue transmissions is the inefficient use of the spectrum (relative to what is possible using digital technology). As pressure on the radio spectrum rises, this finite resource becomes more scarce. Digital radio is seen by some administrations as a potential source of income and spectrum, as a way to encourage the resource to be used more efficiently.

There are many ways in which digital radio systems can improve upon analogue systems:

- Digital signals are more robust than analogue and can be transmitted successfully at lower transmitter powers.
- Digital systems using coded multicarrier modulation offer much improved reception on mobile car radios and portables.
- Advanced digital compression techniques enable low bit rates to be used successfully, whilst still producing sound of near CD quality. This makes digital systems more spectrum efficient.
- The digital bit-stream can be used for transmitting both audio and data.
- A digital radio is much easier to use/tune than is an AM/FM radio.
- There is increasing competition for the public's time from the non-broadcast media such as the CD. By comparison, many AM (in particular) and FM services offer poor audio quality.
- The data capability of digital radio can be used directly or, with some modification, for other related broadcasting activities such as Internet radio.

4 Terrestrial Transmission Systems

This section provides a technical overview of the various digital radio systems available for terrestrial application: DRM, DAB, ISDB-TSB, and HD Radio. These systems operate in various frequency bands and offer different attributes and features.

4.1 DRM – Digital Radio Mondiale

The DRM system encompasses a high level of flexibility in its design. These are noted in this subsection in the signal flow sequence going from the delivery from a program studio or network control centre to a transmission site and on to reception and decoding in a receiver.

4.1.1 Key Features of the System Design for the Markets to be Served by the DRM System

The DRM system is a flexible digital sound broadcasting system for use in the terrestrial broadcasting bands below 30 MHz.

It is important to recognize that the consumer radio receiver of the near future will need to be capable of decoding any or all of several terrestrial transmissions; that is narrow-band digital (for <30 MHz RF), wider band digital (for >30 MHz RF), and analogue for the LF, MF, HF and VHF (including the FM) bands. In addition there is the possibility of satellite delivery reception in the L- and S-bands. The DRM system will be an important component within the receiver. It is unlikely that a consumer radio designed to receive terrestrial transmissions in the near future with a digital capability would exclude the analogue capability.

In the consumer radio receiver, the DRM system will provide the capability to receive digital radio (sound, program related data, other data, and still pictures) in all the broadcasting bands below 30 MHz. It can function in an independent manner, but, as stated above, will more likely be part of a more comprehensive receiver – much like the majority of today's receivers that include AM and FM analogue reception capability.

The DRM system can be used in either 9 or 10 kHz channels, or multiples of these channel bandwidths. Differences on how much of the total available bit stream for these channels is used for audio, for error protection and correction, and for data transfer depend on the allocated band (LF, MF or HF) and on the intended use (for example, ground wave, short distance sky wave or long distance sky wave, with a data application service or without one). In other words, there are modal trade-offs available so that the system can match the needs of broadcasters worldwide.

As noted in more detail in subsequent parts of this subsection, the DRM system has the following structure. It employs advanced audio coding (AAC), supplemented by spectral band replication (SBR), as the main digital audio encoding. These are parts of the MPEG-4 audio standard. SBR significantly improves perceived audio quality so that the overall audio quality of a DRM signal is similar to that of FM (mono). Orthogonal Frequency Division Multiplexing (OFDM) and Quadrature Amplitude Modulation (QAM) are used for the channel coding and modulation, along with time interleaving and forward error correction (FEC). Pilot reference symbols are injected to permit a receiver to "equalize" the channel by comparing a known stored bit sequence with the corresponding received sequence of these special bits, and adjusting accordingly if there are differences in the received compared to the stored sequence.

The combination of these techniques results in high quality sound in a narrow channel with robust reception in an intended coverage area with relatively low transmission power. In addition, source coding schemes using lower bit rates than that used with AAC/SBR are included for lesser levels of audio quality if the AAC/SBR quality level is not desired by a broadcaster. For example, a broadcaster may want to transmit two or more "speech" only programs. These would not require the full performance of AAC/SBR.

4.1.2 Brief Description of the DRM System

(1) Overall design

Figure 4.1: Transmission Block Diagram

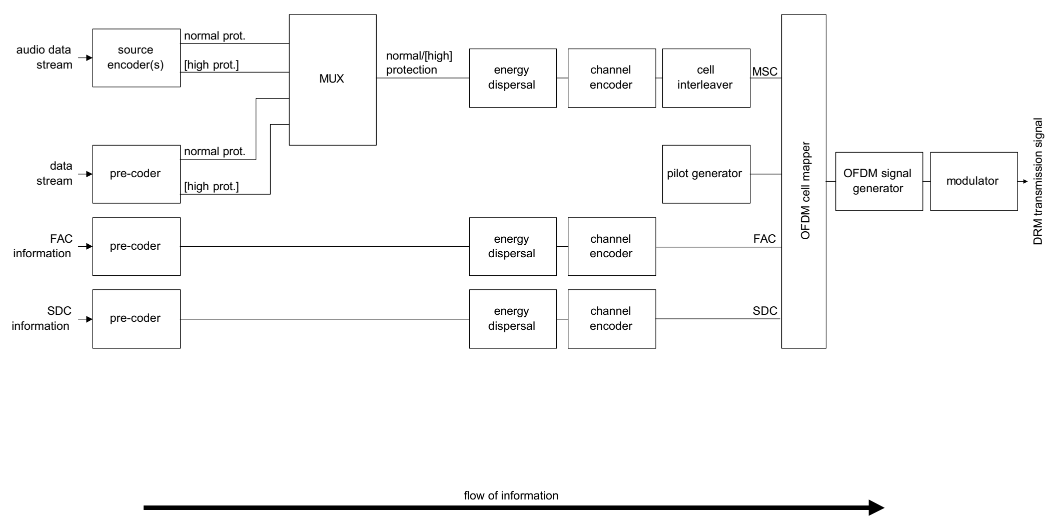


Figure 4.1 depicts the general flow of different classes of information (audio, data, etc.) after their origination in a studio or control centre (that would be depicted to the left of the figure) to a DRM transmitter exciter/modulator on the right. Although a receiver diagram is not included in the figure, it would represent the inverse of this diagram.

There are two classes of basic information:

- the encoded audio and data that are combined in the main service multiplexer;
- information that bypasses the multiplexer that are known as fast access channel (FAC) and service description channel (SDC), whose purposes relate to identification and control for a transmitter and for appropriate decoding selection within a receiver.

The audio source encoder and the data pre-coders ensure the adaptation of the input streams onto an appropriate digital format. Their output may comprise two parts requiring two levels of protection within the subsequent channel encoder.

The multiplex combines the protection levels of all data and audio services in a proper format within the frame structure of the bit stream.

The energy dispersal provides an ordering of the bits that reduces the possibility of unwanted regularity in the transmitted signal.

The channel encoder adds redundant bits as a means for error protection and correction and defines the mapping of the digitally encoded information into QAM cells, which are the basic carriers of the information supplied to the transmitter for modulation.

Cell interleaving rearranges the time sequence of the bits as a means of “scrambling” the signal so that the final reconstruction of the signal at a receiver will be less affected by fast fading than would be the case if “continuous” speech or music were transmitted.

The pilot generator injects information that permits a receiver to derive channel-equalization information, thereby allowing for coherent (includes phase information) demodulation of the signal.

The OFDM cell mapper collects the different classes of cells and places them on a time-frequency grid.

OFDM depends on each of many subcarriers carrying its own sinusoidal amplitude/phase signal for a short period of time. The ensemble of the information on these subcarriers contains what is needed for transmission. In the case of DRM, for a 10 kHz channel, there are hundreds of subcarriers.

The modulator converts the digital representation of the OFDM signal into the analogue signal that will be transmitted via a transmitter/antenna over the air – essentially phase/amplitude representations as noted above modulating the RF.

With a non-linear high-powered transmitter, the signal is first split into its amplitude and phase components for injection in the anode and grid circuits, respectively, and then recombined (by the action of the transmitter itself set at the correct differential delay time), and then recombined prior to final emission. This splitting is not necessary for linear amplification.

(2) Distribution Interface

Referring to the extreme left of Figure 4.1, apart from audio and data applications that are multiplexed, additional information is sent that is required to instruct the transmitter to select the correct mode, error protection level, etc. and to send information in the transmission to the receivers to permit them to switch to the selection of several variables to allow for proper decoding. (The boxes and arrows for this are not shown directly in Figure 4.1.) In the aggregate, this collection of information and the means to get it to the transmitting station is called the “Distribution Interface” (DI).

These signals can emanate from a studio, or from a more elaborate network control centre, and be transmitted via land lines or via satellite circuits to the appropriate transmitter station(s). These details will not be noted here, but can be found in ETSI documents TS 102 821 and TS 102 820, both dated July 2003.

In terms of connections with other parts of the DRM system, these signals, as appropriate, are placed in either the Fast Access Channel (FAC) or the Service Description Channel (SDC) for transmission to receivers.

There are 4 categories of data associated with the Distribution Interface:

- MDI – Multiplex Distribution Interface: covers the transport of data and commands from the DRM multiplexer to the DRM Modulator.
- MCI – Modulator Control Interface: covers the remote signalling of commands and setups to the modulator and transmitter equipment.
- SDI – Service Distribution Interface: covers the transport of data and commands from the studio and other sources to the DRM Multiplexer.
- RSCI – Receiver Status and Control Interface: covers the transport of receiver status information in addition to the DRM multiplex as well as commands to control the receiver's behaviour.

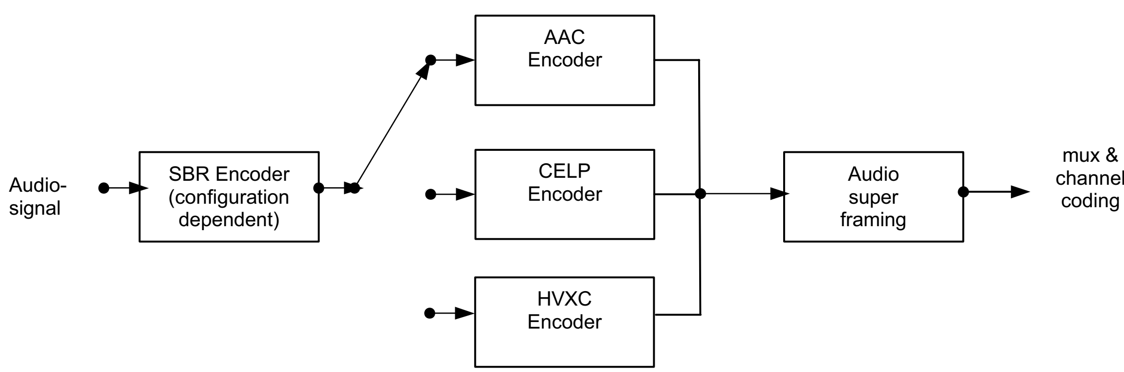
(3) Audio Source Coding

Figure 4.2 depicts the variety of digital audio encoders in the DRM system – in effect, AAC, AAC with SBR, CELP and HVXC – all of which can operate in a range of bit rates, and consequently produce a range of audio quality. (See the ETSI DRM standardization document ES 201 980 v2.1.1, 2004-06.)

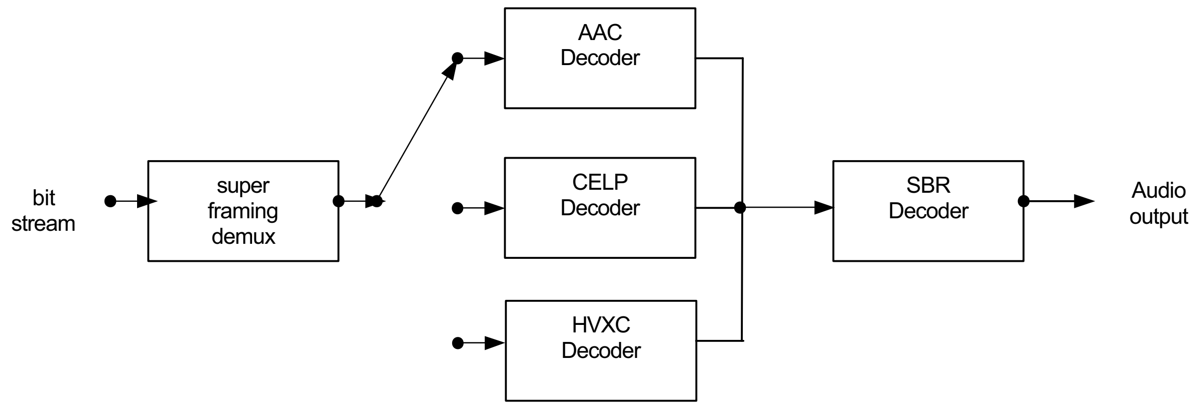
The full range runs approximately from 2 kbps (HVXC minimum) to 25 kbps (AAC maximum) within the 9/10 kHz channels for standard broadcasting in the broadcasting bands below 30 MHz. HVXC and CELP are used for "speech only" applications. AAC and AAC/SBR, within the permissible range, result in excellent music and speech audio quality.

Figure 4.2: DRM Source Encoding and Decoding

DRM Source Encoding



DRM Source Decoding



Extensive tests on these codecs at the sampling rates and resulting "bandwidths" have determined that AAC and especially AAC with SBR produce a perceived audio quality to listeners that is effectively the equivalent of monophonic FM in a 9 or 10 kHz channel. HVXC produces intelligible speech quality with bit rates of 2 to 4 kbps for HVXC and CELP produces excellent speech quality using around 8 kbps. All of these codecs are a part of the MPEG-4 audio standard.

SBR (Spectral Band Replication) is a special means of enhancing the perception of a spectrally truncated low band audio signal by utilizing, on a dynamic basis, the spectral content of the low band information to simulate the missing higher band behaviour. This requires about 2 kbps and therefore does not seriously subtract from a 20 to 25 kbps AAC output.

In concept, the technique is not complicated. Consider a violin as an example. A string stimulated by a bow and the placement of a finger on the string produces a fundamental frequency and harmonics characteristic of a violin. These frequencies can go as high as the audibility of the human ear – say somewhere between 15 and 20 kHz.

For a 9 or 10 kHz channel, the AAC sampling and processing of the violin's output can only cover the lower part of the audio spectrum, for example not higher than 6 kHz. The SBR algorithm examines this lower band spectrum on a dynamic basis and infers what the "missing" higher audio frequency "harmonics" probably are. The level of re-inserted harmonics depends on the 2 kbps SBR helper signal which describes the shape of the spectral energy in the original signal before truncation for AAC coding stereo (which uses an additional 2 kbps of SBR). From the standpoint of a listener, the combined audio output sounds like 15 kHz audio rather than 6 kHz audio.

(4) Multiplexing, including special channels and energy dispersal

This section refers to the left side of Figure 4.1 through "energy dispersal", not including the DI and audio/data encoding portions.

As noted in Figure 4.1, the DRM system total multiplex consists of 3 channels: the MSC, the FAC and the SDC. The MSC contains the services – audio and data. The

FAC provides information on the signal bandwidth and other such parameters, and is also used to allow service selection information for fast scanning. The SDC gives information to a receiver on how to decode the MSC, how to find alternative sources of the same data, and gives attributes to the services within the multiplex.

The MSC multiplex may contain up to 4 services, any one of which can be audio or data. The gross bit rate of the MSC is dependent on the channel bandwidth and transmission mode being used. In all cases, it is divided into 400 millisecond frames.

The FAC's structure is also built within a 400 millisecond frame, and is designed without interleaving, for example, to ensure rapid delivery of the information it contains. The design without interleaving is also to ensure fastest decoding of basic data by the Rx before it can do the audio decoding. The channel parameters are included in every FAC frame segment. The service parameters are carried in successive frames, one service per frame. The names of the FAC channel parameters are: base/enhancement flag, identity, spectrum occupancy, interleaver depth flag, modulation mode, number of services, reconfiguration index, and reserved for future use. These use a total of 20 bits. The service parameters within the FAC are: service identifier, short identifier, conditional access, language, audio/data flag, and reserved for future use. These use a total of 44 bits.

The SDC's frame periodicity is 1200 milliseconds. The fields of information are: multiplex description, label, conditional access, frequency information, frequency schedule information, application information, announcement support and switching, coverage region identification, time and date information, audio information, FAC copy information, and linkage data. As well as conveying these data, the fact that the SDC is inserted periodically into the waveform is exploited to enable seamless switching between alternative frequencies.

(5) Channel coding and modulation

The coding/modulation scheme used is a variety of coded orthogonal frequency division multiplexing (COFDM), which combines the OFDM with the Multi-Level Coding (MLC) based upon convolutional coding. The convolutional coding provides a level of error protection. These two main components are supplemented by time interleaving ("scrambling" of the bit stream) and the provision of pilot (predetermined value) cells for instantaneous channel estimation. All of this mitigates the effects of short-term signal fading, whether selective or flat.

Taken together, this combination provides excellent transmission and signal protection possibilities in the narrow 9 or 10 kHz channels in the LF, MF and HF broadcasting frequency bands. It can also be used for "multi-channel" DRM use; that is 18 or 20 kHz channels, using 2 contiguous ITU-R channels. This level of bandwidth will permit good stereo broadcasting.

For OFDM, the transmitted signal is composed of a succession of symbols, each including a "guard interval," which is a cyclic time prefix that provides a "dead time" to counter intersymbol interference due to multipath delay spread. Orthogonality refers to the fact that, in the case of the design of the DRM system, each symbol contains around between 100 and 200 subcarriers spaced evenly across the 9 or 10 kHz channel in such a way that their signals do not interfere with each other (are orthogonal). The precise number of subcarriers, and other parameter considerations, are a function of the actual letter modes used: ground wave, sky wave, and highly robust transmissions.

QAM is used for the modulation that is impressed upon the subcarriers to convey the information. Two primary QAM constellations are used: 64-QAM and 16-QAM. The former provides the highest audio quality, but is less robust than the latter. In addition, a 4-QAM (QPSK) signal, which is very robust, is used for some of the signalling (but not for the MSC).

The interleaver time span (applied to the MSC) for HF transmission is around 2.4 seconds to cope with time and frequency selective fading by protecting the audio and data from rapid fades during the natural sequence of speech and music. Owing to the less difficult propagation conditions for the LF and MF bands, a shorter interleaver of around 0.8 seconds can be used.

The multi-level convolutional coding scheme uses code rates in the range between 0.5 and 0.8, with the lower rate being associated with the difficult HF propagation conditions. A 0.5 code rate means that only half the transmitted bits within the overall coded block are used for the actual services in the multiplex, whereas a 0.8 rate means 80% are.

4.1.3 Transmitter Considerations

Beyond the modulator box in Figure 4.1 is the transmitter exciter. The DRM system exciter can be used to impress signals on either linear or non-linear transmitters. It is expected that high-powered non-linear transmitters will be the more usual way of transmitting, much as is done now with analogue modulation. However, there are broadcasting service situations where very low powered linear transmissions could be the best way to serve the public.

With respect to non-linear amplification (Class C operation), the incoming DRM signal needs to be split into its amplitude and phase components prior to final amplification. Using QAM modulation, there is a small discrete set of possible amplitudes and phases. The amplitude component is passed via the anode circuitry; the phase component is passed through the grid circuitry. These are then combined with the appropriate time synchronization to form the output of the transmitter.

Measurements of the output spectra show the following: the energy of the digital signal is more or less evenly spread across the 9 or 10 kHz channel, the shoulders are steep at the channel edges, and drop rapidly to 40 dB or so below the spectral density level within the assigned channel, and the power spectral density levels continue to decrease beyond the 4.5 or 5 kHz from the central frequency of the assigned channel with a rapidity that permits conformance to the ITU-R mask for the use of the channels.

(1) Over the air

The digital phase/amplitude information on the RF signal is corrupted to different degrees as the RF signal propagates. Some of the HF channels provide challenging situations of fairly rapid flat fading, multipath interference that produces frequency-selective fading within a channel and large path delay spreads of a few milliseconds or more, and ionospherically induced high levels of Doppler spreads on the order of 1 or more hertz.

The error protection and error correction incorporated in the DRM system design mitigates these effects to a great degree. This permits the receiver to accurately decode the transmitted signal information.

Extensive field tests have verified these performance statements.

(2) Selecting, demodulating and decoding of a DRM system signal at a receiver

A receiver must be able to detect which particular DRM system mode is being transmitted to handle it properly. This is done by way of the use of many of the field entries within the FAC and SDC.

Once the appropriate mode is identified (and is repeatedly verified), the demodulation process is the inverse of that shown in Figure 4.1. Similarly, the receiver is also informed which services are present, and, for example, how source decoding of an audio service should be performed.

For much more detail on DRM system, refer to the following references

- ETSI ES 201 980 v 2.1.1 (2004-06): the “signal in the air” specification
- ETSI TS 101 968 v0.0.2 (2002-08): the data applications specification
- ETSI TS 102 820 and TS 102 821: the distribution interface specifications

4.1.4 DRM+

While DRM currently covers the broadcasting bands below 30 MHz, the DRM consortium voted in March 2005 to begin the process of extending the system to the broadcasting bands up to 120 MHz. DRM Plus will be the name of this technology and wider bandwidth channels will be used, which will allow radio stations to use higher bit rate, thus providing higher audio quality. One of the new channel bandwidths that is likely to be specified is 50 kHz, which will allow DRM+ to carry radio stations at near CD-quality. The design, development and testing phases of DRM’s extension, which are being conducted by the DRM consortium are expected to be completed by 2007-2009. A 100 kHz DRM+ channel has sufficient capacity to carry one mobile TV channel: it would be feasible to distribute mobile TV too over DRM+ than via either DAB or DVB-H.

4.2 DAB – Eureka 147

Eureka 147[1] is a digital radio system developed in Europe for reception by mobile, portable and fixed receivers with a simple non directional antenna. It can be used in terrestrial, satellite, hybrid (satellite with complementary terrestrial), and cable broadcast networks and has been designed to operate at any frequency from 30 to 3000 MHz. In practice, Eureka 147 is being implemented in two spectrum bands, VHF Band III and L Band. **Further details of Eureka 147 can be found in Appendix A: The Eureka 147 System Description.**

4.2.1 System Development

Eureka[2] was established in 1985 by 17 countries and the European Union to encourage a bottom up approach to technological development and to strengthen the competitive position of European companies on the world market. It supports the competitiveness of European companies through international collaboration, in creating links and networks of innovation. The 147th Eureka technical project was to develop a digital radio system, hence Eureka 147.

The Eureka 147 Consortium[3] was founded in 1987 with 16 partners from Germany, France, The Netherlands and the UK. The Eureka 147 standard was defined in 1993 with ITU Recommendations released in 1994 and an initial ETSI standard released in 1995. Eureka closed the Eureka 147 project on 1 January 2000.

The first Eureka 147 prototype equipment was demonstrated in 1988 on the occasion of the Second Session of WARC-ORB conference held in Geneva. The first consumer type Eureka 147 receivers developed for pilot projects were released in 1995. The first Eureka 147 services commenced transmitting in the UK, Denmark and Sweden in 1995. Eureka 147 was officially launched at the Berlin IFA (a major consumer electronics show) in 1997.

The WorldDAB Forum[4] was formed in 1995 to encourage international cooperation and coordination for the introduction of Eureka 147 onto the consumer market. The technical work previously carried out by Eureka 147 now takes place within the Technical and Commercial Committee of the WorldDAB Forum. In August 2003, DRM and WorldDAB announced they would collaborate in the development of their systems.

4.2.2 Principal Advantages and Challenges

Advantages

Eureka 147 is a mature technology that has been implemented in the UK, Germany and Canada and extensively tested in other parts of Europe and in other countries including Australia.

Eureka 147 is defined by international ITU recommendations, European ETSI, Cenelec and IEC standards and national standards (e.g., British receiver standards).

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[1] Eureka 147 is also known as DAB, Eureka DAB, S!147 (S! is the logo for Eureka projects) and ITU System A. T-DAB and S-DAB may also be used to distinguish between terrestrial and satellite versions of Eureka 147.
[2] Further information on Eureka at www.eureka.be
[3] Further information on Eureka-147 consortium at http://www.eureka.be/ifs/files/ifs/jsp-bin/eureka/ifs/jsps/projectForm.jsp?enumber=147.
[4] Further information on WorldDAB forum at http://www.worlddab.org/dab

Many ancillary aspects of the Eureka 147 system, such as multimedia delivery, distribution interfaces and user interactivity are also formally defined in ETSI standards.

Eureka 147 can be implemented for a range of applications such as wide area or local delivery of audio and data services for mobile, portable and fixed reception. It can be delivered terrestrially, via satellite, cable or a mixture of terrestrial and satellite.

Eureka 147 is designed to be used across a wide spectrum range, from 30 to 3000 MHz, but has only been implemented using VHF Band III and the 1452 to 1492 MHz segment of the L Band.

Eureka 147 uses a wideband COFDM modulation system which provides a robust transmission which is multi path resilient and can provide high availability coverage.

Eureka 147 can be implemented using on channel repeaters in SFNs or low power gap fillers and extenders. SFNs may also provide “network gain” giving improved service availability over single channel services.

Eureka 147 can accommodate a varying number of audio services of differing quality with associated data. The audio quality can range from simple mono speech to CD quality. An increase in quality requires higher data rates for each audio service, hence reducing the number of services that can be delivered. Data can also be delivered independently of the audio services.

Eureka 147 uses mature technologies such as MPEG 1 Layer II and MPEG 2 Layer II audio coding systems and COFDM modulation, which are also used in the DVB T video broadcasting standard. This should lead to cheap single chip solutions for receivers.

Eureka 147 has been extensively standardised by European standards organisations and it would be fairly straightforward for these standards to be adopted as Australia standards by Standards Australia.

A growing number of Eureka 147 receivers are now available for portable, PCs, mobiles, in car and in house reception.

Challenges

The MPEG 1 Layer II and MPEG 2 Layer II audio coding systems are now somewhat dated (compared with new systems) but they offer excellent robustness against channel errors due to unequal error protection (UEP). The system allows for inclusion of newer coding systems as independent data, but DAB receivers would need to be adapted or replaced to receive these services.

While a wide range of DAB receivers is already available on the market, they are still generally seen as being too costly for general public acceptance, particularly when compared to the very cheap AM and FM radios that many listeners currently use. However, as Eureka 147 services have expanded, the cost of receivers have considerably dropped in price.

Eureka 147 requires services to be multiplexed together before transmission. All audio programs and data services in a given Eureka 147 channel will therefore have similar coverage and reception quality.

The standard capacity of Eureka 147 multiplexes means that conversion would require existing services to be grouped into blocks of 6 or more services per multiplex, all of which would then cover the same area. In a conversion model, this would pose challenges for many current radio broadcasting markets, which are typically served by a mixture of narrowcasting, community, commercial and national services using AM and FM frequencies with different or overlapping licence and coverage areas giving local, medium or wide area coverage. Conversely, the requirement for multiplexing could over time reduce the number of transmission sites and result in more consistent coverage of services.

Eureka 147 uses spectrum that is often used for analogue and digital television services (VHF Band III), and radio communication services (L Band). If a conversion model is used for the introduction of digital radio finding, sufficient spectrum for the conversion of all analogue radio broadcasting services to digital will not be easy, particularly as L Band will require more transmitters to provide wide area coverage and adequate reception in urban areas.

4.2.3 DAB Development Worldwide as of 2006

More than 40 countries have legislated for the integration of DAB Digital Radio in Europe and Worldwide. Outside Europe the key areas of development are found in Canada, the Asia-Pacific Region and South Africa.

(1) Belgium

DAB Digital Radio launched in Belgium in September 1997 with a multiplex operated by the Flemish public broadcaster VRT. Today, the VRT multiplex covers the Flemish Community and has nine audio stations. Four of these channels are unique to DAB Digital Radio. RTBF, the public broadcaster for the French community, has a multiplex covering the French community with five audio stations, all simulcasts of existing analogue stations.

(2) Canada

DAB launched in Canada in November 1999. Stations in Toronto, Montreal and Vancouver started operating in 1999; Ontario in 2000; and Ottawa in 2003. There are currently a total of 73 licensed Digital Audio Broadcast DAB stations in Canada: 15 stations in Ottawa, 25 in Toronto, 15 in Vancouver, 12 in Montreal and 6 in Windsor. The stations operating in these five cities provide services to some 11 million potential listeners or more than 35% of the population. Seven DAB stations (4 commercial and 3 public) are field testing in Halifax, Nova Scotia.

DAB has yet to be embraced by consumers in Canada. The industry is currently evaluating next steps with respect to digital radio rollout. Implementation of other digital radio systems is under consideration, particularly as rollout of HD Radio in the neighbouring United States proceeds.

(3) Denmark

Danish Broadcasting Corporation (DR) is currently broadcasting 18 DAB “channels.” On September 1, 2005, the commercial broadcasters Sky Radio and Radio 100 FM (owned by Talpa Radio International) commenced transmission on the DAB networks. Both started simulcasting their FM stations utilising 25 percent respectively on the national and the two regional networks. DR continues broadcasting in the remaining 75 percent of both networks.

(4) France

Public broadcaster, Radio France, serves an area covering Paris, Marseille, Toulouse and Nantes, broadcasting six stations in each of its service areas.

(5) Germany

Germany is among the leading European proponents of DAB Digital Radio with a large local and regional network. Current figures put coverage in Germany at 78 per cent of the population, rising to 85 per cent by the end of 2005. Germany is the primary DAB digital radio country in mainland Europe boasting some 120 radio stations, both public and commercial, on a variety of state-wide as well as local digital multiplexes. As far as data services are concerned, Germany is at the fore. The focus lies on news and traffic information, using text and pictures. In March 2005, BLM, the Bavarian Media Authority, launched a pilot project called Digital Advanced Broadcasting that will aim to use DAB to broadcast radio and video content, as well as data services, to new portable receives. The pilot will take place in Regensburg and is expected to last for two years with the aim to work towards comprehensive coverage of FIFA World Cup 2006 via mobile entertainment devices.

(6) Italy

Current population coverage in Italy stands at approximately 45% for commercial operators and 20% for national public operators. There is a scarcity of Band III spectrum and its use and management for digital and /or analogue TV is yet to be fully solved. This has caused delay in the roll-out of DAB in Italy.

Italy has been broadcasting DAB Digital Radio since 1995 when public broadcaster RAI began simulcasting its existing services. In 1998, eight commercial analogue operators formed a consortium called Club DAB Italia in order to simulcast their own stations on their own digital multiplex.

Currently nine national commercial stations are broadcast by the Club DAB Italia consortium in the Milan area and vast adjacent areas and in Rome and adjacent areas, for about 30 percent of population coverage. Also, five national public services are simulcast on a multiplex reaching less than 20 per cent population coverage.

(7) Singapore

Regular DAB Digital Radio services in Singapore were launched on 19th November 1999. The MediaCorp Radio Singapore Pte Ltd dubbed their multiplex SmartRadio. SmartRadio carries 14 radio services - six of which are available exclusively on DAB radio and eight are simulcasts of the more popular FM stations. In April 2005, Rediffusion, which is Singapore’s sole subscription radio broadcaster, was awarded a license to provide the world's first DAB subscription services and operate its own multiplex.

(8) South Korea

In 2002, the Ministry of Information and Communication (MIC) in Korea approved the use of DAB for the transmission of audio, video and data using Digital Multimedia Broadcasting (DMB). In March 2005, six service providers were selected by the Korean Broadcasting Commission (KBC) to receive licences: KBS, MBC, SBS, YTN DMB, K-DMB and KMMB. The six broadcasters will in total carry 6 video, 18 audio and 12-18 data programmes, and all will be free of charge at first to make the service universally available.

(9) Spain

Spain enjoys a strong commitment to DAB Digital Radio, with the current 52 per cent population coverage expected to rise to 80 per cent by 2006.

DAB Digital Radio in Spain began with pilot stations in 1998 and today is a mix of public and commercial broadcasting, with 18 stations transmitting digitally.

Spanish broadcasters are currently experimenting with data services and there are plans for local DAB Digital Radio stations.

(10) Sweden

Whilst there is no coverage requirement built into the digital radio legislation in Sweden, Swedish Radio, the public broadcaster, covers 85 per cent of the population. Since 2002, a temporary reduction of the transmissions to 37 per cent population coverage has been made for financial reasons.

(11) Switzerland

Switzerland currently has approximately 4 million potential listeners (60% of Swiss population). In 2006, the coverage area in the German speaking part of Switzerland will be enlarged and indoor reception enhanced. Also, full DAB coverage across Switzerland has been approved for 2007-2008.

(12) Taiwan

In June 2005 the Taiwan Government awarded 6 commercial multiplex licenses: 3 nation-wide SFN licenses and 3 regional licenses (covering the major conurbations like Taipei and Causing).

(13) United Kingdom

The public service broadcaster, the BBC, has been promoting its DAB Digital Radio stations since September 1995 and at present covers over 85 per cent of the population and includes the major motorway network. Digital One, the UK's only national commercial operator, runs the world's biggest digital radio network, with more than 90 transmitters. Further transmitters are planned to expand the network towards 90 per cent coverage.

Mid- 2005 saw nearly 150 DAB digital radio products in the market with today's figures far in excess of this. The range of DAB radios includes portables, hand-helds, boom boxes, clock radios, micro systems, home cinema and in-car products. Established manufacturers have helped to drive sales with high profile advertisements in the national press.

The BBC has run a number of campaigns on television, radio and online promoting DAB digital radio and programme content. Each campaign has seen a surge in sales of products and consumer awareness rising. During 2005, the BBC continued to promote individual services and content.

The Digital Radio Development Bureau (DRDB) is funded and supported by UK broadcasters including the BBC, Digital One, EMAP Digital Radio, MXR and GCap media. The DRDB's task is to ensure DAB's wide accessibility and swift adoption in the UK.

Table 4.1. Eureka 147 Main System Features

EUREKA 147 Main System Features	
Single Frequency Network (SFN) capability	All transmitters working on a single frequency.
Flexible audio bit rate	Allows reconfiguration of the multiplex.
Data services	Separately defined streams or packets.
Programme Associated Data (PAD)	Embedded in the audio bit stream and adjustable.
Facilitates Conditional Access	DAB ensemble transports conditional access information (CAI) and provides signal scrambling mechanism.
Service Information	Used in the operation and control of receivers.
Operating frequency range	30 MHz to 3 GHz.

4.2.4 Infrastructure Requirements

Eureka 147 is a wideband technology requiring services to be multiplexed before transmission. The use of VHF and UHF bands means Eureka 147 services will be typically transmitted from high sites such as the tops of hills, buildings or towers.

In general, new Eureka 147 services are also likely to be co-located with existing FM radio or television transmission services given the cost of developing new sites and the increasing difficulty in getting local council planning approval for new transmission sites.

In Canada, implementation of the Eureka system uses a new band (L-Band), hence new transmitters, antenna system, exciter and encoders have been required. Stations that were originally broadcasting more than one FM program from the same site can fully encapsulate the multiplexed stream of the DAB system in the STL (studio-to-transmitter link), significantly reducing the costs associated with discrete feeder links. Canada's DAB allotment plan has room for the replacement of all existing AM and FM stations in the L-Band. The plan also includes many empty allotments for future services. Finally, since the plan was based on providing only five programming channels in each DAB multiplex, new audio coding schemes will allow for the possible implementation of two to three additional services in each ensemble.

4.2.5 Synergies with Other Systems

(1) DAB and GSM (Global System for Mobile Communications)

DAB is an efficient broadcasting (e.g., one-to-many) system capable of providing reliably digital services to all users located in a coverage zone in real time. It is especially suitable for the reception to mobile and portable receivers and in the areas in which the direct line of sight between the transmitter and the receiver is not possible.

On the other hand, GSM (Global System for Mobile Communications) and its successors (GPRS and UMTS) are more suitable to deliver on-demand media services to individual clients or relatively small groups of clients. The telecom systems are technically able to provide services to several users in the same time, providing that the number of simultaneous users (or, in other words, the total bandwidth capacity) does not exceed a certain level, or else the network collapses rapidly. Also, the use of telecom services in the "one-to-many" scenario is much more expensive for the user than the use of DAB (or DVB) broadcast systems.

It may be advantageous for both broadcasters and telecoms to provide a combination of both one-to-many and one-to-one applications concurrently. For example, a traffic/travel information service may consist of two parts: a basic part and a value-added part. The former would be carried over the broadcast network to everybody (possibly for free), whereas the latter would be available on-demand over the telecom network and would be paid-up according to a tariff agreed.

As an example of such synergy, Nagra-Futuris has created an IT infrastructure for hosting end-to-end interactive services based on the existing GSM and DAB technologies. The system provides back-channel communication, conditional access, data warehousing, integrated billing/clearing and interfacing to external M-commerce providers. The system allows for deployment of interactive services and dynamic insertion of programme related data. The mobile terminal device is a combination of mobile phone and DAB receiver. It contains a DAB Identification Module (DIM).

The EBU have identified many attractive interactive applications and business opportunities based on DAB/GSM synergy. Such synergetic services may help telecoms to generate more traffic and offer new, rich-content services (games, live and on-demand video/audio clips, etc.).

Synergies of GSM and DAB networks may be useful in the case of DAB single-frequency networks (SFN) at L-Band. To set up an SFN network at L-Band, the transmission sites must not be any further than 18 km apart using Transmission Mode II in order to maintain network timing and to benefit from the network gain of an SFN. Therefore an ideal SFN at L-Band could emulate the infrastructure of mobile phone networks with lower masts and powers.

(2) Synergies with Digital Radio Mondiale (DRM)

DAB and DRM are complementary as they target different markets. DAB is mainly intended for local, regional and national audiences. DRM is designed to be deployed in the frequency bands below 30 MHz to replace existing AM services and targets more large coverage zones. This system has been successfully standardised within ITU and ETSI and is now being implemented in the commercial market. Future listeners will be interested in all services provided by digital radio, hence radio sets should enable the users to receive any digital radio service without concern for the transmission system. In terms of the technologies used, both systems are not too dissimilar; for example, both are using COFDM and similar channel coding strategies. To this end, common integrated circuits are being developed and integrated DAB/DRM receivers could soon appear in the market. A common interface for external devices is also being developed.

In August 2003, DRM and WorldDAB announced they would collaborate in the development of their systems." Reference: http://www.worlddab.org/press.aspx

(3) Synergies with Digital Television

Although the DVB systems (e.g., DVB-S, DVB-C and DVB-T) were primarily designed for television broadcasting, they can and do provide radio (audio-only) programs. DVB-T is a proven technology for digital television and has been implemented in many countries. As DAB, DVB-T is, technically speaking, sufficiently flexible to allow for delivery to portable and mobile receivers. Challenges with implementing DVB-T for digital radio centre on the need for good mobile and portable reception and its large bandwidth usage. DVB-T is not optimised for mobile reception and no mobile or portable hand held receivers are as yet available. The high data rates and wide bandwidth needed to operate the system not only increases power consumption but also makes the design of battery-powered devices difficult. The large bandwidth use required for DVB-T means that many services must be multiplexed together for efficient use of the spectrum and there is a risk that such multiplexes may not be fully utilised, thereby leading to inefficient spectrum use.

Experience suggests that DVB-T platforms designed primarily for digital television are increasingly likely to carry audio-only entertainment and information as well. Most current implementations of DVB-T services for digital television target fixed reception. Consumer-grade mobile DVB-T receivers are likely to be produced with the aim of providing mobile television and multimedia services.

From the technical perspective, DAB and DVB-T are both using the same modulations: OFDM. Therefore it will not be surprising to see common DAB/DVB-T chips developing rapidly. Frontier Silicon announced that they are planning to develop a single chip DVB-T and DAB decoder termed "Logie," for which they have already signed a number of customers.

4.2.6 Future Developments of DAB

The technical developments of the DAB system go into two directions:

- associate audio services with flexible multimedia services including moving pictures
- multi-channel audio
- enhanced audio codec, DAB+

(1) DAB-Based Multimedia Broadcast Systems (DMB) T-DMB

Digital Multimedia Broadcasting (DMB) uses an MPEG-2 TS with additional error protection (Reed-Solomon (188,204) code and interleaving as specified for DVB services) transmitted in a DAB Stream Mode sub-channel. Bosch originally proposed the use of MPEG-2 TS to carry one video service in a DAB Ensemble. Subsequently there were proposals by Bosch and the collaborative project MINT (funded by German BMBF) to use MPEG-4 video coding to fit several video services in a DAB Ensemble. Later there was further development and promotion of T-DMB in Korea, and a parallel development of the S-DMB system for satellite BTH. T-DMB specifications were approved by WorldDAB (December 2004) and were standardized at ETSI (June 2005).

T-DMB receiving devices have become available and are integrated within mobile phones, in portable PCs and small screen portable devices. Several pilot trials and projects are ongoing in Korea, UK, Germany, France and elsewhere. It should be noted that Korea has deployed both Satellite (S-) and Terrestrial (T-) DMB, although these have limited technical similarities leading to very different terminal devices. T-DMB was introduced in Korea in mid-2005 using an existing terrestrial network in Band III (although formal commercial launch has been delayed whilst business issues are coordinated). Frequencies in L-Band are available in much of Europe for possible use with DMB and DAB.

European T-DMB was officially launched on 7 June 2006 in Munich on the occasion of the World Football Cup. The launch was organized by WorldDAB and its partners.

(a) IP over Enhanced Packet Mode

Enhanced Packet Mode (EPM) provides additional error protection for DAB packet mode-based services, such as IP and MOT (Multimedia Object Transfer), by the use of a DAB-FEC frame and the addition of FEC packets (in a similar way to the DVB-H MPE-FEC). The same Reed-Solomon code is used as in DMB. Interleaving is different from T-DMB and allows backwards-compatible reception of EPM services on receivers with conventional DAB packet mode. The EPM specification has been submitted to ETSI.

(b) DAB-IP

The BT Movio's "DAB-IP" system is a DAB application of IP over Enhanced Packet Mode. Technical trials in UK by British Telecom started mid-2005 and ran through to the end of December 2005. Microsoft's solution for video and audio coding as well as digital rights management (DRM) have been selected for this pilot. The electronic programme guide (EPG) designed for BT Movio and standardized by ETSI proved quite successful. DAB-IP enables DAB digital radio to share multiplex capacity with mobile TV and therefore allows TV operators to benefit from the considerable DAB spectrum and infrastructure investments that have been made across Europe. The prototype DAB-IP devices were based on a fully functioning 2.5G mobile phone which included an integrated DMB receiver, so that users could enjoy broadcast digital TV and radio services using advanced EPG.

(c) The German DXB Project

Digital Extended Broadcasting (DXB) is a German-funded project running until 2007. The DXB concept will offer similar services to DVB-H over a DAB-based transmission system. Services may use the IP-protocol over Enhanced Stream Mode (using MPEG-2 TS as with DMB) or via the Enhanced Packet Mode.

It should be observed that an alternative broadcast system for mobile multimedia applications is being developed in the framework of the DVB Project: DVB-H (H stands for handheld). Some EBU research institutes are in the process of looking into the technical and operational merits of DVB-H and

DAB. Notwithstanding the results of such a study, it should be remembered that the ultimate choice may not necessarily be taken on purely the technical grounds. The history teaches us that not always the best technology wins, as the business interests may sometimes be more important (e.g., VHS versus Betamax about VCR technology).

(2) DAB as carrier of multichannel audio

Concerning multi-channel audio, many EBU broadcasters would like to see it introduced not only in the satellite and cable systems but also in terrestrial DAB and DVB-T systems. To this end, the EBU Broadcast Management Committee set up a Focus Group B/MCAT (Multi-Channel Audio Transmission) which is due to start its work in February 2004 . The EBU Village at IBC 2003 in Amsterdam staged a very successful demonstration of some pre-recorded multi-channel material (such as the famous production of Österreichischer Rundfunk's New Year Concert from Vienna) as well as some live broadcasts over the Astra satellite prepared by Bayerischer Rundfunk.

Some argue that multi-channel audio is more appropriate for television, particularly as an adjunct to enhanced TV or HDTV, and less so for radio. The DVB system has recently been extended to be able to accommodate not only MPEG-2 multi-channel audio but optionally Dolby Digital (AC3) and Digital Theatre System (DTS), with the proviso that further hooks for other systems such as AAC may follow.

Others believe that multi-channel audio could enhance users' experience in the radio environment significantly and make DAB even more popular, not only in the home environment but also (or especially) in the car. Many consider that multi-channel DAB could be branded as the future "high definition" radio and could differentiate DAB from FM to drive new business models and make it more attractive for the general public.

There are several possible scenarios how multi-channel audio could be brought into the DAB system efficiently and in a backwards-compatible manner. For example, one possible solution (not necessarily the best) would be to code the basic stereo in the existing standard MPEG 1/2 Layer II and the "surround" component in AAC. The downside is that multi-channel sound requires more spectrum - which is a very scarce resource indeed, and requires new production facilities and increases the production costs.

At IBC 2003, Microsoft, Capital Radioplc, NTL Broadcast and RadioScape announced that they planned to conduct a trial broadcast of 5.1-channel surround sound audio signals over DAB in the central London area. This trial started in October 2003 and involves live IP data casting of Widows Media Audio 9 Professional (WMA Pro) content coded at 128 kbps over L-Band frequencies.

(3) Enhanced Audio Codec, DAB+

This enhancement to DAB formally was published as an ETSI standard on 12 February 2007 (ETSI, TS 102563 V1.1.1).

The new audio codec MPEG-4 HE-AAC v2 offers broadcasters much higher bandwidth efficiency which results in significant cost savings per channel and the possibility to broadcast more channels in a multiplex than before.

New receivers which appear on the market with the new codec will also be backwards compatible with the existing DAB-MPEG Audio Layer II in operation today for DAB services.

The main features for the new audio codec are described by WorldDAB as:

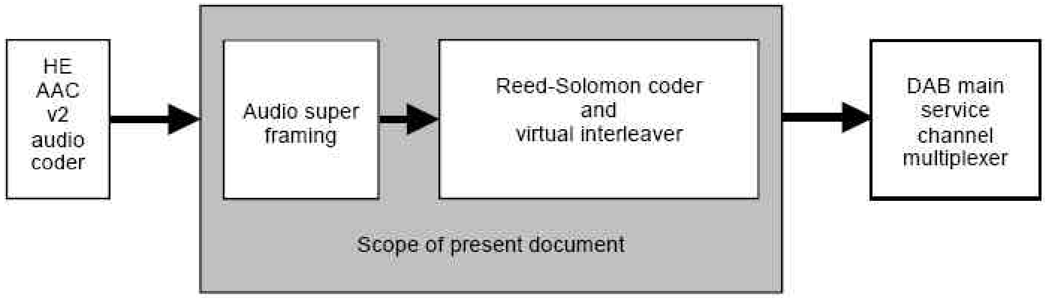
- Latest MPEG-4 audio codec delivers exceptional performance efficiency;
- More stations can be broadcast on a multiplex;
- Greater station choice for consumers;
- More efficient use of radio spectrum;
- Lower transmission costs for digital stations;
- New receivers backwards compatible with existing codec standard;
- Current MPEG Audio Layer II services and consumers unaffected;
- Compatible with existing scrolling text and multimedia services;
- Robust audio delivery with fast re-tuning response time;
- Optimised for live broadcast radio;
- Broadcasters/regulators can select either standard MPEG Audio Layer II, or optional high efficiency advanced audio codec, or both, to suit their country needs.

The following is a brief explanation of how the new codec enhances the DAB standard.

The main Digital Audio Broadcasting specification (ETSI EN 300 401) defines how audio should be broadcast. “The DAB system uses MPEG Audio Layer II, suitably formatted for DAB transmissions. For 48kHz sampling frequency it uses ISO/IEC 11172-3 and for 24kHz sampling frequency it uses ISO/IEC 13818-3.”

For Layer II audio, two sampling rates are permitted, 48 kHz and 24 kHz. Each audio frame contains samples for 24 ms or 48 ms respectively and each contains the same number of bytes. The audio frames are carried in one or two respectively DAB logical frames. The draft technical specification now approved by ETSI defines the way that audio (programme) services are carried when using MPEG 4 HE AAC v2. For AAC, two transforms are specified. For DAB, only the 960 transform is permitted with sampling rates of 48 kHz, 32 kHz, 24 kHz and 16 kHz. Each AU (audio frame) contains samples for 20 ms, 30 ms, 40 ms or 60 ms respectively. In order to provide a similar architectural model to Layer II audio, simple synchronisation and minimal re-tuning delay (i.e. station selection, or “zapping” time), AUs are built into audio super frames of 120 ms which are then carried in five DAB logical frames. In order to provide additional error control, Reed Solomon coding and virtual interleaving is applied. The overall scheme is shown in Figure 4.3.

Figure 4.3: Conceptual diagram of the outer coder and interleaver



For generic audio coding, a subset of the MPEG-4 High Efficiency Advanced Audio Coding v2 (HE AAC v2) toolbox - chosen to best suit the DAB system environment - is used. Some additional tool specifications have been applied to optimise performance for the broadcast environment of DAB digital radio.

More details can be found on the WorldDAB websites at www.worldDAB.com.

4.2.7 Types of Receivers

A selection of DAB digital radios has been on the market since 1999 in models for the home, the car and the PC. Handheld radios also entered the market in 2003 with competitively priced models for radio listeners on the move. Stations can also be accessed using a PC equipped with a suitable receiver/decoder card.

Availability of DAB digital radios varies from country to country where DAB is available. Department stores, independent retailers, supermarkets and multiples in most major cities in the UK carry a wide range of receivers representing a mature retailer market scenario. In the UK there are nearly 150 products in the market rising to around 200 products by the end of 2005. In Germany, Belgium, Nordic countries and Singapore, DAB is becoming available via specialist stores, independent retailers and some department stores in small quantities, and in Italy through manufacturers' catalogues. In Spain and France, receivers can be ordered and delivered on demand.

Prices of digital radios were high at first, but as with most new technologies, over time, prices have fallen dramatically. Receiver prices vary from country to country, so it is impractical to present specific price points within this guide. However, DAB digital radios can now be found retailing from €60. A price guide is available from the WorldDAB Project Office, on request, on the WorldDAB website, www.worlddab.org/dabprod.aspx.

Models delivering additional features have also been developed, with rewind, pause, record functionality and Electronic Programme Guide (EPG) already integrated in some models.

(1) In-Home Receivers

With launch prices high in 1999 and very little broadcast content available, take-up was predictably slow, and product was initially stocked only by a handful of specialist retailers. Today, a wide range of manufacturers have joined the market, and costs have fallen significantly. New players have joined the DAB Digital Radio family with many different models, including both mains and battery powered portable units.

(2) In-Car Receivers

DAB Digital Radio was originally designed for mobile reception and so forms a natural alliance with the in-car radio market. Manufacturers have been quick to realise the potential of DAB Digital Radio on the move and no less than eight companies are currently making a range of products to suit all tastes and pockets. Most manufacturers have established a low to mid-range price point for the in-car digital radio package and some manufacturers are offering line-fit options. For the audio enthusiast there are more expensive products on offer.

In 2005, most car manufacturers started offering DAB as upgrade option (General Motors, Ford, Volvo, Audi, Volkswagen, Jaguar, Land Rover, Mitsubishi, Renault and DaimlerChrysler). Vauxhall and General Motors offer DAB as standard on UK models, with plans to rollout in Europe. In 2006/07, many manufacturers are planning standard fit of DAB on various models.

(3) Handheld Receivers

DAB technology and advances in silicon technology have led to the development of DAB handheld and pocket radios. Manufacturers have moved quickly to produce handheld products for the DAB market and the majority of them have established a low to mid-range price point.

(4) PC Receivers

Alongside in-home, in-car and handheld equipment, DAB Digital Radio can also be enjoyed at home and at work using a personal computer. Several devices were on the market up until 2004, allowing the consumer to tune into DAB stations via either a desktop unit or a laptop, but without the need for an Internet connection. DMB enabled laptops and USB devices are also being developed enabling the possibility of DAB PCs.

Table 4.2: List of manufacturers and their DAB products

Receiver Manufacturer	Type of Receiver					Website
	Portable	Tuner/ Hi-Fi	In-Car	Hand-Held	PC	
Acoustic Solutions	v	v				www.acousticsolutions.co.uk
Alba Radios Limited	v					
Albrecht	v			v		www.albrecht-online.de
Alpine		v	v			www.alpine-europe.com
Arcam		v				www.arcam.co.uk
Arion Technology	v			v		www.arion.co.kr
Audionet		v				www.audionet.de
Bang & Olufsen		v				www.bang-olufsen.com
Blaupunkt			v			www.blaupunkt.de
BT	v					www.shop.bt.com
Bush Digital	v	v		v		www.bushdigital.co.uk
Cambridge Audio		v				www.cambridgeaudio.com
Clarion			v			www.clarion.co.uk
Crown	v					www.crowncorporation.co.uk
Cymbol		v				www.cymbol-hifi.co.uk
ELANsat	v					www.elansat.com
Eltax Ltd	v					www.eltax.com
Ferguson	v	v		v		
Genus Digital	v	v				www.genusdigital.com
Goodmans	v	v	v			www.goodmans.co.uk
Grundig	v			v		www.grundig.com
Harman Kardon		v				www.harman.com
Hitachi	v	v				www.hitachi.com
Homecast Europe				v		www.homecast.de
Intempo Digital	v					www.intempo-digital.co.uk
iTech Dynamic	v					www.itechdynamic.com
JVC		v	v			www.jdl.jvc-europe.com
Kenwood			v			www.kenwood.com
Kiiro		v				
Kiss				v		
Kjaerulff	v					www.kjaerulff1.com
LG Electronics	v	v		v		www.lge.com
M & G Audio	v					
Matsui	v	v				
Maycom	v			v		
Ministry of Sound	v	v	v	v		www.shop.ministryofsound.com
Modular Technology					v	www.modulartech.com
Morphy Richards	v					www.morphyrichard.co.uk

Receiver Manufacturer	Type of Receiver					Website
	Portable	Tuner/ Hi-Fi	In-Car	Hand-Held	PC	
Nevada				v		www.nevadaradio.co.uk
Onkvo		v				
Opel			v			www.opel.de
Orbit	v					www.orbitronics.com
Panasonic	v	v				www.panasonic.de
Perstel	v		v	v		www.perstel.com
Philips	v	v		v		www.consumer.philips.com
Pioneer			v			www.pioneer-eur.com
Proline	v					
PURE Digital	v	v		v		www.pure-digital.com
Restek		v				www.restek.de
REVO Digital			v			www.revo.co.uk
Roadstar			v			www.roadstar.com
Roberts	v			v		www.robertsradio.co.uk
Samsung		v		v	v	www.samsung.co.uk
Sangean	v	v				www.sangean.nl
Sanyo		v				www.sanyo.com
Sharp	v					www.sharp.co.uk
Siemens VDO Automotive			v			www.3vdo.com
Sony	v	v		v		www.sony.co.uk
Steepletone		v				www.steepletone.com
TAGMcLaren		v				www.internationalaudiogroup.com
TEAC	v	v				www.teac.co.uk
Technisat	v			v		www.technisat.com
Terratec		v			v	www.euro-tech.co.uk
Trinloc	v	v		v	v	www.trinloc.de
VDO Dayton			v			www.vdodayton.de

The above list is not exhaustive; new products continuously come onto the market.

Highlighted cells indicate DMB products.

More information about the Eureka 147 system is included as Appendix A.

4.3 Japan's Digital Radio Broadcasting (ISDB-TSB)

4.3.1 Overview

ISDB-TSB (Integrated Services Digital Broadcasting – Terrestrial for Sound Broadcasting) system was developed for terrestrial Digital Sound Broadcasting (DSB) and was included in the ITU-R Recommendation BS.1114-3 in 2004. The system specification was developed by the Association of Radio Industries and Businesses (ARIB) in October 1998. Laboratory experiments and field trials using Tokyo Tower were carried out to verify the system performance in 1999 and the final specification was approved as a Japanese Standard in November 1999.

Two stations were launched in Tokyo and Osaka in the frequency band 188 MHz to 192 MHz in October 2003.

4.3.2 The Methods

A terrestrial TV broadcasting frequency band that fits for mobile communications, OFDM (Orthogonal Frequency Division Multiplexing) that withstands interference caused by multiple paths (delayed waves), a modulation method that fits for communications with cell phones and mobile receivers, powerful error correction function, etc., have been adopted to allow good communications with cell phones and mobile receivers.

Concerning information compression technology and multiplexing technology, MPEG-2 has been adopted after diverse compatible communications with digital broadcastings (such as terrestrial digital TV broadcasting, BS digital broadcasting, CS digital broadcasting) were considered. MPEG-2 offers a common base for signal processing, which leads to reduction in the production cost of receivers by using LSI-chip and consolidation of receivers as well as easy exchange of data with other media.

Since this broadcasting system has the common segment structure with terrestrial digital TV broadcasting, the receivers can be consolidated.

(1) Audio encoding system

MPEG-2 AAC (Advanced Audio Coding) and SBR (Spectral Band Replication) have been adopted. However, SBR is optional.

This system satisfies the ITU-R standard, which enables high-quality multiple channeling at a low bit rate of 144 kbps or so. It has been adopted by BS digital broadcasting and terrestrial digital TV broadcasting. The adoption to the DSB resulted from the consideration of cross-media communications.

(2) Restricted reception system

MULTI2 system has been adopted.

A scramble system has been adopted for charged broadcasting. It is the MULTI2 system that has already been adopted for terrestrial digital TV broadcasting, BS digital broadcasting, and CS digital broadcasting. The adoption to the DSB resulted from the consideration of cross-media communications. Introduction of charged broadcasting depends on the result after the feasibility is examined.

(3) Multiplexing System

MPEG-2 system has been adopted, therefore various digital contents such as sound, text, still picture, moving picture and data can be transmitted simultaneously.

In addition, cross-media communications were considered because MPEG-2 system has been adopted in terrestrial digital TV broadcasting, BS digital broadcasting and CS digital broadcasting.

(4) Transmission channel encoding system

Modulation method

OFDM method that withstands interference with multiple paths has been adopted. One of DQPSK (Differential Quadrature Phase Shift Keying), QPSK, 16 QAM (Quadrature Amplitude Modulation), and 64 QAM can be used. Since different forms of broadcasting are expected, parameters are available for setting carrier modulations and coding rate of inner code.

Error correction system

Reed solomon (204, 188) for external signalling and convolution coding (convolution rates: 1/2, 2/3, 3/4, 5/6, 7/8) for internal signalling have been adopted.

The adoption resulted from the consideration of high coding efficiency and high burst error correction capability for external signalling, various options of coding rates for internal signalling and cross-media communications.

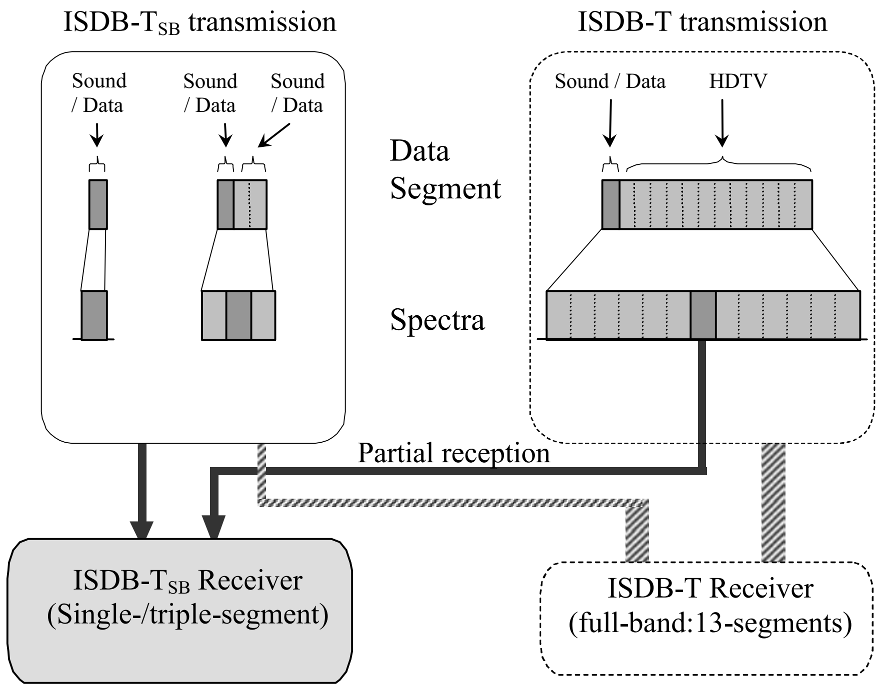
According to the broadcaster's purpose, they can select the carrier modulation method, error correction coding rate, etc., of the system. The TMCC (Transmission and Multiplexing Configuration Control) carrier transmits the information to the receiver pertaining to the kind of modulation method and coding rate used in the system.

(5) Transmission bandwidths

A transmission bandwidth that uses one OFDM segment of 6/14 MHz (approx. 429 kHz) bandwidth has been primarily adopted. In addition, a transmission bandwidth that uses three OFDM segments is also available.

Figure 4.4 shows ISDB-TSB and full-band ISDB-T transmission concept and its reception.

Figure 4.4: ISDB-TSB and full-band ISDB-T transmission concept and its reception



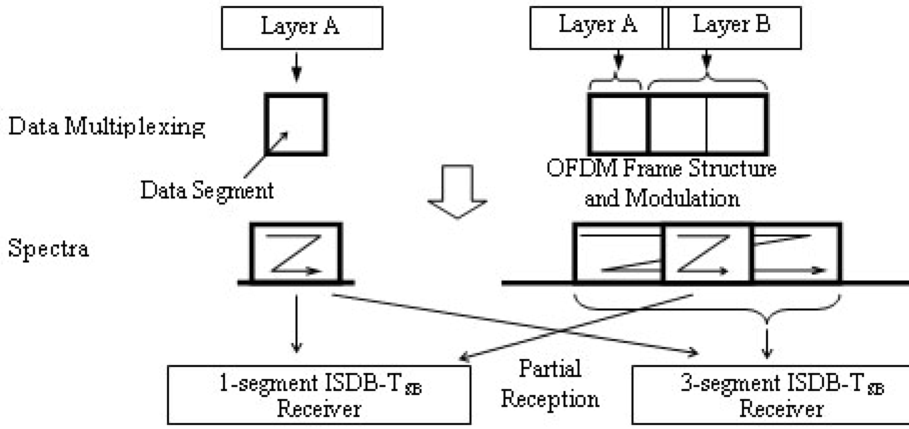
(6) Hierarchical transmission and partial reception

In the triple-segment transmission, both one layer transmission and hierarchical transmission can be achieved. There are two layers of A and B in the hierarchical transmission. The transmission parameters of carrier modulation scheme, coding rates of the inner code and a length of the time interleaving can be changed in the different layers.

The centre segment of hierarchical transmission is able to be received by single-segment receiver. Owing to the common structure of OFDM segment, single-segment receiver can partially receive a centre segment of full-band ISDB-T signal whenever an independent program is transmitted in the centre segment.

Figure 4.5 shows an example of hierarchical transmission and partial reception. In Japan, hierarchical transmission mode has to be used in the case of triple-segment transmission.

Figure 4.5: Example diagram of hierarchical transmission and partial reception



(7) Connected transmission

Efficient transmission

Connected transmission is defined as a transmission of multiple segments (e.g., multiple programs) from the same transmitter with no guard band.

In addition, the channels of independent broadcasters can be transmitted together without guard bands from the same transmitter as long as the frequency and bit synchronisation are kept the same between the channels.

But broadcasters can have their own RF channel in which they can select transmission parameters independently.

The following two advantages are available from connected transmission:

- Facility and maintenance costs are low because only a single broadcasting facility is required.
- Effective use of the frequency is enabled because no guard band between segments is required.

The connected transmission technique is in operation for the first time in the world.

An example of connected transmission for three TS's (TS1, TS2, and TS3) is shown in Figure 4.6. Each TS signal is independently channel-coded. After OFDM-frame adaptation, all segments symbol data are adapted for OFDM-signal generation by single IFFT.

Figure 4.6: Example of connected transmission (three TS's)

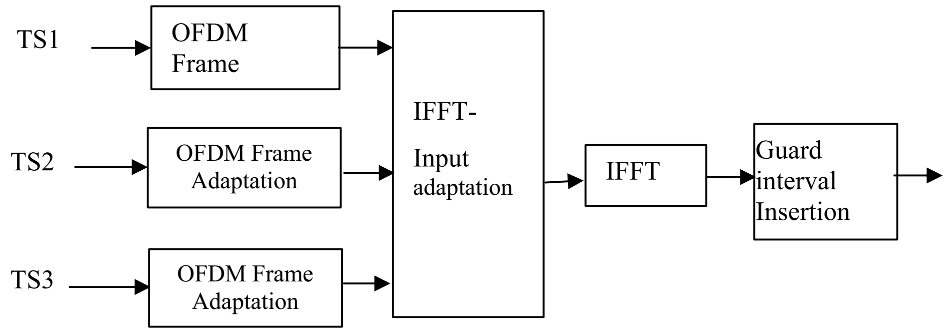


Figure 4.7: CP carrier in an ordinary transmission

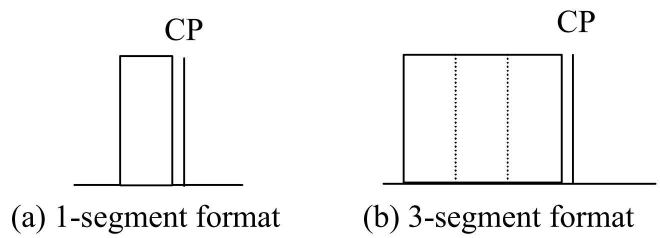
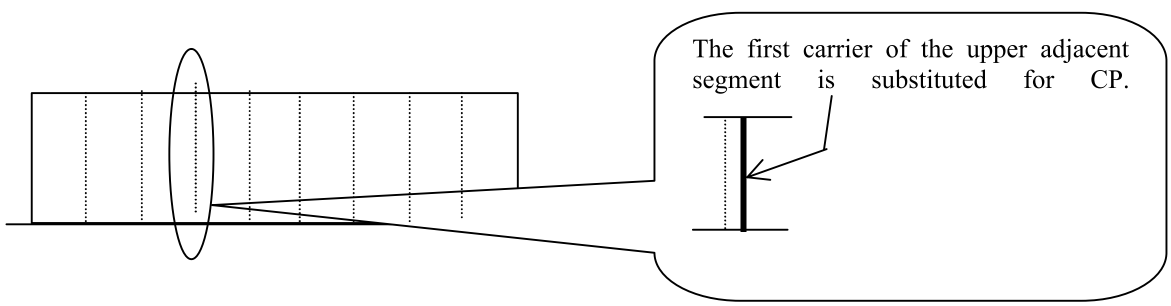


Figure 4.8: CP carrier in connected transmission



Parameter restrictions in connected transmission

The same mode should be applied for all segments. Mode means an identification of transmission mode based on the carrier spacing of OFDM carriers.

The same guard interval length must be used for segments. Because all OFDM symbols in connected transmission should be synchronised with each other, modes having different symbol lengths cannot be mixed.

(8) Transmission capacities

The transmission capacities of the single-segment and the triple-segment are shown in Table 4.3 and 4.4 respectively.

Table 4.3: Information bit rates for the single-segment transmission (Segment BW=6/14MHz)

Carrier Modulation	Convolutional Code	Information Rates (kbps)			
		Guard Interval Ratio 1/4	Guard Interval Ratio 1/8	Guard Interval Ratio 1/16	Guard Interval Ratio 1/32
DQPSK QPSK	1/2	280.85	312.06	330.42	340.43
	2/3	374.47	416.08	440.56	453.91
	3/4	421.28	468.09	495.63	510.65
	5/6	468.09	520.10	550.70	567.39
	7/8	491.50	546.11	578.23	595.76
16QAM	1/2	561.71	624.13	660.84	680.87
	2/3	748.95	832.17	881.12	907.82
	3/4	842.57	936.19	991.26	1021.30
	5/6	936.19	1040.21	1101.40	1134.78
	7/8	983.00	1092.22	1156.47	1191.52
64QAM	1/2	842.57	936.19	991.26	1021.30
	2/3	1123.43	1248.26	1321.68	1361.74
	3/4	1263.86	1404.29	1486.90	1531.95
	5/6	1404.29	1560.32	1652.11	1702.17
	7/8	1474.50	1638.34	1734.71	1787.28

Table 4.4: Information bit rates for the triple-segment transmission*[5]

Carrier Modulation	Convolutional Code	Information Rates (kbps)			
		Guard Interval Ratio 1/4	Guard Interval Ratio 1/8	Guard Interval Ratio 1/16	Guard Interval Ratio 1/32
DQPSK QPSK	1/2	0.842	0.936	0.991	1.021
	2/3	1.123	1.248	1.321	1.361
	3/4	1.263	1.404	1.486	1.531
	5/6	1.404	1.560	1.652	1.702
	7/8	1.474	1.638	1.734	1.787
16QAM	1/2	1.685	1.872	1.982	2.042
	2/3	2.246	2.496	2.643	2.723
	3/4	2.527	2.808	2.973	3.063
	5/6	2.808	3.120	3.304	3.404
	7/8	2.949	3.276	3.469	3.574
64QAM	1/2	2.527	2.808	2.973	3.063
	2/3	3.370	3.744	3.965	4.085
	3/4	3.791	4.212	4.460	4.595
	5/6	4.212	4.680	4.956	5.106
	7/8	4.423	4.915	5.204	5.361

5 In the case of the triple-segment transmission, information rate can be calculated by the combination of segment information rates.

4.3.3 Characteristics

(1) More channels

Terrestrial TV broadcasting will be discontinued in July 2011. However, in the case of radio, existing AM, FM, and SW analog services are expected to continue, thus digital radio is being positioned as an opportunity to provide more channels.

(2) Consortium

At present, digital radio broadcasting is operated by a consortium where corporations interested in digital radio broadcasting have participated. The official name of the consortium is a corporate judicial body called the Association for Promotion of Digital Broadcasting or the DRP (Digital Radio Promotion) for short. The establishment of the consortium was permitted by the Ministry of Internal Affairs and Communications. The DRP has two offices, in Tokyo and Osaka. The operation fund is provided by the member companies. Members include NHK, radio stations, TV stations, data broadcasting companies, trading companies, automakers and other companies interested in digital radio in the private sector. Over 70 organizations and companies have joined the consortium throughout Japan.

The objectives of the DRP are as follows:

- Implementation of experimental broadcasting for practical application
- Development of broadcasting services
- Research and study on trends in demand
- Promotion and spread of reception

(3) Experimental broadcasting for practical application

The DRP is the only corporate judicial body licensed. Its experimental stations are located in Tokyo and Osaka. The broadcasting facilities are owned and operated by the DRP.

4.3.4 Receivers

(1) Receiver test centre

A receiver test centre has been installed in the DRP office to check the operation of receivers and to support development efforts.

The major activities are as follows:

- To define and revise a specification for standard test streams, and print and distribute its copies
- To make connection experiment items and connection manuals, and distribute its copies

- To define a specification for transmission signals on experimental radio waves
- To operate experimental radio waves and to publicize operation schedules

(2) Trial receivers

At present, no receivers are sold in the commercial market.

Thus, different types of trial receivers were developed for the use of experimental hearing.

Trial receivers include PC-card receivers that have an antenna on the top of a PCMCIA card, portable receivers (1-segment only) for the DRP, and PDA (Personal Digital Assistant) receivers where a digital radio adapter is mounted.

(3) Receivers expected

In addition to the above mentioned trial receiver types, the following types of receivers can be expected:

- cell phone type receivers
- ordinary and smaller palmtop type receivers
- car stereo type receivers for mobiles, and so on

4.3.5 Overview of Services

Among the current services being broadcast, the following types of contents are unique to digital radio broadcasting.

(1) Multiple voice broadcasting

Listeners can choose a news item, foreign language course, cooking program, etc., in addition to multiple-language concurrent broadcasting of weather forecast and stories.

(2) 5.1 surround broadcasting

5.1 surround broadcast is being provided which includes still images and textual information linked to its programs, for example, photos during performances in a live concert.

(3) Broadcasting of simplified moving images

Actions of a DJ in a studio booth, music promotion images or so are being broadcast linked to the programs.

(4) Download service experiment

With the interactive function of cell phones and PDAs, experiments are being provided, including sales of tickets and CDs, and tallying up of questionnaires. Such experiments also include download service of music titles that were broadcast.

4.3.6 Outlook for the Future

At present, digital radio broadcast experiments for practical application are underway through providing different contents of services and operation forms.

The following subjects must be handled successfully for the spread and development of digital radio broadcasting:

- To transfer the experimental broadcasting into actual broadcasting and to expand service areas
- Early release of receivers in the commercial market
- Early start of services in major cities in Japan
- Nationwide deployment of digital radio broadcasting after 2011, when analogue TV broadcasting comes to ends and frequency re-allotments are completed

4.4 iBiquity HD Radio System

The HD Radio system developed by US-based iBiquity Digital was designed for regions where limited spectrum prevents the allocation of new frequencies for digital broadcasting. The HD Radio system allows broadcasters to simultaneously transmit an analogue and digital signal without the need for additional spectrum for the digital signal. The HD Radio system takes advantage of unused portions of the spectrum on either side of the analogue carrier (as defined by the service frequency allocation "mask") and implements frequency re-use by including digital carriers in quadrature to the existing analogue carrier. In either case, the analogue signals are in close proximity to the digital signals and great care must be taken to prevent unwanted interference between them.

The HD Radio system is designed to work in hybrid mode (compatible analogue and digital) as well as to migrate to an all-digital system once analogue radios have been largely replaced in the future. See Figures 4.10, 4.11, and 4.12.

The HD Radio system offers a number of advantages for broadcasters, consumers and regulators. The HD Radio system replicates the existing coverage patterns of each radio station thereby retaining the existing economic value of the station. Broadcasters can convert to digital broadcasts with a relatively modest investment and retain the vast majority of their existing physical plant. In addition, the introduction of the digital signal in the existing channel allows the broadcaster to retain the station's existing dial position. Because the system supports simulcast of the analogue and digital signals, consumers are able to upgrade to digital over an extended period, taking into account normal equipment replacement cycles. Regulators benefit because there is no need for spectrum allocations or licensing of new stations. However, many countries are cautious about IBOC technology because it has the potential in certain circumstances to cause some degradation to existing analogue services, particular at the edge of the existing analogue service area.

The HD Radio system offers the following features:

- CD quality audio in the FM-band and FM quality stereo audio in the AM band.[6]
- Digital coverage nearly equivalent to existing analogue coverage. In areas where the digital signal is lost, the system automatically blends to the analogue back-up signal to ensure coverage is never less than existing coverage.
- Advanced coding technologies and time diversity between the analogue and digital signals ensure a robust signal.
- The FM system has demonstrated significant robustness in the presence of severe multipath, and the AM system has demonstrated significant robustness in the presence of impulse noise.
- The FM system offers options for introducing new data services ranging from 1 to 300 kbps depending on the mode of operation.

The HD Radio system has been tested in North and South America. It is currently in operation in approximately 250 stations throughout the United States and is expected to be in use by approximately 650 stations by the end of 2005.

[6] See the Report of the National Radio Systems Committees, DAB Subcommittee Evaluation of the iBiquity Digital Corporation System Part 1 – FM IBOC, November 29, 2001 ("FM NRSC Report") and Part 2, AM IBOC, dated April 6, 2002 ("AM NRSC Report").

4.4.1 HD Radio Standards Activity

Both the AM and FM HD Radio systems have received international endorsements as well as endorsements in the United States. The AM HD Radio system is included in Recommendation ITU-R BS.1514-1, adopted October 2002 where it is classified as the "IBOC DSB System." The FM HD Radio system is included in Recommendation ITU-R BS.1114-4, adopted May 2003 where it is classified as "Digital System C." In the United States, the Federal Communications Commission (FCC) endorsed both the AM and FM HD Radio systems on October 10, 2002.[7] Moreover, the National Radio Systems Committee (NRSC), an industry standards setting body sponsored by the National Association of Broadcasters and the Consumer Electronics Association, endorsed the FM HD Radio system in a report dated November 29, 2001[8] and the AM HD Radio system in a report dated April 6, 2002.[9] The NRSC endorsement was an outgrowth of an extensive testing program of both the AM and FM HD Radio systems. The NRSC supervised independent testing of the HD Radio system in both the laboratory and in the field under a comprehensive set of conditions. The tests were designed to assess both the performance of the digital system as well as the compatibility of the digital system with existing analogue operations in the AM and FM bands. In the laboratory, the digital system was subjected to a range of conditions associated with typical broadcasts in the AM and FM band. For example, the FM system was tested in the presence of multiple forms of multipath interference as well as numerous examples of co-channel and adjacent channel interference. In the case of AM, the digital system was tested in the presence of impulse noise in addition to the typical co-channel and adjacent channel interference associated with the AM band.

Field tests were conducted using commercial AM and FM stations selected for their characteristics in terms of interference from adjacent channel stations as well as to represent a variety of antenna and implementation configurations. For both the laboratory and field tests, objective measurements were recorded and considered in the evaluation process. In addition, thousands of audio samples were produced and used to conduct an extensive subjective evaluation process. General population listeners were asked to rate a variety of sound samples from the laboratory and field tests to assess the real world response to the introduction of the HD Radio system. The test results demonstrated that the HD Radio system consistently outperformed existing analogue AM and FM radio. Moreover, the tests established that the introduction of the HD Radio system will not cause harmful interference to existing analogue broadcasts in the vast majority of cases. In those cases where new interference is expected to occur, it is expected that new interference will be most common in peripheral areas outside the core coverage areas of a station. The NRSC concluded that this minimal risk of additional interference is more than outweighed by the improved audio quality and performance that the HD Radio system repeatedly demonstrated throughout the test programme.[10]

4.4.2 HD Radio AM and FM Receivers

HD Radio receivers are inherently simpler and lower cost than new band receivers since much of the circuitry required for the digital signals is common to that used to process the

[7] Digital Audio Broadcasting Systems And Their Impact on the Terrestrial Radio Broadcast Service, MM Docket No. 99-325, First Report and Order (October 10, 2002).
[8] DAB Subcommittee Evaluation of the iBiquity Digital Corporation IBOC System Part 1 – FM IBOC dated November 29, 2001 ("NRSC FM Report").
[9] DAB Subcommittee Evaluation of the iBiquity Digital Corporation IBOC System Part 2 – AM IBOC dated April 6, 2002 ("NRSC AM Report").
[10] *See* NRSC FM Report at 9; NRSC AM Report at 8.

existing analogue signal. As a result, HD Radio receivers are expected to cost no more than 20% more than existing analogue receivers.

Figure 4.9 - Typical HD Radio Automobile Receivers

The first phase of the HD Radio receiver roll out is focusing on automobile and home hi-fi receivers. Aftermarket automobile receivers and home receivers began reaching the market in early 2004. OEM automobile receivers are scheduled to be introduced for the automobile model year 2006, which should be launched in the third quarter of 2005. Figure 4.9 shows fully functional automobile receivers designed to fit into the standard car mount frames.

Along with the introduction of second and third generation HD Radio semiconductors, featuring lower power consumption and cost during the two or three years after the initial receiver introduction, portable and lower cost receiver products are expected to be introduced.

4.4.3 HD Radio System Technical Design Overview

The HD Radio system is designed to permit a smooth evolution from current analog Amplitude Modulation (AM) and Frequency Modulation (FM) radio to a fully digital In-Band On-Channel (IBOC) system. This system can deliver digital audio and data services to mobile, portable, and fixed receivers from terrestrial transmitters in the existing Medium Frequency (MF) and Very High Frequency (VHF) radio bands. The system is designed to allow broadcasters to continue to transmit analog AM and FM simultaneously with new, higher-quality and more robust digital signals, allowing broadcasters and their listeners to convert from analog to digital radio while maintaining each station’s current frequency allocation.

The HD Radio system allows a broadcast station to offer multiple services. A service can be thought of as a logical grouping of application data identified by the HD Radio system. Services are grouped into one of two categories:

Core Services:

- Main Program Service (both Audio (MPA) and Data (PAD))
- Station Information Service (SIS)
- Advanced Application Services (AAS)

The flow of service content through the HD Radio broadcast system is as follows:

- Service content enters the HD Radio broadcast system via Service Interfaces;
- Content is assembled for transport using a specific protocol;
- It is routed over logical channels via the Channel Multiplex; and
- It is waveform modulated via the Waveform/Transmission System for over-the-air transmission.

The system employs coding to reduce the sampled audio signal bit rate and baseband signal processing to increase the robustness of the signal in the transmission channel. This allows a high quality audio signal plus ancillary data to be transmitted in adjacent frequency partitions and at low levels that do not interfere with the existing analog signals.

4.4.4 Core Services

(1) Main Program Service (MPS)

The Main Program Service (MPS) is a direct extension of traditional analog radio. MPS allows the transmission of existing analog radio-programming in both analog and digital formats. This allows for a smooth transition from analog to digital radio. Radio receivers that are not HD Radio enabled can continue to receive the traditional analog radio signal, while HD Radio receivers can receive both digital and analog signals via the same frequency band. In addition to digital audio, MPS includes digital data related to the audio programming. This is also referred to as Program Associated Data (PAD).

(2) Station Information Service (SIS)

The Station Information Service (SIS) provides the necessary radio station control and identification information, such as station call sign identification, time and location reference information. SIS can be considered a built-in service that is readily available on all HD Radio stations. SIS is a required HD Radio service and is provided dedicated bandwidth.

(3) Advanced Application Services (AAS)

Advanced Application Services (AAS) is a complete framework in which new applications may be built. In addition to allowing multiple data applications to share the Waveform/ Transmission medium, AAS provides a common transport mechanism as well as a unified Application Programming Interface (API). On the transmission side, broadcasters utilize the common AAS interface to insert service(s) into their signal; receiver manufacturers utilize the AAS ‘toolkit’ to efficiently access these new services for the end-user. AAS includes separate audio programming such as reading services and other secondary aural and data services.

(4) Waveforms and Spectra

The HD Radio system provides a flexible means of transitioning to a digital broadcast system by providing three new waveform types: Hybrid, Extended Hybrid, and All Digital. The Hybrid and Extended Hybrid types retain the analogue FM signal, while the All Digital type does not.

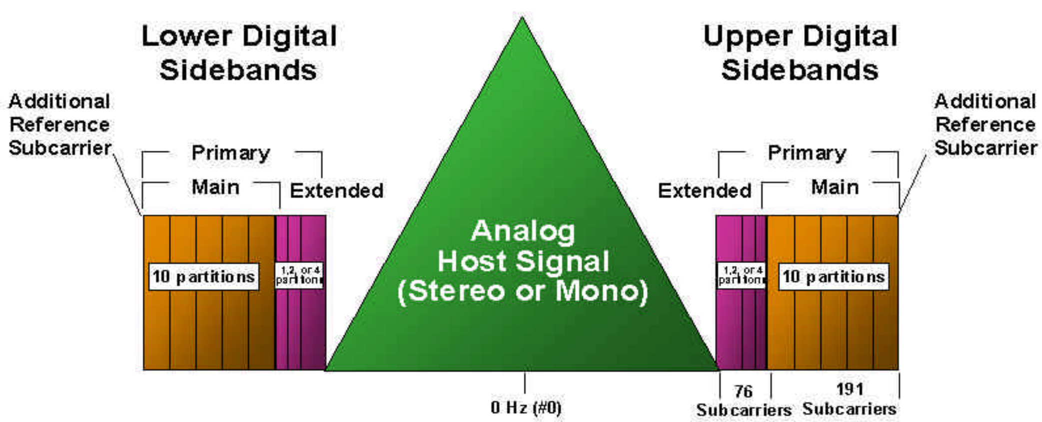
All three waveform operate well below the allocated spectral emissions mask as currently defined by the United States Federal Communications Commission.

The digital signal is modulated using orthogonal frequency division multiplexing (OFDM). OFDM is a parallel modulation scheme in which the data stream modulates a large number of orthogonal subcarriers, which are transmitted simultaneously. OFDM is inherently flexible, readily allowing the mapping of logical channels to different groups of subcarriers.

(5) Hybrid Waveform

The digital signal is transmitted in sidebands on either side of the analog FM signal in the Hybrid waveform. The power level of each sideband is approximately 23 dB below the total power in the analog FM signal. The analog signal may be monophonic or stereo, and may include subsidiary communications authorization (SCA) channels. See Figure 4.10.

Figure 4.10: Hybrid spectrum allotment of FM HD Radio system



Extended Hybrid operation involves use of us to four Extended Partitions in addition to the 10 Main Partititions.

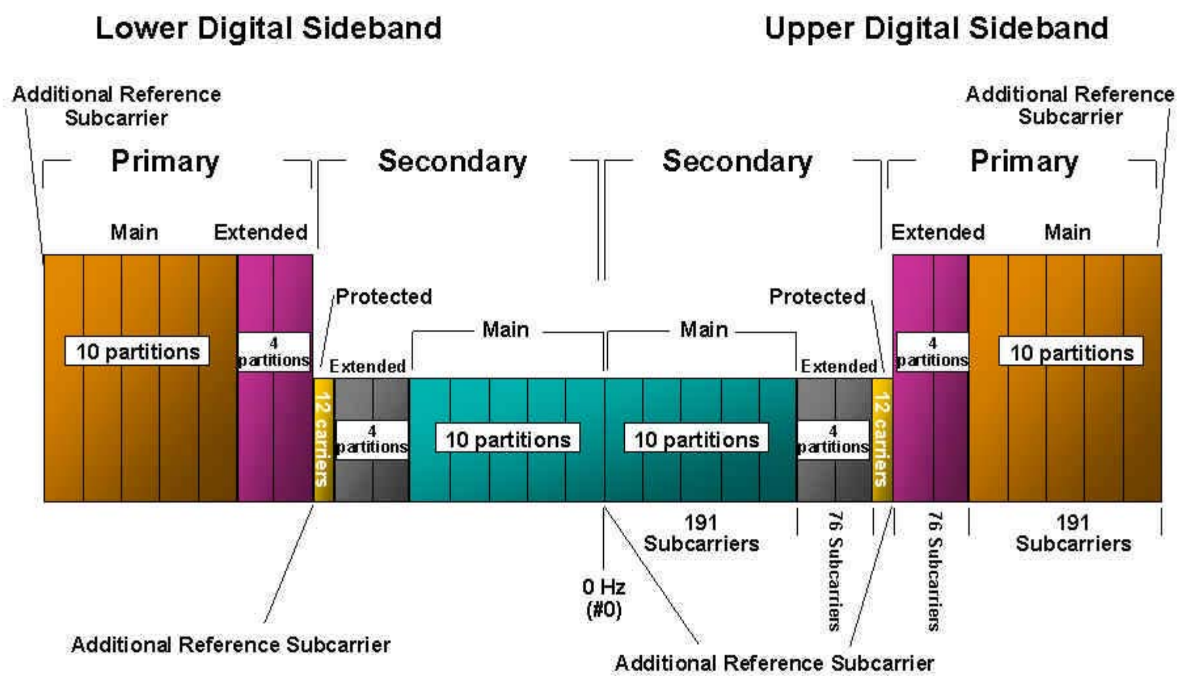
(6) FM Extended Hybrid Waveform

In the Extended Hybrid waveform, the bandwidth of the Hybrid sidebands can be extended toward the analogue FM signal to increase digital capacity. This additional spectrum, allocated to the inner edge of each primary sideband, is termed the primary extended sideband. See Figure 4.10.

(7) FM All Digital Waveform

The greatest system enhancements are realized with the All Digital waveform, in which the analogue signal is removed and the bandwidth of the primary digital sidebands is fully extended as in the Extended Hybrid waveform. In addition, this waveform allows lower-power digital secondary sidebands to be transmitted in the spectrum vacated by the analogue FM signal. Approximately 300 kbps of data is available in All Digital mode. See Figure 4.11.

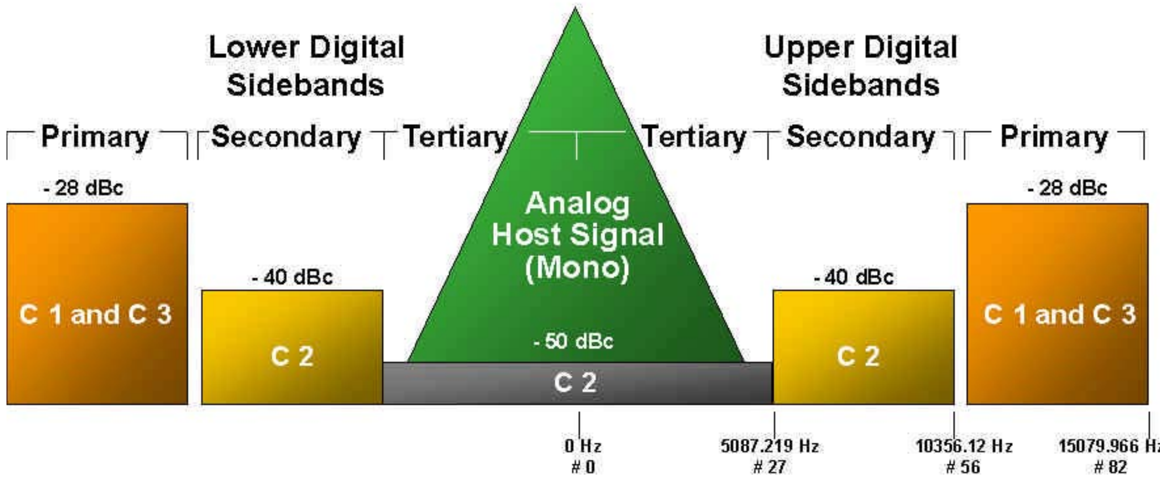
Figure 4.11 - All digital spectrum allotment of FM HD Radio system



(8) AM Hybrid and All Digital Waveforms

Unlike the FM HD Radio system, the AM system contains no extended hybrid capacity. The allocation scheme is represented in Figure 4.12.

Figure 4.12: Hybrid AM HD Radio system spectrum allotment

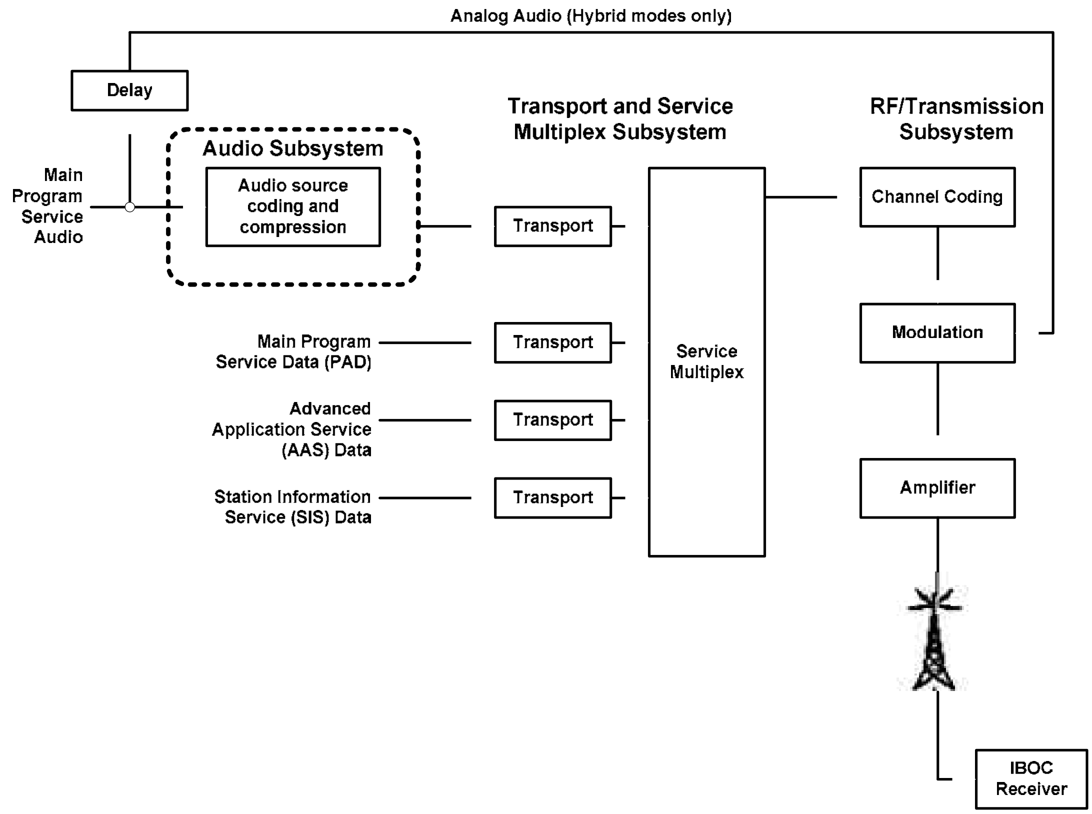


4.4.5 HD Radio Subsystems

A basic block diagram representation of the system is shown in Figure 4.13. It represents the HD Radio digital radio system as three major subsystems:

- Audio source coding and compression
- Transport and Service Multiplex
- RF/Transmission

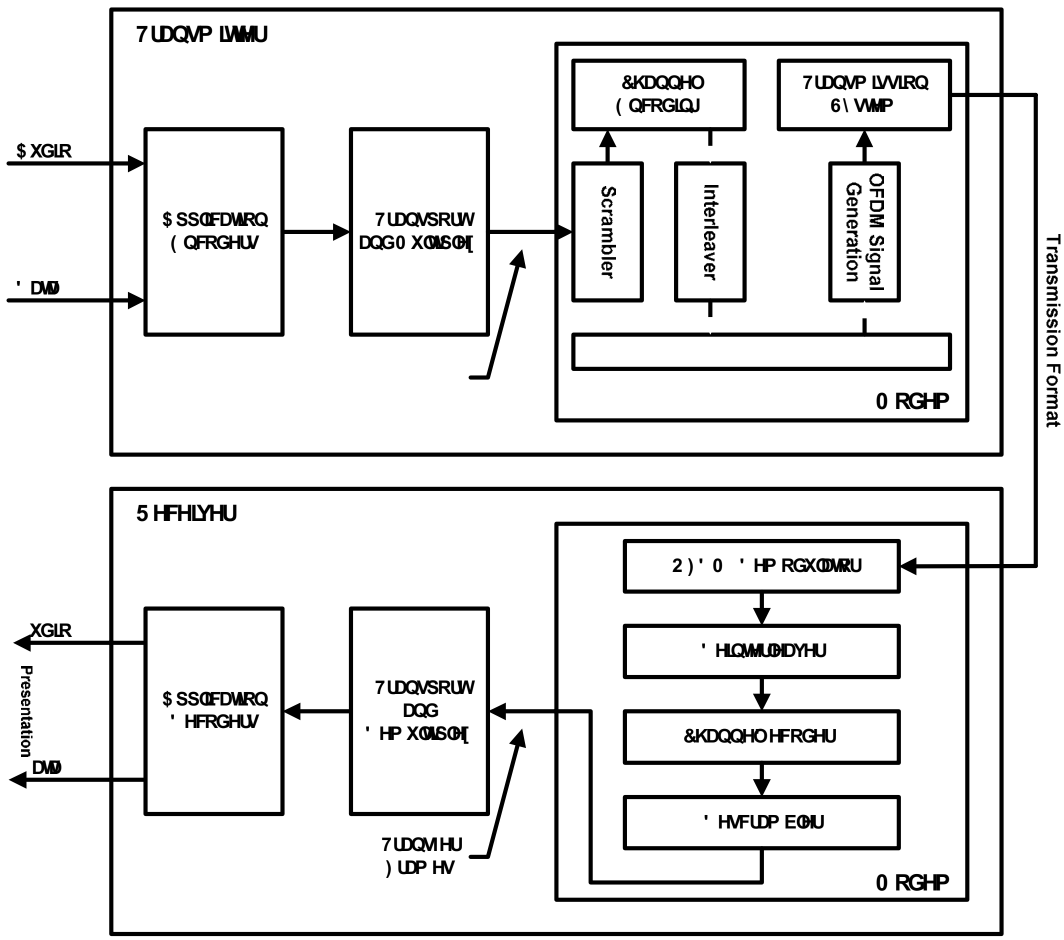
Figure 4.13: Functional Block Diagram of HD Radio System



(3) RF/Transmission System

"RF/Transmission" refers to channel coding and modulation. The channel coder takes the multiplexed bit stream and applies coding and interleaving that can be used by the receiver to reconstruct the data from the received signal, which because of transmission impairments, may not accurately represent the transmitted signal. The processed bit stream is modulated onto the OFDM subcarriers that are transformed to time domain pulses, concatenated, and up-converted to the FM band.

Figure 4.14: Block diagram of HD Radio transmission and reception multiplexing

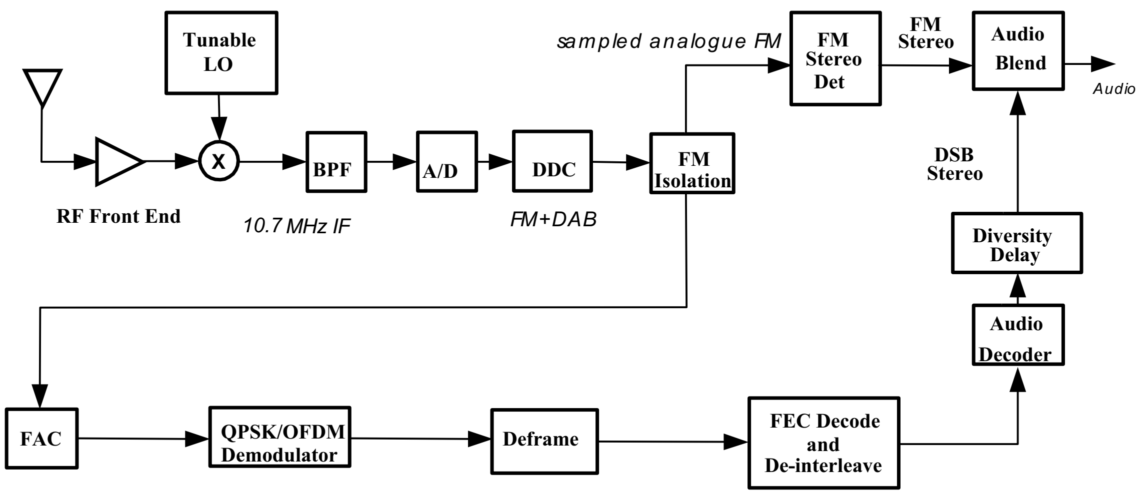


4.4.6 Receiver Systems

A functional block diagram of an HD Radio receiver is presented in Figure 4.15. The signal is received by the antenna, passed through an RF front end, and mixed to an intermediate frequency (IF), as in existing analogue receivers. Unlike typical analogue receivers, however, the signal is then digitised at IF, and digitally down-converted to produce in-phase and quadrature base band components. The hybrid signal is then separated into an analogue component and a digital component. The analogue FM stereo signal is digitally demodulated and demultiplexed by the FM receiver to produce a sampled, stereo audio signal.

The base band digital signal is first sent to the modem, where it is processed by the First Adjacent Cancellation system to suppress interference from potential first-adjacent analogue FM signals. The signal is then OFDM demodulated, deframed, and passed to the FEC decoding and deinterleaving function. The resulting bit stream is processed by the codec function to decompress the source-encoded digital audio signal. This digital stereo audio signal is then passed to the blend function.

Figure 4.15: FM hybrid IBOC receiver functional block



4.4.7 Features Common to North American Digital Radio Systems

(1) Sound Quality

Sound quality of digital radio systems has improved dramatically in recent years with progressively lower bitrates being shown in various applications as achieving near CD quality. Rates well below 96 kbps are routinely utilized in digital radio systems in operation in North America and meeting with wide customer acceptance.

(2) Multipath Resistance

OFDM based systems are made to be resistant to multipath within a guard interval. In the case of the Eureka system, the guard interval is set to 62 ?s (18.6 km at the speed of light). This means that any echoes coming from up to 18.6 km will be considered as constructive to the signal. This allows the use of on-channel repeaters (that are treated as active echoes).

Note also that some systems, such as the Eureka system, also use unequal error protection and error concealment techniques. This allows for a graceful degradation of the digital signal quality when fading occurs and allows for S/N requirement reductions for the receiver. The Eureka system is especially noted for achieving multipath free reception, but narrower bandwidth systems such as the

HD Radio system have also been shown to be multipath-free even in challenging propagation conditions.

(3) Frequency Response

It is difficult to evaluate the exact frequency response of a codec, because it will change dynamically depending on the available bit rate and the difficulty of encoding the instantaneous audio material. A quick example can be demonstrated using a single carrier frequency sweep on any codecs. Typically total frequency response of 20 kHz is measured in such tests, even at 16 kbps. On the other hand, encoding a rich stereo program on the same codec at 16 kbps, may result in a pure monophonic signal of less than 5 kHz.

Consequently, the codec has to be tailored to the program content being broadcast. See Table 4.5 for MPEG Layer II implementation recommendations.

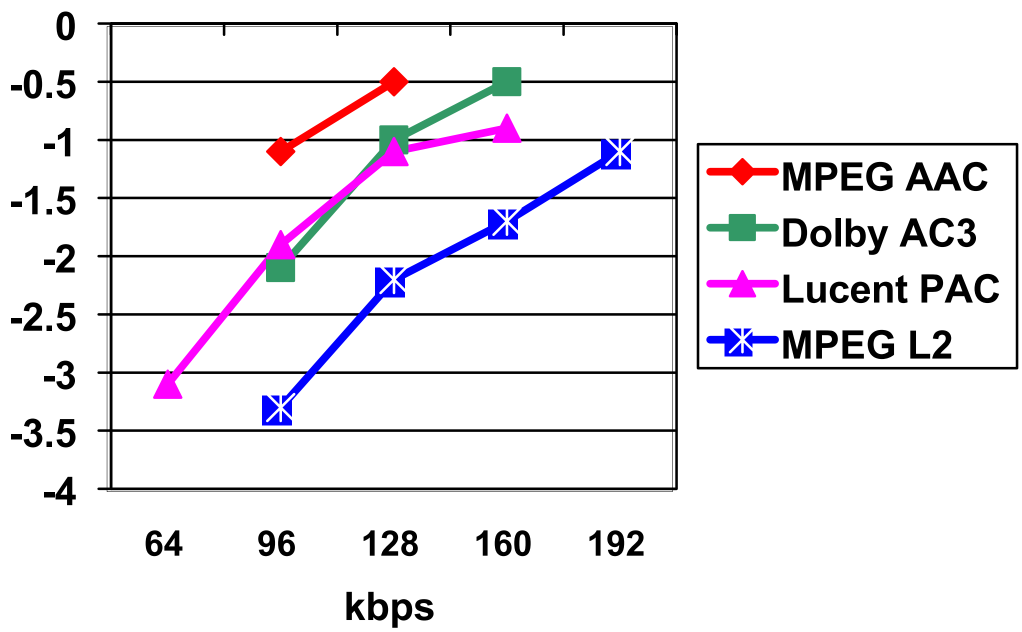
Table 4.5: Recommended MPEG II data and sampling rates for various program material

Voice program:	48 kbps . 24 KHz Sampling rate / Mono
Mono program with music:	80 kbps / 48 KHz Sampling rate / Mono
Oldies music:	112 kbps / 48 KHz / Joint Stereo

(4) Audio Quality Ratings

The basic audio quality is defined by the codec used and the ruggedness of the transmission channel. The performance quality of different codecs when critical material is encoded has been measured as shown in Figure 4.16.

Figure 4.16



Source: Journal of the Audio Engineering Society, Vol. 46, No. 3, March 1998, p. 164.

Where the signal quality refers to the Subjective Difference Grade ITU scale (ITU-R BS.562) as follows:

0 = Imperceptible

-1 = Perceptible but not annoying

-2 = Slightly annoying

-3 = Annoying

-4 = Very annoying

Not shown in the previous graph is the usage of the Sub-band replication (SBR) technique. This technique allows for a higher coding efficiency by coding the lower bands (up to 8 kHz to 12 kHz) using a basic codec (MPEG AAC, L2 or others) and replicating the upper frequency using statistical and predictive information. This technique usually demonstrates an increase in efficiency of up to 30% and it is generally backward compatible with the source codec. The iBiquity Digital HDC audio codec relies on SBR technology in achieving good results at 96 kbps and lower.

4.4.8 Infrastructure Requirements

At this writing, approximately 250 HD Radio stations have commenced operations in the United States with an additional 300 stations currently licensed to commence operations in the coming year. Consumer receivers went on sale in early 2004 and considerable work has been done on implementation options that may significantly reduce installation costs at many stations.

In particular, the use of separate antenna radiating systems has been successfully demonstrated which can eliminate the need for combiners at higher power FM stations.[11] Furthermore development of a "Gen-2" system[12] that will multiplex the HD Radio data stream at the studio site, much as is done with composite analogue STLs, is expected to be introduced in the coming year. Low-level combining is preferred and has been shown to be cost effective at power levels up to an analogue 7 kw transmitter power output.

[11] *See IBOC Space Diversity Testing*, Talmadge Ball, Proceedings of NAB 2003 BEC; also see "Dual Antenna Report" Denny & Associates at http://www.nab.org/scitech/fccfilingattachmentb.pdf.

[12] Gen 2 refers to a planned evolution in HD system design that will permit multiplexing of all digital signals at the studio to achieve significant reductions in feeder link bandwidth requirements.

Deployment Status

Current deployment statistics for the HD Radio System in the United States are shown below.

Figure 4.17: Deployment status of HD Radio in the United States as of February 2005

570 Licensees	
140 Markets	
46 Top 50 Markets	
212 Licensed Groups	
18 Licensed Top 20	
49 States Serviced*	
* Includes Washington, DC and Puerto Rico	
Population Served	198,000,000
Listeners Served	32,000,000

250 On The Air	
71 Markets	
37 Top 50 Markets	
38 States Served	
Population Served	150,000,000
Listeners Served	23,000,000

Rnk	Market	#	On
1	New York	10	6
2	Los Angeles	19	13
3	Chicago	11	11
4	San Francisco	13	10
5	Dallas	5	1
6	Philadelphia	12	7
7	Houston	3	1
8	Washington, DC	4	4
9	Boston	15	10
10	Detroit	17	15
11	Atlanta	16	11
12	Miami	18	8
13	Puerto Rico	6	0
14	Seattle	10	9
15	Phoenix	0	0
16	Minneapolis	6	4
17	San Diego	0	0
18	Nassau-Suffolk	2	0
19	Baltimore	4	1
20	St. Louis	5	4
21	Tampa	1	1
22	Denver	17	9
23	Pittsburgh	1	0
24	Portland	12	6
25	Cleveland	4	2

Rnk	Market	#	On
26	Cincinnati	10	8
27	Sacramento, CA	5	0
28	Riverside, CA	3	0
29	Kansas City, MO/KS	6	5
30	San Jose, CA	3	2
31	San Antonio, TX	0	0
32	Salt Lake City	2	0
33	Milwaukee	2	2
34	Providence, RI	2	0
35	Columbus, OH	3	2
36	Middlesex, NJ	3	0
37	Charlotte, NC	3	0
38	Orlando, FL	4	2
39	Las Vegas	3	2
40	Norfolk, VA	3	1
41	Indianapolis	12	9
42	Austin, TX	2	1
43	Greensboro, NC	0	0
44	New Orleans	4	2
45	Nashville	4	2
46	Raleigh-Durham	3	1
47	West Palm Beach, FL	4	3
48	Memphis	1	1
49	Hartford, CT	5	1
50	Jacksonville, FL	2	1

Courtesy: iBiquity Digital

4.5 Issues related to Terrestrial Systems

4.5.1 Spectrum Availability

Government policies on frequency management and spectrum pricing affect all radio broadcasting development. In some countries the radio spectrum is looked upon as a means of raising revenue.

Administrations in different countries operate different policies, but there are common threads. Whether frequencies are allocated through auctions or by other means, spectrum is undeniably a scarce resource and especially so in the bands that are most useful for digital audio broadcasting (including the existing shortwave bands, existing AM and FM bands, as well as the upper VHF and L-Bands).

In many countries administrations have allocated spectrum for digital broadcasting, both radio and TV.

The DRM system is designed to work within the existing band and channel structure for all the broadcasting bands below 30 MHz. As such, its use of the spectrum conforms to the Geneva Treaty of 1975 for ITU-R Regions 1 and 3, the long wave and medium wave channels have a 9 kHz bandwidth, or multiples thereof, depending on the channel assignment; the Rio Treaty of 1981 for Region 2 medium wave specifies a 10 kHz bandwidth; and the shortwave channel bandwidth is 10 kHz for all the HF broadcasting bands. Thus, no new spectrum is required. Furthermore, based upon ITU-R decisions during 2003, DRM signals can be used operationally in these bands, with the existing channel bandwidths, interspersed with the analogue broadcasts. That is, there are no specially allocated segments of bands for digital transmissions. Ongoing testing has verified the feasibility of this approach.

In the US, the government has approved HD Radio as a way to alleviate the need for new spectrum to implement terrestrial digital radio.

The constraints and uncertainties that cloud the issue of frequency allocations for new digital terrestrial services in the VHF and L-Bands are not such a problem for AM digital developments. There is some prospect that the congestion now in the AM bands could be reduced with digital broadcasting. Potentially, there is much to be gained from digital broadcasting in the short-wave bands because current analog systems require a number of simultaneous broadcasts to ensure reliable reception under changing ionospheric conditions.

Case Study: Allocations in Region 1

In the UK, where spectrum is being allocated for seven Eureka 147 DAB multiplexes, the granting of license has been in VHF Band III, which is very suitable for terrestrial DAB (T-DAB) transmissions. Across Europe, both VHF and L-Band frequencies will be used for T-DAB services. At a planning meeting held set up by the CEPT (European Conference of Postal and Telecommunications Administrations and held in Wiesbaden) in 1995, frequency blocks in three bands were considered:

- VHF Band I (47 – 68 MHz)
- VHF Band III (174-240 MHz)
- L-Band (1452 – 1467.5 MHz)

The Wiesbaden plan made allotments for digital audio broadcasting in VHF channels 11 and 12 and in the L-Band, and considered the implications of protecting non-DAB services within the planning area. These include airborne military services and television services in the VHF bands as well as fixed and aeronautical telemetry in the L-Band.

Overall, there were sufficient allotments made in the Wiesbaden plan for the initial needs of DAB, but looking ahead, additional frequency allocations will be needed in Europe. Most organisations planning to launch today and expand T-DAB services favour VHF frequencies.

The position on T-DAB frequencies in other parts of the world is similarly complicated and underlines the point that frequency allocation is an outstanding issue that will remain high on the DAB agenda for some time to come.

At the International Telecommunications Union (ITU) in Geneva, the Regional Radiocommunications Conference 2006 (RRC-06) took place from 15 May to 16 June 2006. The new agreement, GE06, includes the frequency plans for T-DAB and DVB-T in Band III and for DVB-T in Bands IV/V for Region 1 (parts of Region 1 to the west of meridian 170°E and to the north of parallel 40°S) and in the Islamic Republic of Iran (see Figure 4.18 below).

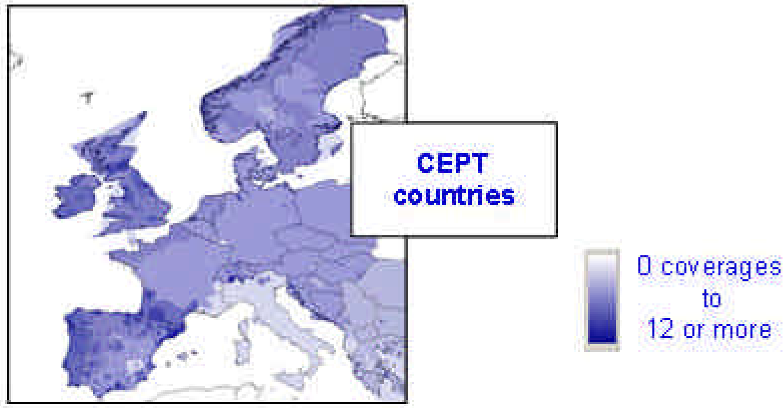
Figure 4.18: RRC-06 planning area

The table below shows the results of the RRC-06. The results are evaluated with regard to the proportion of the assigned requirements relative to the submitted ones.

	Band III		Bands IV/V
	T-DAB	DVB-T	DVB-T
Total	8817	7411	56 533
Assigned	8379	6703	55 409
% Assigned	95.0%	90.4%	98.0%

The planned allotments and assignments for T-DAB in part of the planning area centred around Europe are shown in Figure 4.19 below.

Figure 4.19: T-DAB coverages in Band III



The number of coverages can be estimated by analyzing the coverage maps taking into consideration, when relevant, the overlapping areas between allotments or assignment areas. The table below shows the estimated number of coverages (distinguishing between nationwide coverage and partial coverage) for the CEPT counties.

	Estimated number of coverages T-DAB in Band III (in CEPT)	
	Nationwide	Partial
Average	2.4	1.7
Median*	3.0	1.0
Max	5.0	9.0

**Median: 50 % of the countries have this number or more*

The above table shows that in the majority of the European countries, within CEPT, obtained 3 nationwide coverages for T-DAB and 1 additional partial coverage.

An estimation of the channel usage in Band III for T-DAB is shown in Figure 4.20 below.

Figure 4.20: Estimation of the channel usage in Band III for T-DAB (RPC4: suitable for mobile reception; RPC5: suitable for portable indoor reception)

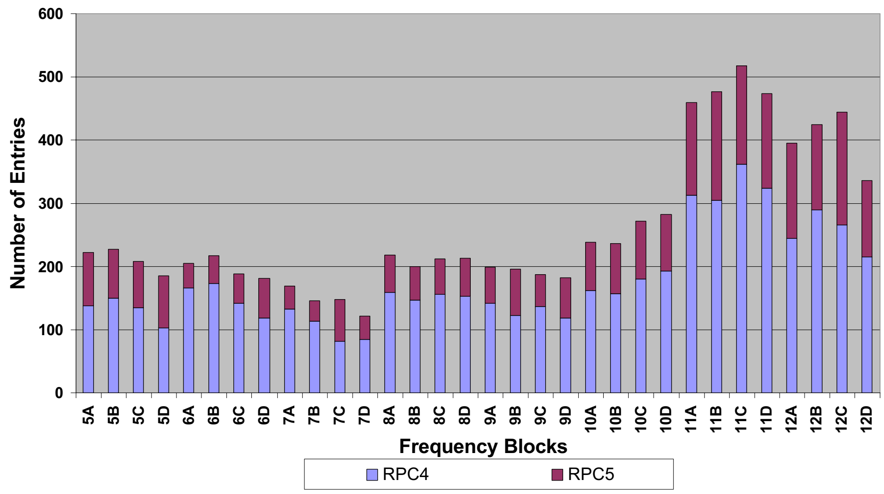
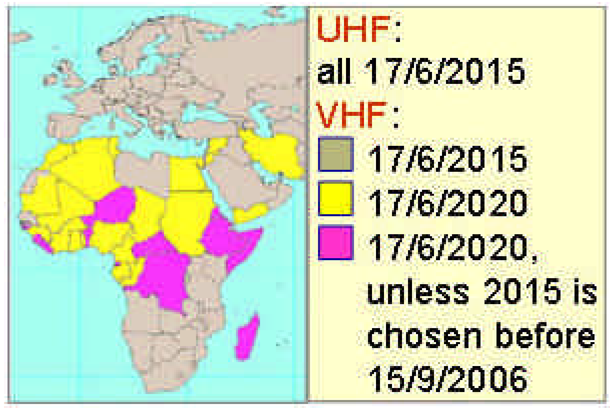


Figure 4.20 shows that T-DAB mobile reception (RPC4) represents the major proportion of the T-DAB requirements and that channels 11 and 12 are the most used for T-DAB.

The GE06 Agreement offers a great deal of flexibility for using a digital entry in the plan for another application provided that the peak power density in any 4 kHz is respected. Such flexibility can allow, for example, for using a DVB-T entry by 4 x T-DAB entries or by 4 x T-DMB entries and also can allow for accommodating future developments of digital technology.

The end of the transition period for Band III has been fixed to 2015 (for some non-European countries is 2020, see details in Figure 4.21). During that period analogue television has to be protected. Around 25% of the T-DAB entries in the new plan have to be coordinated with analogue television in neighbouring countries before implementation. As a consequence, certain constraints (time constraints, power reduction, particular antenna patterns, etc.) might be imposed to those T-DAB requirements during the transition period. In addition, around 7% will have to coordinate with other digital requirements and around 2.5% with other primary services (e.g., PMR - Private Mobile Radio).

Figure 4.21: End of the transition period



In summary, the full potential of the new digital plans will not be available until the analogue switch off.

In terms of propagation performance at L-Band using T-DAB, concern about the efficiency of the 1.5 GHz Band has been largely removed as a result of extensive technical evaluation and field measurements, mainly by Canada's Communications Research Centre (CRC). It was found that indoor reception at L-Band is comparable to that achieved at VHF frequencies. The reason is that the shorter wavelength at L-Band offsets the increased attenuation through walls at the lower VHF frequencies.

4.5.2 The Implications of Simulcasting

Whilst the benefits of digital broadcasting and the opportunities offered by this technology are clear to broadcasters, there is concern about the time and cost implications of the transition from analogue to digital. Until the coverage from digital broadcasts matches that from existing FM and AM services, it is unrealistic to cut existing transmissions and disfranchise listeners. It could be some years before the new digital services provide comparable coverage and a receiver base is established. Only then can the analogue services be closed down.

The transition from analogue to digital is helped in many countries by cooperation between public and private broadcasters, that jointly develop the necessary infra-structure and create attractive new programmes, and suitable regulatory arrangements. Examples of such cooperation exist in Canada, Sweden, the UK, France, Italy, etc.

(1) HD Radio (IBOC)

IBOC transmission schemes are particularly well suited for ensuring a smooth transition to digital services. Since they are designed for compatibility with the existing analog signals, there is little or no disenfranchisement of listeners at the onset of service. New receiver costs are minimised since much of the existing circuitry can be shared by the analogue and digital portions of the receiver. And over time, as IBOC receiver penetration reaches a "critical mass," individual broadcasters can be expected to have significant flexibility in determining when and how to phase out the analogue portion of the IBOC signal all together.

In addition, the simulcasting of audio material in some IBOC systems, while done primarily to facilitate time diversity, can also mitigate the undesired digital receiver behaviour experienced in cases of severe signal obstructions or extreme cases of interference. In these cases, systems without time diversity (such as Eureka 147) exhibit what is called a "cliff effect" failure, in that the audio signal is perfect one second, and completely gone ("muted") the next. In a simulcast IBOC system, the existence of the "backup" analogue signal for purposes of time diversity has the added effect of eliminating the cliff effect failure mode, since in those cases the receiver will blend to analogue and the audio program, while degraded, will not go away all together, and is likely to remain with the listener throughout the impairment.

These developments are at a relatively early stage and their viability has to be assessed, but the work carried out to date is encouraging. The audio quality achievable with simulcasting remains to be established.

(2) DRM (Digital Radio Mondiale)

Two types of simulcast are present in the DRM design. The first is confined to a 9 or 10 kHz channel. Half the channel is used for an analogue signal capable of envelope detection (in order that a conventional AM radio receiver can demodulate the signal). The other half is a DRM digital signal that requires digital demodulation. The second technique requires 18 or 20 kHz of 2 adjacent channels where one channel contains standard AM and the other contains either a 4.5/5 or 9/10 kHz DRM signal.

For Regions 2 and 3 the simulcast solution is potentially much simpler as the Long and Medium Wave bands have been allocated 18/20 KHz channels. In Region 3 the 18 kHZ allocation is also protection against night time sky wave interference.

4.5.3 Coverage

The move from analogue to digital transmission raises important questions under the heading ”coverage.”

One of the main differences between analogue and digital broadcasts is the mode of failure when the received signal starts to fail. It happens at the edge of the service area and at locations within the coverage footprint where the signal strength is affected by shadowing or interference. When the signal strength reduces, analogue reception is often described as degrading “gracefully.” By contrast, a digital signal will at some point fail suddenly and completely. Whilst usually robust in areas of generally poor analogue reception, the digital signal gives little indication as it approaches a point of failure.

Within a defined coverage area, the service availability from analogue and digital services will be affected by the type of receiver (fixed, mobile or portable), by the type of environment (urban, rural), and by the topography. It is also a function of the transmission frequency and the system performance.

COFDM signals (such as those used in the Eureka 147, DRM and AM and FM IBOC schemes) have characteristics which facilitate the planning of single frequency networks

(SFN's) to a greater or lesser extent and make it easier to extend coverage. Provided that a broadcast on the same frequency from a different transmitter, or a reflected transmission from the main transmitter, arrive at the receive antenna within the system's guard interval, the reflected signal will combine in a constructive way to reinforce reception.

One of the objectives for the Eureka 147 system was to transmit a digital signal (a number of digitised analogue radio programmes plus data) to a mobile receiver over a difficult transmission channel. Extensive testing has confirmed that this requirement has been achieved successfully. The same characteristics of Eureka 147 ensure much more rugged reception on portable receivers.

Recent development of IBOC systems in the US has also emphasised robust performance in a multipath fading channel. Using sophisticated signal processing techniques such as Complementary Punctured Coding, along with time and frequency diversity, the next-generation IBOC systems are expected to exhibit fading channel performance commensurate with that achieved in the Eureka 147 system, but this remains to be demonstrated.

Tests and operational broadcasting have shown that DRM coverage is equivalent to the corresponding analogue service it is replacing. Coverage, in this sense, refers to the intended broadcast area, wherein the digital signal retains its high audio quality.

5 Satellite Transmission

The Broad Picture

For many years, satellites in geostationary orbits (GSO's) have been used successfully by broadcasters for distributing programmes and services from the originating studios to terrestrial transmitting stations. It is cost effective and reliably delivers high quality signals to each transmitting station. This method of distribution is of particular benefit to international broadcasters that in the past relied on SSB and DSB short-wave signals for feeds to remote relay stations.

Today, radio broadcast services can be included in the bandwidth used for an FM satellite television signal, or as part of a bundle of digital channels as used in Astra Digital Radio. In all cases, satellite tuners are needed to receive these broadcasts, which are usually transmitted in the Ku Band.

Direct radio broadcasts from geostationary satellites to fixed receivers with externally mounted line-of-sight antennas is routine and presents no problems. It is a much more demanding requirement to reach receivers that are mobile or portable, but the majority of radio listeners have radio receivers of this type. Any radio transmission system, terrestrial or satellite, which fails to deliver a satisfactory service to such receivers will probably not find widespread acceptance.

The main difficulty in providing a satellite broadcast to an audience on the move is occasional blockage by buildings etc. This can reduce the signal by 10 to 20 dB, which it is impractical to compensate for with an increased link margin.

5.1 WorldSpace – ITU-R System D

WorldSpace is a commercial organisation based in Washington D.C. with world-wide interests. It has planned for three geo-stationary satellites, named AfriStar, AsiaStar and AmeriStar to provide global coverage. They will have L-Band payloads and each satellite covers its designated target area with three "spot" beams. Each beam has two transponders (one transparent and one with on-board processing). The aim is to provide digital radio and ancillary services to audiences in the footprints of these satellites using ITU-R Digital System D. As the names imply, the continental zones to be served by these satellites are Africa, Asia and Central and South America. See Figure 5.1.

The primary aim of the original WorldSpace concept is to provide a simple radio service, but as the project has developed, there is now more emphasis on mobile and multimedia features involving data and image transmission. Trials of MPEG-4 video have been successfully completed recently.

The WorldSpace project is innovative and has a number of points in its favour. These include the size of the coverage areas in relation to the cost of the satellites, advanced low bit-rate audio coding and straight forward satellite uplinking arrangements. An enhanced service using terrestrial repeaters for reliable mobile reception is has been successfully trialled and is being planned for introduction soon.

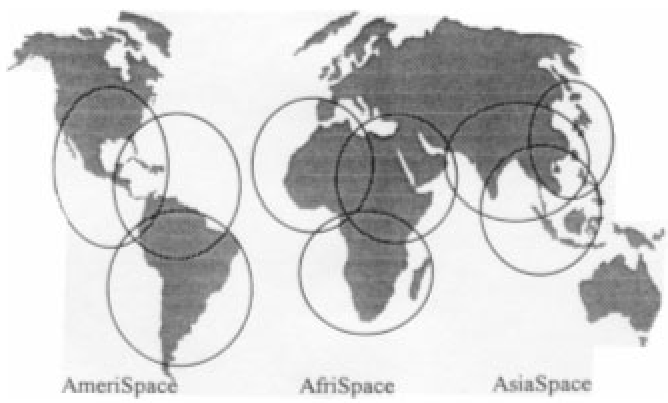
The approximate target regions for the transmissions from each satellite in the WorldSpace system are shown in Figure 5.2. AfriSpace commenced operation in 1999 and AsiaSpace in 2000. AmeriSpace has not been launched. Consideration is also being given to planning an additional satellite for providing service for Europe.

WorldSpace has successfully launched several public benefit services including:

- General Distance Education (e.g. CLASS)
- Specialised Medical Practitioner Education
- Science Promotion
- Health Awareness
- Empowerment (Women and Girls, Business, etc.)

Satellite DSB can also be used to provide a reliable emergency and disaster warning broadcast system. Before the Asian Tsunami in December 2004, WorldSpace had successfully trialed a cyclone warning system for Indian Fishermen. After the Asian Tsunami WorldSpace has been working with affected countries and aid agencies to provide disaster relief and rehabilitation services in India and Indonesia.

Figure 5.1. WorldSpace Coverage Map (Transmission Footprints)



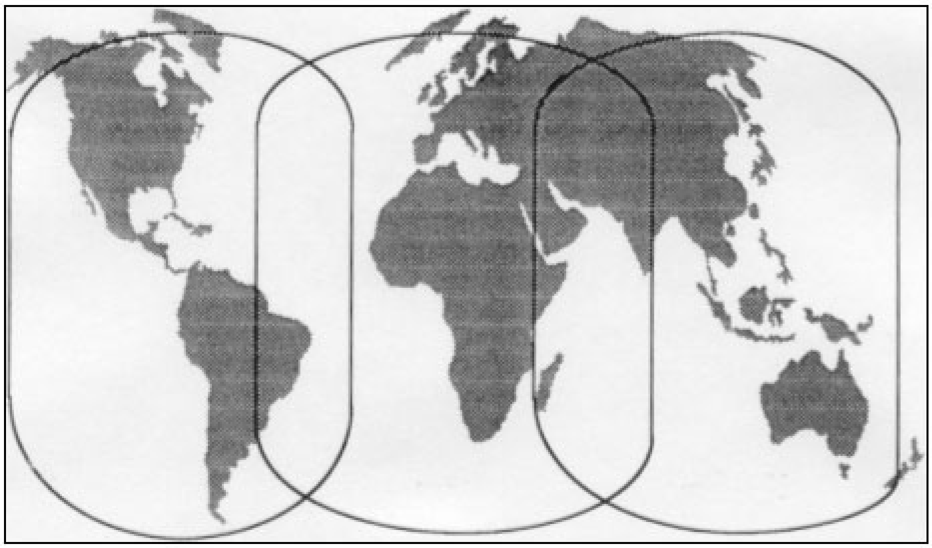
Depending on the audio quality required for each service, each transponder of each beam on one of the satellites is capable of carrying up to 96 x 16 kbps services. Audio coding developed by the Fraunhofer Institute (FHG) for the project is based on the MPEG Layer 3 algorithm with customisation to suit the WorldSpace project. The coding rate for each service is available in simple multiples of a basic 16 kbps channel, up to a maximum of 128 kbps. Subjectively, the system offers audio quality standards :

Better than AM	16 kbps
Mono FM	32 kbps
Better than ‘near stereo’ CD	64 kbps
Stereo CD	128 kbps

Data services are provided at up to 128 kbps per channel, using either a shared or dedicated channel.

Broadcasters using the service are able to uplink their programmes from either centralised hubs or via individual feeder links located in any of the uplink footprints of the 3 satellites. Whilst this ease of access for broadcasters is a plus feature, the low power single channel uplink with its large footprint could be vulnerable to jamming. There have not been any occurrences of jamming to either satellite in more than four years of continuous operation, and procedures are in place to handle such attempts.

Figure 5.2. WorldSpace Up-link Coverage



AmeriSpace AfriSpace AsiaSpace

Broadcasters using the WorldSpace system have the choice of using a low power uplink local to the studio (PFLS), or routing their service(s) to a remote, high power uplink (TFLS) site. This arrangement is possible by the use of Frequency Division Multiplex Access (FDMA) for the uplink.

When received at the satellite, the signals from a PFLS are ”assembled” by the on-board processors to form a broadcast multiplex. The arrangement will allow each of the three spot beams to downlink its own multiplex on the processed transponder. In short, the on-board processing simplifies the uplinking procedures.

The downlink for each beam uses Time Division Multiplex Access (TDMA) and the baseband processing on-board the satellite carries out the FDM to TDM conversion.

For its transmission system, WorldSpace uses a system it has developed itself (early in 1998, WorldSpace made details of the system available to the ITU-R and the system is now designated ITU-R System Ds). The WorldSpace decision to use time division multiplexing (TDM) provides a greater link margin (the extent to which the clear sky carrier to noise ratio exceeds the threshold for reception level) than would be available with a COFDM system such as Eureka 147 (T-DAB). A greater link margin can be used to serve a larger coverage area, but cannot overcome the problem of blockage, which is a fundamental problem for all satellite systems.

Subsequently WorldSpace developed a hybrid satellite and terrestrial repeater system designated as ITU-R System Dh to provide reliable reception in vehicles. This system has other enhancements including time diversity. This system has also been utilised by XM Satellite Radio to provide DARS services in the US (refer to Section 5.2).

5.1.1 Receiver Systems

Receivers for the WorldSpace system are described on the Worldspace website at: www.worldspace.com/receivers/bundle.html. Agreements reached in June 1996 with SGS-Thomson and ITT Intermetall to produce a very large number of silicon chipsets marked an important milestone in the development of receivers for this project. By implication, the RF specification for the system had at that stage been completed.

An announcement about the manufacturers of the WorldSpace Starman receivers was made in June 1997. The named manufacturers were:

- Hitachi
- JVC
- Panasonic
- Sanyo

More recently WorldSpace has licensed a number of manufacturers in India, China, Korea, Indonesia and Thailand to manufacture low-cost receivers for domestic and export markets. New generation receivers for the enhanced hybrid satellite/terrestrial service are currently being developed.

5.2 SIRIUS Satellite Radio / XM Satellite Radio

These two US-based organisations have implemented and are operating satellite radio systems providing a variety of mobile/fixed services throughout the 48 contiguous states, in Canada and offshore. Both intend to offer services within Mexico as soon as regulatory approvals can be obtained.

The services are currently audio channels of music or voice. They typically offer 100 audio channels, 60 of which are various genres of music and 40 of which are voice (talk, news, sports, etc.). Demonstrations have been made of possible future offerings of data and video.

The services are offered to subscribers at rates which vary as a function of subscription length; the highest being a monthly rate of approximately $13 US, and $499 US for the life of the radio being the lowest assuming a five year lifetime. However, discounts and promotions (some of which include the purchase of the radio) provide great variability.

The services are provided to mobile vehicles (private automobiles, trucks, boats and airplanes) and to homes and businesses. The number of subscribers at the end of 2006 is over 14 million with the preponderance being in motor vehicles. There are two types of receivers for this market. The first type is called aftermarket where subscribers wish to add a satellite radio capability to their existing car. This is accomplished by purchasing an auxiliary receiver with a satellite antenna at a local retailer, many of which also install and activate the equipment. Connection to the car's audio system is either direct, through the FM radio or through the cassette player depending on the existing radio's design and user preference. The second type is called OEM where subscribers buy a new car with the satellite receiver installed, either at the factory or dealer, and the car is delivered with the satellite radio capability activated. Costs of such receivers vary, the current range being approximately $150-$300 US without promotions.

Various models of the aftermarket receiver exist such as plug-and-play, home, transportable (e.g., boombox), boat, etc. One of the more popular is the plug-and-play receiver, which is sold with a dock for home installation and a dock for automobile installation. The subscriber can simply move the receiver from one location to another, thus avoiding the need to purchase a second one. The future trend is believed to be towards OEM radios as well as to reduce costs of receivers, primarily due to improved ASIC chipsets (which are the heart of the receiver) and consequent increased manufacturing volume.

The Sirius and XM services are similar except for the music channels where Sirius has no commercial advertising. The systems are different however. Both systems use the 12.5 MHz bandwidth assigned (Sirius radio frequency allocation is 2320.0-2332.5 MHz and XM is 2332.5-2345.0 MHz) by employing approximately the top and bottom 4 MHz for satellite transmission with TDM/QPSK modulation and the center 4 MHz for terrestrial repeaters. These terrestrial repeaters take the satellite signal and rebroadcast it in the urban cores of large cities with COFDM/QPSK modulation to overcome service outages from blockage. Sirius transmits all its channels in one contiguous block approximately 4 MHz wide while XM divides its channels in half, transmitting them in two blocks each approximately 2 MHz wide.

The Sirius and XM orbital designs are also different. Sirius employs a constellation of 3 satellites in an inclined, elliptical geosynchronous orbit while XM employs 2 satellites in geostationary orbit.

Both systems use satellite space, time (4 seconds) and frequency diversity to achieve very high availability of service (e.g., above 99%). Sirius chose its orbit to maximize subscriber elevation angle to the satellites in the northern third of the United States which reduces the need for terrestrial repeaters and lowers the probability of outages from blockage and foliage attenuation. Currently, Sirius employs approximately 100 repeater sites and XM approximately 800.

5.2.1 Sirius Overview

The first Sirius spacecraft was launched on July 1, 2000. Exactly five months later, on December 1, the third spacecraft was launched, completing the three satellite S DARS (Satellite Digital Audio Radio Service) constellation. The three spacecraft are deployed in inclined, elliptical, geosynchronous orbits, which allow seamless broadcast coverage to mobile users in the contiguous United States. Terrestrial broadcast repeaters provide service in urban cores. The system is in operation, providing the first ever S-DARS service.

The constellation design results in satellite ground tracks over North America with two satellites always above the equator. High elevation look angles from the mobile ground terminals to the satellites minimize performance degradation due to blockage, foliage attenuation and multipath.

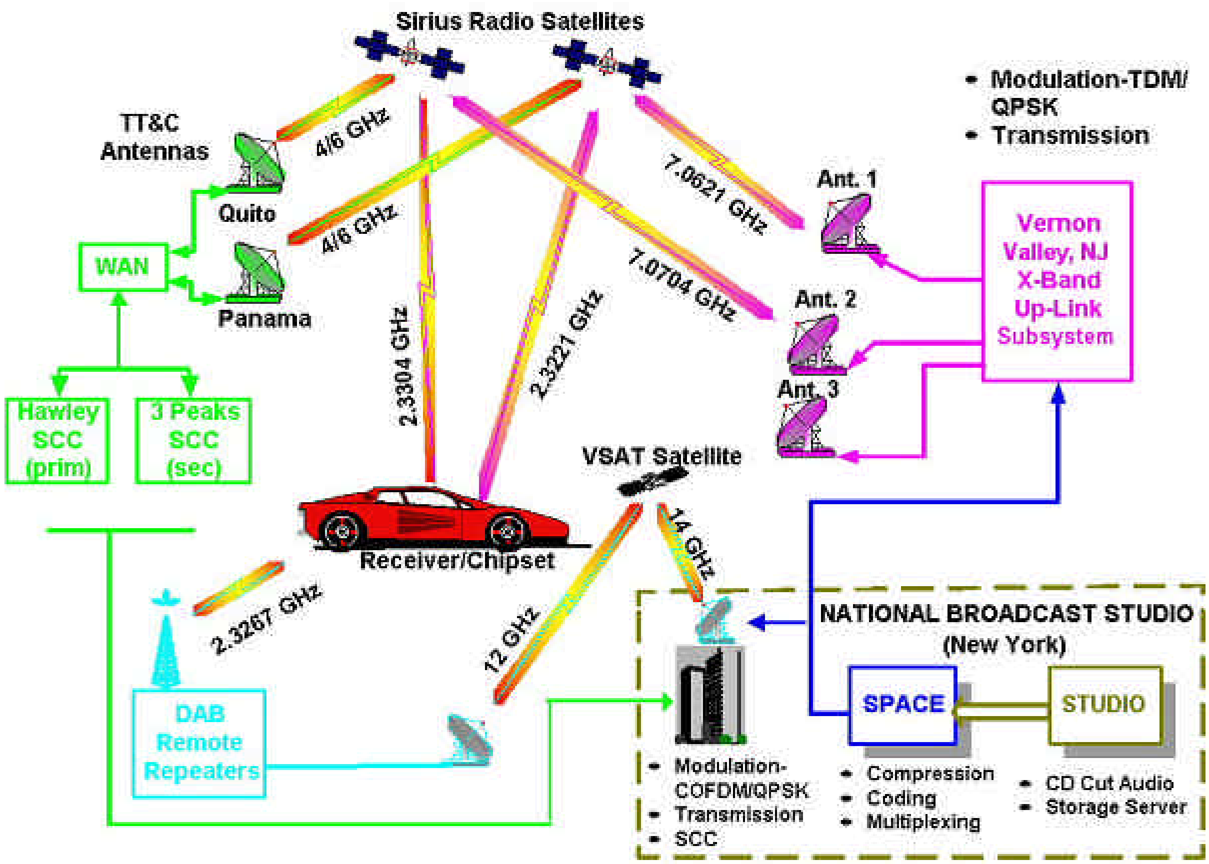
The spacecraft were built by Space Systems/Loral using the 1300 bus modified for operation in high inclination orbits. Each spacecraft was launched using a dedicated Russian Proton booster. The satellite payload is a bent pipe repeater using 7.1 GHz for the uplink and 2.3 GHz for the broadcast transmission. The repeater high power amplification stage consists of 32 Traveling Wave Tube Amplifiers phase combined to yield a total RF output power of nearly 4 kW at saturated operation. The satellite antennas are mechanically steered to maintain the transmit beam centered on CONUS (Contiguous United States) and the receive beam centered on the uplink earth station located in Vernon Valley, New Jersey.

The satellite payload design and performance are described. The principal spacecraft bus systems are described with emphasis on improvements made for operation in the inclined, elliptical geosynchronous orbits.

The two active satellites transmit the same signal at different frequencies with a 4-second delay between them, which is inserted at the uplink earth stations. In the urban core of large cities where satellite blockage can be very high, terrestrial transmitters rebroadcast the satellite signal. The satellites' different orbital positions, transmission frequencies and signal delay provide the diversity while the receivers' equalizer and maximal ratio combiner (e.g., sums the two satellite and terrestrial repeater signals) provide the other listed techniques. Moreover, the achievement of high elevation angle is an extremely important attribute, and its achievement required the adoption of a unique orbital configuration.

Originally Sirius Satellite Radio had planned for geostationary satellites at 80° and 110° West longitude. The resulting low elevation angles to mobile users in the northern third of CONUS would cause service outages whose number and duration result in an unsatisfactory quality of service irrespective of the diversity employed and the satellite Effective Isotropic Radiated Power (EIRP) level. Satisfactory service might be achieved by deploying an enormous number of terrestrial repeaters but this was judged impractical given the economic and regulatory issues involved. Consequently, an orbital constellation was designed by Sirius Satellite Radio and implemented by Space Systems/Loral (SS/L) that provides high elevation angles over this critical area.

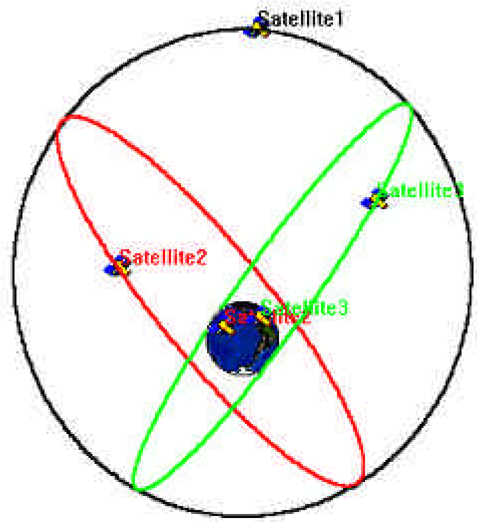
Figure 5.3. Sirius SDARS Delivery System



Sirius SDARS Delivery System

The Sirius constellation consists of three satellites in inclined, elliptical geosynchronous orbits whose planes are 120° apart, as shown following. The satellite orbital elements are given in the accompanying table, and the satellites' ground tracks are also shown. The orbital configuration was designed so that each satellite spends 16 hours north of the equator, during which time it is used for transmission, and 8 hours south of the equator when it is inactive.

Figure 5.4. Sirius Constellation



Each satellite follows the other around the ground track in following picture with 8-hour separation. The perigee in the southern hemisphere is 24,500 km, which is above the Van Allen belt, and the apogee in the northern hemisphere is 47,100 km.

Semi-major axis	42,164 km
Eccentricity	0.2684°
Inclination	63.4°
Argument of Perigee	270°
RAAN* • FM-1 • FM-2 • FM-3	 285° 165° 45°
Apogee Altitude	47,102 km
Perigee Altitude	24,469 km

**Right Ascension of Ascending Node*

Figure 5.5. Sirius Ground Track

The decision to use inclined, elliptical orbits rather than geostationary orbits for Sirius was made approximately one year into the design and manufacturing phase of the project. At the same time, modifications were made to the payload requirements, affecting spacecraft configuration and support subsystems. While the majority of the satellite hardware remained unchanged, a number of modifications were required. Several design trades and decisions were influenced by the existing state of design and development. The program was carefully replanned to accommodate late arrival of new or modified hardware and software, while maintaining the integrity of the overall system testing. The highest priority was given to quality and reliability of the end product. No "shortcuts" were taken during the development or qualification of new hardware or software. The overall success of the program demonstrated the ability to respond quickly to a change in the implementation plan. The following table summarizes the modifications made to the 1300 design for the Sirius application.

Changes Made For Sirius Inclined Elliptical Orbits

Parameter	Geostationary	Inclined Elliptical
Dry Mass	1300 kg	1575 kg
RF power (operating)	2.5 kW	4 kW
DC power – EOL Solar Array Battery	 7.5 kW 7.5 kW	 8.5 kW 8.8 kW
Control System	3-wheel mom bias	4-wheel mom bias On-board orbit propagator
Control Modes	Orbit Normal	Orbit Normal Yaw Steering
TX Antenna	Fixed Gregorian Gain 27.8 dBi; Cross-pol 24 dB	Gregorian; two axis steering 360º rotating shaped subreflector Gain 27.2 dBi; Cross-pol 28 dB
RX Antenna	Fixed offset fed	Offset fed; two axis steering
Solar array	2x4 panel HES	2x5 panel HES
Battery	2x32 cell - 149 AH	2x34 cell - 178 AH
TT&C	X, C and S bands CONUS ground station Limited motion antennas	C and S bands 2 near equatorial ground stations Full motion antennas
Launch Vehicle	Ariane	Proton

The launch of the Sirius Satellite Radio constellation marks the first use of satellites for Digital Audio Radio Service broadcasting in the United States. The three high power direct broadcast satellites will provide service for millions of subscribers. The Sirius Radio system is the world's first satellite broadcast system using non-geostationary orbits.

The use of inclined elliptical orbits coupled with multiple modes of transmission diversity provides notable advantages for broadcast service to the mobile market. Pioneering technology was developed and implemented by Sirius Satellite Radio and Space Systems/Loral in order to accomplish this unique achievement.

5.2.2 Deployment Status

Current population and transmission status of Sirius and XM satellite radio services is shown in the following chart.

	Continental US Coverage	Satellites	Ground Repeaters
Sirius Satellite Radio	100%	3 in HEO	~100
XM Satellite Radio	100%	2 in GSO	~800

5.3 Mobile Broadcasting Corp. and TU Media Corp. – ITU-R System E

Mobile Broadcasting Corporation is a commercial organisation based in Tokyo, Japan and TU Media Corporation is a commercial organization based in Seoul, Korea. Although they have one geostationary satellite in common, each of them can use its own transponders and is independently providing high quality digital audio, medium quality digital video and multimedia data services to vehicular, portable and fixed receivers using the satellite and a number of terrestrial repeaters. The same frequency band, 2630 - 2655 MHz, is used by sharing polarization. The service area of Mobile Broadcasting Corp. is Japan and the service area of TU Media Corp. is Korea. Broadcasting signal can be received by receivers with small antennas. To generate enough EIRP for mobile reception, the satellite is equipped with a large transmitting antenna and high power amplifiers. After the launch of the satellite in March 2004, the commercial service in Japan was started in October 2004, currently including 30 audio channels, 8 video channels and about 60 items of multimedia data services. The commercial service in Korea will be started in May 2005, including 22 audio channels and 12 video channels.

The major issue related to signal propagation in BSS (sound) is signal loss due to blockages on the signal path from the satellite to the receiver. Two techniques are used to overcome this issue. One of them is bit-wise interleaving, which is used to overcome the instantaneous signal loss caused by blockages, such as bridges over highways, in vehicular reception environment. Invalid data generated due to the signal loss are distributed over several seconds through the deinterleaver and corrected through the decoder of forward error correction code in the receiver. The period of the signal loss which can be recovered by this technique is approximately a second. The other method is introducing terrestrial repeaters. The terrestrial repeaters retransmit the satellite signal and are expected to cover the area where signal loss occurs due to blockages, for example, buildings and large constructions. In the circumstances, where terrestrial repeaters exist, multipath fading appears at the receiver because more than two broadcasting signals are received at the same time. The CDM (Code Division Multiplexing) and RAKE combining technique is adopted, so that the same frequency band is used for the satellite and the terrestrial repeaters.

The system was approved by ITU in July 2000 as ”Digital System E’ in Recommendation ITU-R BO.1130, System description and selection for digital satellite broadcasting to portable, vehicular and fixed receivers in the bands allocated to BSS (sound) in the frequency range 1400 – 2700 MHz.”

5.3.1 Receiver Systems

The services are provided to persons, to mobile vehicles (automobiles, trucks, boats and airplanes) and to homes. There are several types of receivers for this market at the moment; palmtop receiver, PC card receiver, plug-and-play receiver and mobile phone type receiver. Palmtop receiver is a dedicated receiver with 3.5-inch LCD, which is small and light enough to be carried to any place. PC card receiver is used with a notebook computer and you can enjoy video services on the display and audio services through the speakers while you are using the computer. Plug-and-play receiver is used with cradles, which are installed in car and at home, so that you can use the receiver not only in car but also at home without buying extra receiver. Mobile phone type receiver is embedded in a mobile phone and you can receive video or audio services at anyplace without bringing additional equipment, though the display is rather smaller than that of dedicated palmtop receiver.

6 Internet Radio (IR)

6.1 Introduction

Traditionally, audio programmes have been available via dedicated terrestrial networks broadcasting to radio receivers. Typically, they have operated on AM and FM platforms, with the more recent addition of digital radio-frequency spectrum, including DAB, DRM and IBOC. This paradigm is about to change.

Radio programmes are increasingly available not only from the terrestrial networks, but also from a large variety of satellite, cable and, indeed, telecommunications networks (e.g., fixed telephone lines, wireless broadband connections and mobile phones). Very often, radio is added to television broadcasts. Radio receivers are no longer only dedicated hi-fi tuners or portable radios with whip aerials, but are now assuming the shape of multiple multimedia-enabled computer devices (desktop, portable, PDA, Internet radios).

This sea of changes in radio technologies impact dramatically on the radio medium itself - the way it is produced, delivered, consumed and paid-for. Radio has become more than just audio - it can now contain associated metadata, synchronized slideshows and even short video clips. Radio is no longer just a "linear" flow emanating from an emission mast - audio files are now available on-demand or stored locally for time-shifted playout. It is the convenience of the user, rather than the broadcaster-imposed schedule, which matters now.

Internet Radio (IR) is a relatively recent phenomenon. Nevertheless, during the past ten years Internet has become a very important distribution mechanism for audio and video streams and files. Audience statistics show that IR is increasingly popular, especially among young people and users in offices.

This paper introduces the concept of IR and provides some technical background. It gives some examples of actual IR services now in place in different countries. Finally, it provides some guidance on how traditional radio broadcasters need to adapt and adjust in order to be capable of meeting the requirements of the rapidly changing multimedia environment.

6.2 Bringing Radio to the Internet

Internet penetration worldwide is very close to the one billion users mark. Almost 70% of the American population have access to the Internet from home, and one-third access the Internet at work. Canada, South Korea, Japan and Germany follow closely at 60-70%. The use of the Internet is growing at a tremendous rate. Recently published statistics suggest that, on average, 31 connections are made per month, and more than 26 hours are spent browsing the Internet each month to visit 66 sites and view 1268 pages. Eighty-seven per cent of uses send e-mail massages, 60 per cent use instant messaging services and 55 per cent download files. Twenty-two per cent of users worldwide have already tried video on the Internet.

The American Media Research company, Arbitron/Edison (www.arbitron.com), released, in 2005, results of a major study on Internet and Multimedia in the US. This study suggests that an estimated 55 million consumers use Internet radio and Internet video services each month.

The study identified the following reasons why people listen to Internet radio, as opposed to off-air radio:

To listen to audio not available elsewhere	17%
To control/choose the music played	15%
Fewer commercials	14%
Greater variety of music	13%
Clearer signal than over-the-air radio	8%
Less DJ chatter	8%
Because it is "new"	7%

Internet listening appears to be concentrated among well-known Internet radio brands such as America Online's AOL® Radio Network; Yahoo!® Music, Microsoft's MSN Radio, WindowsMedia.com and Live365. Every week, these stations reach an average of 4.8 million listeners aged 12 and older during the hours of 06.00 – 00.00. Listeners to these five major Internet radio brands account for roughly one out of four of the 20 million weekly Internet radio listeners in the US.

6.3 Internet Radio peculiarities

Radio over Internet differs from other delivery media in three ways:

- It is a relatively new way to experience radio via a computer device. The consumer uses a new interface (screen, keyboard, mouse) and is able to search and select different content according to the station name, country of origin, genre or style, as well as viewing the currently played programme ("Now Playing"). The station's frequency (as in FM or AM) or multiplex (as often in DAB) is irrelevant. The users can shortlist their preferences by compiling personalized favourites lists. In addition, it is possible to generate a virtual station schedule according to one's preferences. An "on-demand radio" is also offered by many traditional broadcasters on their websites; this allows the user to click and play the archived programme items which were broadcast via conventional terrestrial channels during the previous seven days or so.

- IR widens the choice of service providers. These can be traditional radio broadcasters, new (Internet-only) stations, portals or independent users.

- Radio content on the web can differ from radio broadcasting as it has evolved over the last century. Whereas on terrestrial networks the choice of stations is relatively limited, there are thousands of IR stations. It is often possible to choose from a list of most popular stations or to find a station which is playing a particular song from a "Top 50" list. Since computers can use hard disc memory, it is possible to time shift play out.

One of the fundamental differences between IR and conventional radio is the absence of barriers to public transmission. Consequently, even a small, local station can potentially become a global player, or at least an international station.

6.4 Internet Radio as a complement to established radio services

Since 1995, most traditional broadcasters have set up websites in order to provide complementary information for their listeners and viewers. The websites can provide a variety of textual and pictorial on-line services, as well as on-demand audio or audio/video clips associated with news events and live (continuous) reproduction of existing radio and television programmes.

For conventional broadcasters IR could usefully complement existing on-air broadcasts. IR works best as a narrow-cast medium targeting a small number of concurrent users. Should this number increase to more than a thousand (or several thousands), Internet streaming servers are generally not capable of providing the streams economically. In other words, IR is only really useful if it is kept relatively small. For example, it is probably not very sensible to use Internet for big one-off events such as Live 8 on 2 July 2005,[13] as satellite or terrestrial networks can reach many more people.

IR is best suited to niche content, such as education, specialist music, and programmes aimed at ethnic minorities, which may be of interest to a relatively small number of people. Often it is considered too extravagant to use scarce spectrum for such programmes.

IR can offer a solution for communities scattered across the world. For example, there may not be enough fans of gypsy music in a given part of the world to justify a local broadcast station, but if we add listeners around the world who are interested in this kind of entertainment the potential audience will look a lot healthier.

While it is easy to introduce a new IR stream for niche radios, it is more difficult, if not impossible, to find spectrum for FM station, which is already very congested in some large agglomerations. One example is SR International's Immigrant Languages Service, which is primarily intended for immigrants within Sweden, but also reaches audiences abroad through its webcasts.

The scalability of IR is a major issue. When audiences are relatively small (e.g. several hundreds concurrent listeners), bandwidth – and thus cost – is reasonable. However, when audiences increase, operational cost may escalate. In a way, a station may become a victim of its own success. A peer-to-peer (P2P) approach may help reduce distribution cost. Multicast is also an option, but it requires multicast-enabled routers which may not be readily available everywhere. Also, multicast excludes on-demand delivery.

IR is inherently interactive. IR websites are places for listeners to interact not only with the station, but also with each other. These interactions are usually achieved through text messages, e-mail forums or chat rooms, as well as in a growing number of cases, audio and video messages. Indeed, listeners may become active contributors to the website audio-visual content. For example, programme files could be mailed in from around the world direct from artists or music groups. As an example of interactivity and audience active participation, NRK - and other European broadcasters - have organised country-wide contests of amateur pop groups, allowing users to vote and select the most popular group.

IR websites have a unique possibility to offer both live and on-demand audio programmes. Many radio stations have created on-demand online archives enabling their listeners to hear programme items that were originally broadcast on-air, for example, up

[13] Musicians and artists from around the world joined together to influence the struggle to end global poverty. There were pop music concerts from 9 different places around the globe on the same day with several hundreds million watching on TV and listening to the radio. Among others, WorldSpace UPOP Music Channel 29 transmitted the concerts in real time (live).

to seven days before. One example is the BBC Radio Player. This on-demand service allows users to time-shift broadcasts and frees them from the constraint of adhering to station schedules. On-demand transfers control to the listeners: they can create their own schedule of programmes.

Web radio has the advantage of allowing broadcasters to measure audience directly (see Section 6.9). Broadcasters using a Windows Media Server, or other streaming media, will have detailed reports of the streams played, while those using web servers can estimate audience sizes by viewing the traffic statistics found in the web server log file, an automatically-generated list of all the files served.

IR adds a global audience which may be important for ethnic minorities scattered around the world. While terrestrial radio is generally limited to a certain geographical territory, IR's audience is effectively global and is redefined according to shared interest. IR radio introduces a concept of a multitude of niche audiences spread globally and not necessarily limited to one geographical region or country.

6.5 Internet-only stations: IR Portals and Music Portals

There are a number of web radio sites that offer customizable programming using their own players or ones already loaded onto your PC. Most sites feature dozens of different musical genres from baroque to zydeco and some allow you to tune in to live broadcasts from around the globe.

There are also Internet portals which help the user find a suitable IR station. Portals such as radio-locator.com allow users to search for stations according to genre (or format), name, location (city, state or country), frequency (if the station is already on the air) or even owner. Often several thousand stations are available on such portals. Some radio portals are listed in Section 6.12.

Lists of FM and AM radio stations can be made available over the Internet to mobile devices such as a Palm OS or Windows CE handheld computer using suitable software.

6.6 Streaming technology for radio services

With recent technological improvements such as rapid adoption of high-speed connectivity and ever increasing computer processing and storage power, streaming over the Internet (sometimes called webcasting) has become a mainstream media delivery platform. Universal standards for audio and video delivery have emerged to gain widespread adoption in the marketplace. In addition, user experience of watching video and listening to audio online has improved dramatically. Issues such as incompatible formats and versions or browser incompatibility are now less critical.

There are different standards for encoding and delivering audio files and streams online. Following the pioneering developments of RealNetworks, Windows Media and QuickTime, it now seems that MPEG-4 will dominate. MPEG-4 represents a major step forward in audio/video coding, as it supports new types of media objects, such as 3D and synthetic objects. It supports interactivity at the client and server side. It is highly scalable and covers video resolutions from a thumbnail size suited to mobile applications to HDTV for home cinema, and from monophonic audio at 20 kbps to multichannel audio in the MBps range.

The streaming system architecture comprises four elements: capture and encoding, serving, distribution and delivery and media player.

Capture and encoding takes the source audio from the microphone and exports it into a compressed (encoded) computer file. These files are stored on a content server which controls the real-time delivery of the stream. The distribution channel (usually the Internet) connects the server to the player. The media player renders the media on the PC or another device (hand-held wireless devices, games consoles, interactive TV, etc).

As Internet is overlaid on telecommunications infrastructure, IR is now widely available via a variety of two-way communication networks, both wired and wireless. narrow-band (dial-up) at home and broadband connections in offices, via WLAN hot spots in airports, congress centres and other public places. The number of listening hours is staggering.

Broadband access is obviously a big plus and some of the streams are so good you can enjoy them over your home stereo system.

IR services can be delivered in a variety of configurations ranging from direct server-client to podcasting.

(1) Server-client

Unicasting is a classical approach to radio streaming. Requests from clients (users) to receive a stream are managed by a server or a cluster of servers. In case of clustering, load balancing is used to improve reliability of the stream delivery, especially if one of the servers breaks down. The server cluster feeds a common Internet line used to transmit the streams to the clients. The total bandwidth provided by such a server farm is proportional to the number of clients and the bitrate of streams. This means that doubling the number of clients or bitrate will double the system capacity and cost.

Unicasting also has a "scaling" problem. Since all the streams are transmitted to the Internet from one source, a server quickly reaches its upper capacity limit, resulting in a "server busy" message.

(2) Distribution networks

The Content Delivery Network (CDN) consists of a large number (typically several thousand) of "edge"[14] servers situated around the world. Each server uses the same home page and is uploaded with the same content. The user gets content from the nearest server, so that the access delay is minimal. The CDN approach distributes the load among the geographically separated servers and increases the possible number of concurrent requests and streams that may be handled. The CDNs can potentially cater to several thousand simultaneous streams but are very costly. For example, Akamai's globally distributed edge computing platform comprises more than 15,000 servers in more than 1,100 networks in 70 countries.

[14] The word "edge" is used here to mean "close to the user".

(3) WiMAX

WiMAX is a new IP-based communications technology based on the IEEE 802.16-2004 standard, which will provide broadband wireless access to portable devices like laptops, personal digital assistants (PDAs) and smartphones. WiMAX will complement fixed DSL and WiFi networks by providing mobility and portability. It will offer seamless hand-over between WiMAX, WiFi and mobile 2G/3G networks. It will bring new dimension (mobility) to broadband TV and Radio. For more information, see www.wimaxforum.org.

(4) Multicasting

Multicasting is a solution to serve a single stream to multiple users. The multicast-enabled network routers clone (replicate) the Internet datagrams (packets) for each user requesting the stream. Therefore the same content is conveyed to a group of users. Multicasting cannot use automatic rate changing and is not suitable for on-demand services. If multicasting is to be used for several sites at the same time, then Virtual Private Networks (VPN) should be used to bring the stream from the originator to these sites, and then multicast locally.

(5) P2P networks

Peer-to-peer (P2P) networks refer to computers that communicate directly with other computers without passing through intermediaries. It enables users to pool resources, such as processing power, storage capacity and bandwidth to overcome the problems of congested Internet links and server crashes. Internet radio broadcasters are beginning to use P2P systems to distribute their content in what looks like a win-win situation, with consumers obtaining a more reliable service and broadcasters benefiting from drastically reduced bandwidth fees.

Since P2P networks have the potential to create distribution channels which are more efficient than traditional broadcasting, some analysts have gone as far as to suggest that this will inevitably bring about a massive paradigm shift. In a P2P scenario, there would be no need for the "middleman" - consumers would download content directly from programme producers. This would lead to a massively reduced role for traditional broadcasters who would be relegated to providing only live sport and breaking news.

P2P systems use several distinct techniques to distribute files more efficiently. One of the most widespread is "swarming," which was pioneered by BitTorrent. In this technique, peers share portions of data: files are broken down into small pieces and then distributed randomly between peers who exchange the pieces in order to complete a sort of jigsaw puzzle.

The Danish-based company, Octoshape, which has worked closely with Danish Radio, claims that its GridCasting solution saves 97% of bandwidth compared to the traditional server farm solution. As with other P2P technologies, the more people who download files, the faster they download. Other potential benefits include higher quality bitrates, instant play, no buffering and fewer interruptions.

In Britain, the BBC is working with Kontiki P2P technology to provide a new online service that will allow viewers to download radio and TV programme from the previous seven days free of charge.

(6) Podcasting

Podcasting is a way to 'subscribe' to radio programmes and have them delivered to your personal computer. Podcasting stands for Personal On-Demand (narrow)casting. It combines blogging with audio files that can be played on your PC or MP3 player. It involves a "push" of specially encoded multimedia content to subscribed PCs via RSS 2.0 protocol. Podcasting allows the listener to choose not only to what to listen to, but also when and where. Users can return feedback and comments. It is not limited to radio and music (typically encoded in MP3) but can include video, films, games, etc. Is not limited to broadcasters, virtually anybody who has content can become a podcaster.

Subscribers to Radio podcasts can automatically receive the latest edition of the programme in the form of a file. This file can then be easily transferred to a portable MP3 player. To do this, users need an Internet connection and a piece of podcast software which is usually available free of charge. This software can check the radio station for content updates and automatically download them to the player as soon as they are available. As a general rule, programme files can be made available shortly after broadcast, but in some cases this may be several hours later.

There is a multitude of podcasting software available from www.podcastingnews.com. This software varies from one computer platform to another (Windows, Mackintosh, Linux, etc). The same website also provides software for publishing podcasts.

6.7 Internet Radio terminals and playback devices

Internet radio terminals are user devices which can reproduce streaming content. In the beginning, streams could be played by a software application on the PC. Now we are seeing media players in mobile devices and in home entertainment products such as the set-top box. Today, a PC user may have three or more players installed to provide support for different codecs available in the market. Thankfully, PC makers have made it easy with pre-loaded music players, from Apple's ITunes and QuickTime, to Real Player and Windows Media Player.

Players can be used in three different ways: as a content portal, a stand-alone player, or a plug-in to a web browser. In the latter case, the streaming content may become an integral part of a synchronised rich media experience, combining text, graphics, audio and video (using SMIL[15]).

Audio-only players are still very popular, as there is huge demand from music lovers to download tracks over the Internet. They serve as a jukebox to organise music libraries and set up playlists. They can also rip CDs, store MP3 files on the hard drive and download to portable music players such as iPod. Examples include WinAmp from NullSoft, and iTunes from Apple.

Today, about 95% of all media players installed on the desktops worldwide, are Microsoft's Windows Media. RealPlayer and QuickTime follow closely by 86% and 82%, respectively. Flash players are becoming increasingly popular for multimedia, whereas MP3 are mostly used for downloadable audio files.

[15] Synchronised Multimedia Integration Language

An interesting example of a PC audio player is the BBC Radio Player which is a PC application that allows Internet users to download BBC radio programmes via a programme guide for up to 7 days after broadcast. BBC is now in the process of trialling an Integrated Media Player (iMP) which will allow for both radio and television programme downloads but, due to copyright restrictions, only to the UK territory. For the users' benefit, the programme guide is available a week in advance and a week behind. Users are able to download programmes as soon as they have been broadcast on TV and Radio and can watch them as many times as they like during seven days. iMP also allows users to subscribe to a series which automatically downloads each programme immediately after being broadcast.

Music download is now a feature of 3G mobile phones. Motorola and Apple joined forces to market a device combining iPod with mobile phones. For the moment, downloading songs still has to be carried out via the Internet and the user's PC, although in the near future the mobile phone could do it via 3G networks direct.

It is also worth mentioning the possibility to attach a small FM transmitter to a portable iPod player for listening on car radios. This is important, as radio listening in the car may be affected. Some people may choose to listen to their personal collection of pre-recorded files on iPods, rather than listening to local FM or AM stations. Just as commuters are catching up to the idea of satellite radio for their cars, a new wireless approach called "Roadcasting" will allow you to tune your radio to music playlists coming from other cars on the motorway.

A special category of IR terminal devices are disguised computers which look like old radios but can connect to Internet radio stations. An early example of this approach is Kerbango from 3Com (no longer available on the market). Newer Internet radio receivers include products from Reciva, Acoustic Energy, Noxon, Slimdevices, SoundBridge, Solutions and others. For example, Acoustic Energy uses a wireless broadband connection and supports Real Audio, Windows Media and MP3. Radio stations' URLs are store on a central database which can be easily updated on request to accommodate any other radio stations. Currently, more than 10,000 stations from virtually any country worldwide and of more than sixty different genres are available. Typically, the prices of Internet radios range between $100 and $200 US.

Another consumer electronics device which allows consumers to listen to Internet radio and Internet music is Streamium from Philips. The concept here is different because you need a separate PC and a broadband Internet connection. The PC and Streamium can be located in two different rooms (which is convenient because of the fan noise of the PC) and are connected wirelessly using 802.11g connection (bandwidth 54 MBps). An LCD display shows audio metadata (song titles, artist names, remaining and elapsed play time, etc.), so you do not need to have your TV turned on when listening to your music or radio. There are many other appliances in the market that, when connected to a PC, play radio or music on the home stereo or surround equipment in the living room (e.g., AudioTron from Vermont, PhoneRadio from Penguin, etc.).

6.8 Internet Radio's relation with the traditional radio

The comparatively low entry barriers for broadcasters have led to a proliferation of Internet radio sites. This has increased the importance of promotion and product differentiation. However, broadcasters enjoy a significant competitive edge. They benefit from both strong brand recognition and the ability to cross-promote across Internet, radio and TV networks.

In order to promote their Internet services, broadcasters must communicate the all important web addresses to listeners. It is not the aim of this paper to explore marketing techniques, but suffice to say that broadcasters can achieve this in a variety of ways: during live programmes; in advertising campaigns on radio, TV, Internet or in print; and with e-mail marketing campaigns, press releases and giveaways.

Where Internet radio really comes into its own is as a marketing tool in its own right. Radio is an "experience product" which consumers must sample before they become regular listeners. There is evidence from the BBC and others that Internet radio players can boost listening figures for traditional radio by encouraging listeners to experiment and discover new programmes. Furthermore, some shows already have as many "catch-up" listeners online as they do for the original live broadcasts.

The BBC Radio Player provides consumers with lists of the most popular radio programmes and links to allow listeners to click through to shows related to their favourite genres. The BBC hopes that later versions of its player will offer hints for listening, along the lines of the "if you liked that, you may like this" services offered by Amazon and Q Magazine. As things stand, the BBC claims that its player adds millions to listening figures.

Internet radio is also a useful platform for collecting data and for building communities of dedicated listeners. Message boards and chat rooms create communities, with the added benefit that in order to register, listeners must fill out customer profile forms and give their contact details. Information gathered in online competitions can also contribute to listener databases for the purposes of market research.

6.9 Measuring audience

One of the outstanding features of Internet radio is that audiences can be measured with precision and accuracy, as every hit of the keyboard key or mouse is logged. In conventional broadcasting, research results may depend on user behaviour, the methodology used and the audience sample taken, so these results are often open to argument and criticism.

Measuring web audience and understanding web user behaviour is vital to online businesses. Consumer statistics data is used to keep a record of a website's hits and traffic patterns and can help in understanding visitor behaviour. This data may provide the overall number of visits to the website during the specified time frame in terms of parameters such as Page Views, Unique Visitors, Most Popular Pages, Most Visited Documents, Most Visited Dynamic Pages and Forms, Top Downloaded Files, Most Accessed File Types, and others.

As modern websites tend to be dynamically created and designed, and can embed audio and/or video files and streams, Media Monitoring statistical evaluations are needed. Early attempts involved Arbitron[16] Internet radio listening and the way the popularity of Internet radio stations was assessed. Arbitron's MeasureCast Rating gives total time spent listening (TTSL) estimates and provides regular weekly and monthly webcast audience reports. TTSL is the sum total of hours that listeners tune into a given station or portal (network).

For example, during the week of October 28 of 2002, Clear Channel Worldwide was the top ranked Web radio network with 1,566,183 Total Time Spent Listening (TTSL). MusicMatch was ranked number 2 with 1,205,175 and StreamAudio was third with

[16] http://www.arbitron.com/home/content.stm

1,006,579 hours of listening. In addition to duration of listening, Arbitron also publishes demographic highlights such as the peak listening day, peak listening time, geography, age and gender categories, etc.

While such statistical evidence is very useful, it does little to help media service providers and webcasters who need much more detailed insight into user behaviour. To this end, media statistics products or services should be used.

Compared to static web pages, streaming media requires much more bandwidth (more data is transferred in the same time unit) and is more sensitive to Internet infrastructure problems, such as latency, packet loss and jitter resulting in poor audio.

Because of the large performance variations that occur on the Internet, it is important for content providers to measure the performance of their media to gain an objective view on what their users are experiencing. The media monitoring statistics may help content providers to learn how their sites are performing, how they compare to the competition and where they can actually make improvements. Measurements can reveal geographic differences that may be related to the ISP services, backbone problems that can be quickly identified and repaired, insufficient caching or server power that should be beefed up, etc.

Media Monitoring statistics may be standalone or can be integrated with other visitor data. It provides answers to questions like how many visitors start the audio or video stream? How long do they watch or listen? How often do they click on play, pause or stop? What is the quality of the reception? It allows content providers to find out, for instance, if the online sales of a particular CD increase after visitors have listened to it online, or whether visitors return to the website more often after they have seen a video.

Modern Media Monitoring statistics also provides a possibility to use bookmarks by visitors and measures how often web pages are being added to the favourites of visitors. In addition, the measurement of visitor loyalty has been improved. For every visitor, it is now determined, often by using cookies, whether they are visiting the site for the first time,or if they have been there before.

In providing Streaming Media there are several parameters that are analogous to those monitored for the websites. If we replace Webpages with Streams and Visitors with Requests, we may consider the following parameters for media monitoring:

- number of requests for each stream (per day, week, month, etc.)
- origin - where do requests for streams come from (e.g., which IP number, organisation, country, etc.)
- most demanded streams or most demanded parts of streams
- peak number of successfully provided streams

Some additional specific media-related parameters are those related to media players, quality delivered and user behaviour, as follows:

- Which Media Player (Audio/Video/Graphics)?
- Which speed (bandwidth) for a combination of audio and video programme?
- Start-up time
- Audio quality for a given bandwidth
- Video quality, including video frame rate for a given bandwidth
- Connect time
- Redirect time
- Initial buffer time
- Recovered, lost and dropped packets
- Number of successful buffering attempts
- Duration of buffering (average)
- Total playing time for each user / average playing time for each stream
- Hits and Duration Chart for each stream / all streams
- Number of finished Streams (who and how many have seen it to the end)
- Number of linear Hits (without Stop or Pause)
- Number of loops made

6.10 Case studies

6.10.1 VRT

The Belgian public service broadcaster, VRT Radio, started broadcasting on the Internet in 1997. VRT's radio player offers a mix of six traditional radio stations and three exclusive digital services. Of the more than 300,000 unique visitors it attracts every month, more than 80 per cent listen to live streams, while 10 to 15 per cent listen to both live and on-demand programmes. VRT has seen its bandwidth consumption double over the past year and currently uses up to 45 terabytes of bandwidth a month.

VRT automatically records and uploads all of its on-demand content. News bulletins are available on-demand roughly three minutes after the live broadcast is over. Programmes more than 60 minutes long are available about 20 minutes after the broadcast. VRT's radio player works on both Windows and Mac platforms and is a browser-embedded application that requires Flash 7 and Windows Media Player or QuickTime Player. VRT streams in two formats: MP3, at 6, 32 and 96 kbps; and WMA, at 20 and 64 kbps.

VRT has a global rights agreement with organizations including IFPI (the International Federation of the Phonographic Industry) covering both live and on-demand streaming. This is quite common in Europe and contrasts with the situation in North America, where broadcasters usually pay a fee per listener.

6.10.2 Virgin Radio

Virgin Radio boasts one of the world's most successful Internet Radio networks. According to Virgin Radio, which uses the Limelight LUX tool to monitor its online traffic, it reaches 1.1 million consumers who listen for an average of 4.4 million hours a month. In 2005, Virgin won two prestigious online awards, scooping both the Webby Award and the People's Voice Webby Award for radio. (The Webby Awards is the leading international prize honouring excellence in Web design, creativity, usability and functionality.) In 2006, Virgin became the first UK station to make a daily podcast.

Virgin Radio has been available on the net for nearly a decade. In 1996, Virgin was the first station in Europe to broadcast 24 hours online, initially using Real Player. Nowadays, Virgin has four radio stations, which are available online in a variety of different formats and speeds. Virgin stations are currently available in the following formats:

- Windows Media 20 kbps mono
- Windows Media 64 kbps stereo
- Real SureStream 8 kbps - 32 kbps mono
- Streaming MP3 32 kbps mono
- Streaming MP3 128 kbps stereo
- Ogg Vorbis ~20 kbps mono
- Ogg Vorbis ~160 kbps stereo

In addition, Virgin Radio is available in Real AAC 128 kbps stereo, and QuickTime 64 kbps stereo.

Virgin concentrates on UK listeners - who are the majority of those that listen online - and is fully licensed for broadcasting to the UK over the Internet. This is covered by Virgin's music licensing fees, which cost over £1.2 million a year.

6.10.3 Swedish Radio multichannel audio distribution

In addition to some 15 Internet Radio channels which are regularly broadcast from www.sr.se, Swedish Radio has since midsummer 2001 been distributing multichannel audio files via their web-site on-demand. The audio content is coded in 5.1 DTS (Digital Theater System) format. The SR website posts nearly 40 audio-only clips of downloadable multichannel material, ranging from about one minute duration to shows of over one hour. There has been a huge interest for downloading these audio programmes worldwide: to date more than 4 million successful downloads have been made. Users can play these files directly from the hard drive or from a CD and reproduce them via a surround sound loudspeaker system. The cost incurred for broadcasters is very minor.

6.11 Summary and Conclusions

Conventional radio broadcasting on AM and FM has been around for about a century. New digital broadcasting technologies such as DAB, XM radio, DRM and others are becoming very popular in many parts of the world. Traditional on-air radio has many strengths and is still a vibrant medium. It is likely that it will remain the principal delivery mechanism of radio content for quite some time.

Internet opened a new possibility for radio enthusiasts. During the last ten years or so Internet Radio has been a major focus of technical innovations and operational experiments. Now Internet Radio has become a mature medium with its distinctive characteristics. There are many tens of thousands of Internet Radio stations worldwide, ranging from big portals down to small local and individual streaming stations.

The main assets of Internet Radio are its global reach, interactivity and personalisation. While today the users need a computer device and a broadband connection to access Internet radio stations, in future they will be able to enjoy it on a number of portable wireless devices. Internet radio will become ubiquitous.

Internet Radio has proved to be most successful if associated to conventional radio broadcasting over terrestrial or satellite networks. Nevertheless, many standalone Internet Radio stations have reached a break even point to become commercially successful.

Internet Radio redefines radio content. Not only does it introduce new music and speech formats, but also can embellish them with text, graphics and video. It allows users to listen to a wide selection of audio items when and where it is convenient. These on-demand radio services may dramatically affect the pattern of listening and listening habits.

Internet Radio has highlighted many legal and regulatory issues that need to be addressed. These issues relate to copyright, licensing, content regulation, merchandising, advertising and security. However, these topics exceed the scope of this paper.

6.12 Some Important Radio Portals

Beethoven

www.beethoven.com

For classical music lovers. Features include live requests, free e-mail accounts, chat rooms, contests, classical music news and special offers for enthusiasts. Users can tune in to either the free low-bandwidth stream at 28 kbps using Windows Media Player or the $5.95 per month 96 kbps stream with Real One Player. It also provides links to online libraries of classical music and various opera, ballet, and art sites. The navigational bar is not uniform throughout the site so it is difficult to get to certain areas.

Launch: Music on Yahoo

launch.yahoo.com

As well as listening to Internet Radio, users can watch music videos, shop for ringtones, search for song lyrics, play games and customize a station to play favourite artists. Alongside the US version, there are editions for France, Germany, Italy, Spain, the UK and the Republic of Ireland. A "Turn Off Explicit Lyrics" option allows parents to control what their children are playing. For $36 a year, users can upgrade to the commercial version with twice as many stations. The sound quality on the free player is excellent, although users will get commercials.

Live 365

www.live365.com

Live 365 broadcasts from over 100 countries, in 22 genres, and boasts more than 600 million unique listeners since its launch in July 1999. Users can add artists to a favourites list, rate songs and stations and see which tracks have recently played, although some play lists do not load onto the player. Tracks do not contain explicit lyrics. The VIP All

Access Pass for $3.65 a month gives better audio sound, although it is difficult to sift through the stations for VIP members-only.

Radio VH1

www.vh1.com/radio

Radio VH1 has more than 70 stations plus music news, including scrolling ticker. Within each station is a description of the music, the line up of musical acts and the DJs. Currently, VH1 is not available for Mac users.

IM Tuning

www.sonicbox.com

Users need to download free IM Radio Tuning Software - with the minimum requirements of a 56K modem - to access hundreds of live stations, from Electronica to Kids and Variety. By clicking on a "Tell Me More" button, listeners can receive e-mails with artist and song information. There are Smile and Frown buttons for voting. Enhanced sound quality is available via the iRhythm Remote Tuner, which uses wireless technology to play Internet music over home stereos.

Last FM

www.last.fm

This London-based station offers a number of features, including show business gossip and a forum for launching new artists. By typing in three favourite singers, users can obtain a list of stations featuring these performers. As users add tracks they build a profile which can be compared with others who have similar tastes. If users skip a song or give it a bad rating, they will never hear it again.

MTV Radio

www.mtv.com/mtvradio

MTV aims to appeal to a wide variety of musical tastes. Users can choose from four radio stations: On Air, MTV.com, Celebrity and International. Although the player has VCR-like controls and artist ticker features, users must return to the site to see the full list of stations they want to change.

Radio-Locator

www.radio-locator.com

Radio-Locator provides a broad list for finding a US radio station, Internet streaming radio and world radio. It claims that it is the only web-site which provides a comprehensive list of radio stations worldwide. It has links to over 10,000 stations and over 2,500 online streams in 148 countries. There are drop down menus to search for stations. Users do not need to register to listen to music. The only thing missing is links to Internet-only stations.

SHOUTcast

http://www.shoutcast.com

SHOUTcast is Nullsoft's Free Winamp-based distributed streaming audio system. It is a free-of-charge audio homesteading solution that allows anyone on the Internet to broadcast audio from their PC to listeners across the Internet, or any other IP-based network (Office LANs, college campuses, etc.). SHOUTcast's underlying technology for audio delivery is MPEG Layer 3, also known as MP3 technology. The SHOUTcast system can deliver audio in a live situation, or audio on-demand for archived broadcasts.

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APPENDIX A The Eureka 147 System - System Description

Overview

The Eureka DAB System has been designed to ensure rugged and reliable reception by listeners using fixed, portable or mobile receivers with non-directional antennas. The system is spectrum and power efficient (equivalent or better than FM radio) and can be operated at any frequency up to 3 GHz for mobile reception and at higher frequencies for fixed reception. It is suitable for use on terrestrial, satellite, hybrid (satellite with complementary terrestrial) and cable networks. It currently uses the following audio compression techniques, MPEG 1 Audio Layer 2 and MPEG 2 Audio Layer 2 and supports a range of audio coding rates. It has a flexible digital multiplex, which can support a range of source and channel coding options. This includes programme associated data (PAD) services and independent data services (IDS).

Eureka 147 is currently the only digital audio system that has met all the requirements of the ITU for a new digital sound broadcasting system. It is designated ‘Digital System A’ and has the status of a world-wide standard (ITU-R Recommendations BS 1114 and BO 1130 for terrestrial and satellite sound broadcasting respectively). It is an open standard, fully specified within the European Telecommunications Standards Institute (ETSI), in ETS 300 401.

The system provides strong error protection in the transmitted signal. The information transmitted is spread in both the frequency and time domains and the effects of channel distortions and fades are eliminated from the recovered signal in the receiver. This is achieved even when the receiver is in a location with severe multipath propagation, whether stationary or mobile.

Efficient utilisation of the spectrum is achieved by interleaving multiple programme signals and by the system’s ability to operate additional transmitters as gap fillers in a single frequency network (SFN). A gap-filling transmitter in this arrangement receives and re-transmits the Eureka 147 signal on the same frequency.

Major System Features

Like almost all digital radio systems, Eureka 147 uses standard audio compression techniques and COFDM. As Eureka 147 was the first standardised digital radio system, the audio compression techniques used in all Eureka 147 implementations are now somewhat dated.

A Eureka 147 transmission has an emission bandwidth of 1.536 MHz, which is capable of providing a range of useful data rates depending on the level of protection. The multiplex contains audio programs; program associated data and, optionally, other data services. Each audio program or data service is independently error protected with a variable coding overhead, the amount of which depends on the requirements of the broadcasters (transmitter coverage and reception quality). A specific part of the multiplex contains information on how the multiplex is configured, so that a receiver can decode the signal correctly, and, possibly, information about the services themselves, the links between different services, and conditional access information for subscription services.

Eureka 147 is a mature system with 29 standards and related documents published by the European Telecommunication Standards Institute (ETSI). The ITU has included details of the Eureka 147 system in its Digital Sound Broadcasting (DSB) Handbook and Recommendations BS.1114 and BO.1130.

Modes of Operation

Eureka 147 provides four transmission mode options that allow for a wide range of transmission frequencies, between 30 and 3000 MHz, and network configurations. For the nominal frequency ranges, the transmission modes have been designed to provide good mobile reception by overcoming multipath echoes, which occur when the signal bounces off buildings and other objects and receivers must deal with multiple and slightly out of phase versions of the same signal.

Mode I is most suitable for a terrestrial SFN in the VHF range, because it allows the greatest distances between transmitters. Mode II is most suitable for hybrid satellite/terrestrial transmission up to 1.5 GHz and local radio applications that require one terrestrial transmitter. Mode II can also be used for a medium to large scale SFNs in the L Band by inserting, if necessary, artificial delays at the transmitters and/or by using directive transmitting antennas. Mode III is most appropriate for cable, satellite and complementary terrestrial transmission, since it can be operated at all frequencies up to 3 GHz for mobile reception and has the greatest phase noise tolerance. Mode IV is most suitable for medium to large scale SFNs in the L Band while still accommodating mobile reception at reasonable highway speeds (up to approximately 120 km/h). However, it is less resistant to degradation at higher vehicle speeds than this.

Table A.1: Eureka 147 Transmission Parameters

System Parameter	Transmission Mode			
	I	II	III	IV
No. of radiated carriers	1536	384	192	768
Nominal Maximum transmitter separation for SFN	96 km	24 km	12 km	48 km
Nominal frequency range for mobile reception	≤ 375 MHz	≤ 1.5 GHz	≤ 3 GHz	≤ 1.5 GHz
Speed/Coverage trade off	No	No	No	Yes
Frame Duration	96 ms	24 ms	24 ms	48 ms
Total Symbol Duration	1246 μs	312 μs	156 μs	623 μs
Useful Symbol Duration	1000 μs	250 μs	125 μs	500 μs
Guard Interval Duration	246 μs	62 μs	31 μs	123 μs
Null Symbol Duration	1297 μs	324 μs	168 μs	648 μs

Data Capacity

Audio and data services are carried in the main service channel (MSC) of the Eureka 147 multiplex. This channel supports a gross data rate of 2.304 MBps. However, the net data rate (e.g., the actual capacity available for use) depends on the protection level applied to services. For audio only services the net capacity of the ensemble varies between 783 (highest protection) and 1728 kbps (lowest protection). The corresponding range for data only services is 576 and 1728 kbps. At a median protection level the available net capacity for both audio and data services is 1.152 MBps.

Within the MSC each audio or data service is carried in a subchannel. Up to 63 subchannels can be supported, each of which is treated individually as far as error protection is concerned.

Data Services

Each audio program contains PAD with a variable capacity (minimum 667 bps, up to 65 kbps) which is used to convey information together with the sound program. Typical examples of PAD applications are dynamic range control information, a dynamic label to display program titles or lyrics, speech/music indication and text with graphic features.

Additionally, general data may be transmitted as a separate service. This may be either in the form of a continuous stream segmented into 24 ms logical frames with a data rate of n x 8 kbps (n x 32 kbps for some code rates) or in packet mode, where individual packet data services may have much lower capacities and are bundled in a packet sub multiplex. A third way to carry independent data services is as a part of the Fast Information Channel (FIC) that carries multiplex control and service information. Typical examples of independent data services that could use the FIC are a Traffic Message Channel, correction data for Differential GPS and paging.

Some elements of Service Information (SI) data can also be made available to the listener for program selection and for the operation and control of receivers. For example, the name of a program service; the program type, title and language; transmitter identification and controls for switching to traffic reports, news flashes or announcements.

Number of audio services in a multiplex

Eureka 147 uses MPEG 1 Layer II and MPEG 2 Layer II audio compression standards and permits full data rate coding at the sampling frequency of 48 kHz and half data rate coding at the sampling frequency of 24 kHz. Half data rate coding is not fast enough to capture all of the information in a speech signal so this sampling rate is only used where some distortion.

Eureka 147 is capable of processing mono, stereo and dual channel (e.g., bilingual) programs. A range of encoded data rate options are available (8, 16, 24, 32, 40, 48, 56, 64, 80, 96, 112, 128, 144, 160 or 192 kbps per monophonic channel). In stereophonic or dual channel mode, the encoder produces twice the data rate of a mono channel. The range of possible options can be utilised flexibly by broadcasters depending on the quality required and the number of sound programs to be broadcast.

A stereophonic signal may be conveyed in the stereo mode, or particularly at lower data rates in the joint stereo mode. This mode, typically used at 144 - 224 kbps, uses the redundancy and interleaving of the two channels of a stereophonic program to maximise the overall perceived audio quality.

The degree of error protection (and hence ruggedness) can also be varied to meet the needs of the broadcasters. In the case of audio services, five protection levels (1 to 5) have been specified in order to cater for a variety of applications. Level 5 affords the lowest protection and is designed for cable systems. It allows a high number of program services, but does not have the strong error protection necessary for operation in multipath environments. Protection Level 3 is better suited to mobile operation. To allow more flexibility in accommodating subchannels, Protection Levels 4 and 2 have also been introduced with somewhat weaker and stronger performance than Protection Level 3 (respectively). Protection Level 1 is suited to applications with a very high sensitivity to transmission errors while Protection Level 4 is intended for less demanding applications (for example services addressed to fixed receivers).

Table A.2 outlines the typical number of services that can be delivered for a selection of audio data rates for different levels of error protection.

Table A.2: Example of possible number of programs

Audio data rate (kbps)	Protection level (increasing protection)				
	5	4	3	2	1
24*	N/A	64	48	36	24
32	54	41	36	29	24
64	27	20	18	14	12
128	13	10	9	7	6
192	9	7	6	5	4
224	7	6	5	4	3
256	6	5	4	3	3

** At most audio data rates, Eureka 147 uses Unequal Error Protection an error protection procedure which allows the bit error characteristics to be matched with the bit error sensitivity of the different parts of the audio frame. At the lowest data rate, 24 kbps, Eureka 147 uses Equal Error Protection, an error protection procedure which ensures a constant protection of the bit stream.*

Audio Quality

ITU R Recommendation BS.1115 specifies use of MPEG 1 Layer II at 256 kbps (stereo mode), for broadcast applications requiring CD quality. This recommendation is based on subjective listening tests undertaken in 1992. At the time, MPEG 1 Layer II at 192 kbps (joint stereo mode) was also tested but was found to only marginally meet the audio quality requirement. Additional tests in 1993 failed to reveal sufficient improvement in the codec to warrant inclusion of this lower data rate in the ITU recommendation.

Further listening tests were performed in 1995, as part of the US Electronic Industries Association's (EIA) evaluation of digital radio systems. A range of audio coding systems were tested including MPEG 1 Layer II at 224 and 192 kbps (joint stereo modes). The findings of this work indicate the MPEG 1 Layer II codec at 224 kbps is capable of meeting the basic audio quality criteria specified by the ITU R. The lower rate of 192 kbps again failed to meet the required quality.

Spectrum Issues

Eureka 147 Channel Plans

In 1995, the introduction of terrestrial Eureka 147 was discussed by the European Conference for Posts and Telecommunications (CEPT) in Wiesbaden.[17] In cooperation with representatives of regional and international organisations such as the EBU, the European Commission and the ITU a total of 73 channels to be used for future and current digital audio broadcasting services was agreed. Each channel is 1.536 MHz wide with appropriate guard bands between each channel and at the edge of each band.

[17] Final Acts of the CEPT T_DAB Planning Meeting (3)", Maastricht 2002,CEPT, http://www.ero.dk/52EB3135-F356-49FF-A970-B32D2C745921?frames=0

The European CEPT channel plan encompasses four frequency bands, namely VHF Bands I, II and III and L Band. Allotments were made to allow the implementation of two Eureka 147 ensembles in any given country or area in Europe. The majority of these allotments were in VHF Band III and the lower part of the L Band (1452 MHz 1467 MHz). Allotments in the 230 240 MHz sub band of VHF Band III are subject to coordination with national defence users and the L Band was divided into terrestrial and satellite segments. Further consideration of L Band allotments was made at a second CEPT conference at Maastricht in 2002.

A second channel plan has been developed for Canada that covers only the L Band. This plan also provides for 23 channels, but with different guard bands to the CEPT Plan.

Comparing the characteristics of the two plans, the Canadian channel plan provides an interchannel guard band some 18% greater than the CEPT channel plan. Maximizing the spacing between adjacent channels is desirable, as this contributes to improved adjacent channel isolation which results in less stringent implementation constraints. In contrast, the CEPT channel plan trades off a larger interchannel guard band for increased guards at the band edges to facilitate sharing with other services operating near the band edges.

To facilitate receiver tuning and minimize scan times, manufacturers will assume, or at least prioritise, the use of certain centre frequencies as defined by the CEPT and/or Canadian channel plans. The use of ”non standard” frequencies could result in the need for manual tuning or, alternatively, require the receiver to undertake a complete scan of the band(s) based on the 16 kHz grid spacing. The latter is likely to take considerably longer and could be seen as a distinct disadvantage. Although manufacturers have been encouraged to incorporate the Canadian channel plan in their designs, it remains unclear what level of support will be afforded to the plan and whether there are cost implications for manufacturers in supporting both channel plans.

For Australia, there is a further complication if VHF Band III is used for digital radio. In this scenario, adoption of the Canadian channel plan would result in a ”mixed” frequency table arrangement (e.g., use of the CEPT channel plan at VHF Band III and the Canadian channel plan at L Band). In view of these uncertainties, adoption of the Canadian channel plan would appear justified only if significant benefits, in terms of improved adjacent channel isolation, were shown to be associated with the wider channel spacing of this plan. In the absence of any published data, the Communications Laboratory undertook measurements of the adjacent channel isolation afforded by the two channel plans, using a limited range of transmitting and receiving equipment available at that time. The results of these tests indicate no significant difference in adjacent channel performance.[18]

Planning Parameters

The planning parameters that could be used for the implementation of Eureka 147 services draw on a number of ITU and European sources:

The ITU DSB Handbook

EBU ”Technical bases for T DAB services network planning and compatibility with existing broadcasting services,” Document BPN 003 Rev. 1, May 1998;

[18] Communications Laboratory Technical Note 99/01, ‘The impact of European and Canadian L-Band channel spacings on adjacent channel operation”, 20 April 1999.

Chester 97, "The Chester 1997 multilateral coordination agreement relating to the technical criteria, coordinating principles and procedures for the introduction of terrestrial digital video broadcasting (DVB T)," 25 July 1997;

ITU R Recommendation BT.1368, "Planning criteria for digital terrestrial television services in the VHF/UHF bands," 14 April 1998.

Propagation Properties

General aspects of Propagation Properties are covered in the Spectrum Usage section of this report. The two bands in which Eureka 147 are likely to be implemented are VHF Band III and L Band.

VHF Band III

VHF Band III is well suited to the provision of terrestrial digital radio services over large coverage areas. The frequencies are still sufficiently low for good reception in moving vehicles of Eureka 147 Mode 1 transmissions. VHF Band III has less man made noise than VHF Bands I and II and does not suffer from a number of the anomalous propagation characteristics which are a problem in VHF Band I.

L-Band (1452-1492 MHz)

L-Band can be used for both terrestrial and satellite digital radio services. L Band may be used to provide the following types of coverage, assuming average terrain conditions:

- small local coverage areas up to a radius of approximately 35 to 40 km using a single, moderate power transmitter;
- larger local area coverage ranging up to a radius of approximately 60 km using a single main transmitter of moderate power and augmented by a number of gap fillers and coverage extenders;
- large area coverage (> 60 km radius) can be achieved by the use of single frequency networks employing a number of moderately spaced synchronized transmitters; and
- coverage along corridors or motorways using repeaters employing highly directional antennas (e.g., coverage extenders).

The higher frequency, shorter wavelength of an L Band transmission means that it is severely affected by local obstructions to a degree that is not encountered at VHF Band III. Conversely, the much smaller transmit antennas lend themselves to small cellular networks with discretely placed antennas. Also, the much smaller receive antenna would be attractive for small portable applications.

Present indications are that L Band is less attractive to radio broadcasters than VHF. One reason is the different ways that VHF and L Band signals propagate over distance. There is a concern that the higher building penetration losses of L Band transmissions make it less attractive than VHF Band III for indoor reception. There have been a number of studies to assess how different buildings attenuate L Band transmissions and, while they show that attenuation can be large, they show that L Band can be used to provide indoor reception with a well designed terrestrial retransmission network. Canadian authorities consider L Band to be suitable for terrestrial digital radio services and are using only L Band for their Eureka 147 services. In the US, S Band has been used for terrestrial digital radio repeaters and GSM phones have been implemented at 1800 MHz and can provide adequate indoor reception.

Recent system developments

Digital radio is likely to turn from a simple audio-only service, merely simulcasting existing analogue programmes, into a far more interactive and rich experience across several platforms including DAB, using scrolling text and on demand digital services. This section describes some technical developments of the Eureka 147 DAB system, as performed by the WorldDAB Forum.

As this section shows, the technical possibilities of DAB are practically unlimited. The challenge is to harness the technical developments and to restrict them reasonably to those for which an international consensus of broadcasters, manufactures and other players could be reached.

Multimedia Object Transport (MOT)

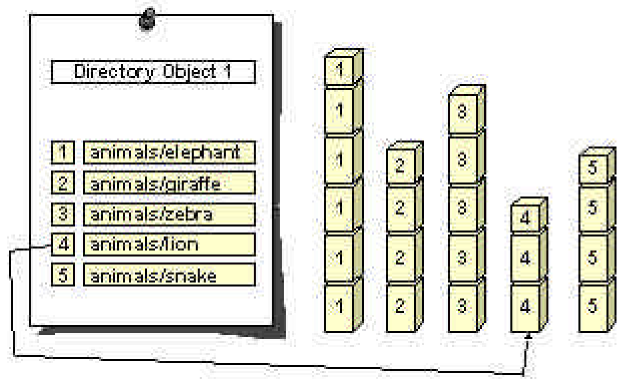
The MOT protocol allows the standardised transport of audio-visual information, such as still pictures and web pages. It can be used in the PAD and packet mode. MOT is particularly suitable for two applications: Broadcast Website (BWS) and Slide Show (SLS).

The basic principle of the MOT data carousels[19] is that each file to be broadcast is divided into segments of equal length and then the segments for all files are repeated cyclically in the broadcast stream. Each segment is tagged with an identifier to say which file it belongs to and a segment number to identify which segment of the file it is. Segmenting the file in this way means that the system will still work in an error-prone channel because, even for large files, the minimum amount of data that must be received without error is just a segment rather than the whole file. If a segment is received in error, the receiver can just wait for the next time that segment is broadcast, and the file identifier and segment number allow the receiver to correctly reconstruct each file.

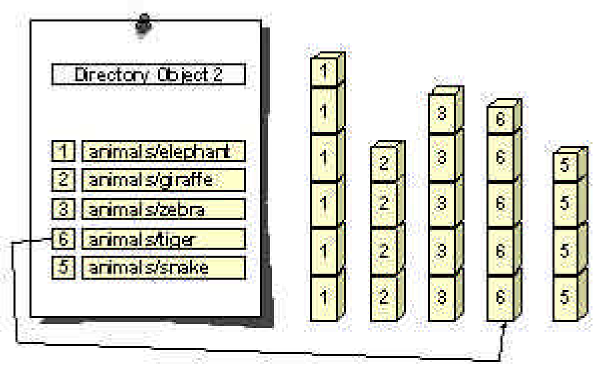
This on its own, however, is not sufficient; with a "sea" of segments, the receiver can reconstruct the files but cannot know either how to access them or how to manage them. What is needed is a "table of contents" for the carousel that contains a list of all the files contained within the carousel. With suitable version control applied to this "table of contents," it is possible to detect any change to the carousel simply by examining the version of the table of contents. If a file is changed, the version number for the file will change. This will, in turn, change the "table of contents," which will result in a change in its own version number. A simple comparison of the "table of contents" before and after the change allows the receiver to determine exactly what has changed, and to perform any cache management as appropriate.

[19] EN 301 234 V1.2.1 Digital Audio Broadcasting (DAB); Multimedia Object Transfer (MOT) protocol.

In MOT, the "table of contents" function is handled by the MOT Directory Object and its operation is illustrated below:



If we replace the file animals/lion with a new file called animals/tiger, the carousel would then appear as shown below:



The receiver can tell that the carousel has changed because the Directory Object has a new version, and by comparing the old and new Directory Objects, it can immediately determine that the file animals/lion has been replaced by animals/tiger.

The MOT Directory Object serves two functions:

To provide reliable management of the files so that any changes to the carousel are understood by the receiver.

To provide a name and other information for each file so that it may be accessed by an application.

Dynamic Label

This application carries text information and control characters with a length up to 128 characters in the PAD channel. It requires a simple alphanumeric text display of 2 lines, 32 characters each.

If the length of the text to be displayed is longer than 64 characters, the text can be incremental or scrolling.

Broadcast Website

BWS is a local interactive service; the user selects information already received by a browser. This "radio web" service allows the access to a limited number of websites, as chosen by the broadcaster ("walled garden"). BWS can be rendered either by a PC or a car navigation platform using a ¼ VGA display (320 x 240 pixels). HTML version 3.2 and a storage capacity of 256 kB are required.

Slide Show

This application involves sequences of still pictures (JPEG or PNG). The order and presentation time of this service are generated by the broadcaster. The transmission time depends primarily on the file sizes of the pictures and the chosen PAD data rate. For example, a CD cover coded as JPEG 320 x 240 requires a transmission time of 22s (PAD or packet mode data rate of 16 kbps is assumed). No local interaction is required.

A visual component, associated with audio, would potentially greatly help radio advertisers to increase advertising revenue. For example, instead of talking about the new model Volvo had just released, it would be good if we could see some pictures while we hear about its great features.

Electronic Programme Guide (EPG)

The DAB Electronic Programme Guide (EPG) allows programmers to signpost on a screen on the radio their key music positions, programmes and benchmark features, and set up opportunities to record or auto-retune the radio to their station.

Schedules can be sent to the receiver several days in advance of broadcasts, allowing opportunity to highlight and lock listeners into a new on-air activities early on. They can also be updated frequently to reflect last-minute changes to on-air output.

Experience of Television EPGs show that they can build station loyalty and time spent watching, and provide a significant enhancement to recall of on-air promotional trails.

It is expected that the EPG will become a standard feature on many DAB Digital Radios, as it has become a worldwide technical standard that can be freely adopted by receiver manufacturers. The EPG was the result of a two-year task force made up of broadcasters and receiver manufacturers working together within WorldDAB, the forum that promotes development of Digital Radio to the Eureka 147 standard.

As in TV, EPG will be useful to help to user to find, preview, select, listen and record radio programmes, particularly if there are many, possibly several hundreds, radio programmes in a given area.[20] The EPG will be used to provide programme listings information for both audio and data services and as a mechanism for the user to select services, programmes and related content. A key requirement is that the EPG must work on a range of receivers with differing display capabilities, resources and back-channel capabilities. To achieve this, a flexible multi-layer structure has been defined. The EPG data is broken down into service information (ensembles and services) and programme information (schedules, programmes, groups and

[20] Currently there are 320 DAB radio programmes on air in the UK, including 50 in London.

events). Additionally programmes and events can be linked together into groups (e.g. for grouping programmes together into serials or series).

EPG will be useful to promote new programmes and to attract new listeners. It is also enable for future technologies such as Personal Media Recording (DAB equivalent of PVR). Manual or automatic time-shifting of the programme will be possible for the user to choose what and when they want to listen.

An EPG standard "XML Specification for DAB Electronic Programme Guide" is being developed by WorldDAB.[21] Work is still continuing into the transportation and compression of the EPG data. EPG is currently being broadcast experimentally on 8 multiplexes in the UK.

DAB Virtual Machine (DAB Java)

Analogous to DVB Multimedia Home Platform (MHP), but suitably scaled down to fit into narrow-band DAB channel, DAB Java provides a flexible and extendible platform (middleware) for all new DAB data services. DAB Java is standardised by ETSI.[22] The platform enables the rapid implementation and deployment of new business ideas by enabling the applications (and applets) to access DAB resources. Future data services for DAB will be realized most efficient based on DAB Java in terms of time to market and platform independence. This approach enables DAB to be integrated in large scaled Java – based software environments, e.g. cars using widely accepted standards.

The concept of virtual machine has been chosen to allow for execution of any DAB applications independently of the hardware specific configuration. The DAB Java Framework is divided in three basic modules or packages: a) a DAB-specific extension of the Java API, b) a runtime support for the DAB applications execution environment, and c) a DAB I/O package for signalling the DAB Java extension over the DAB signal.

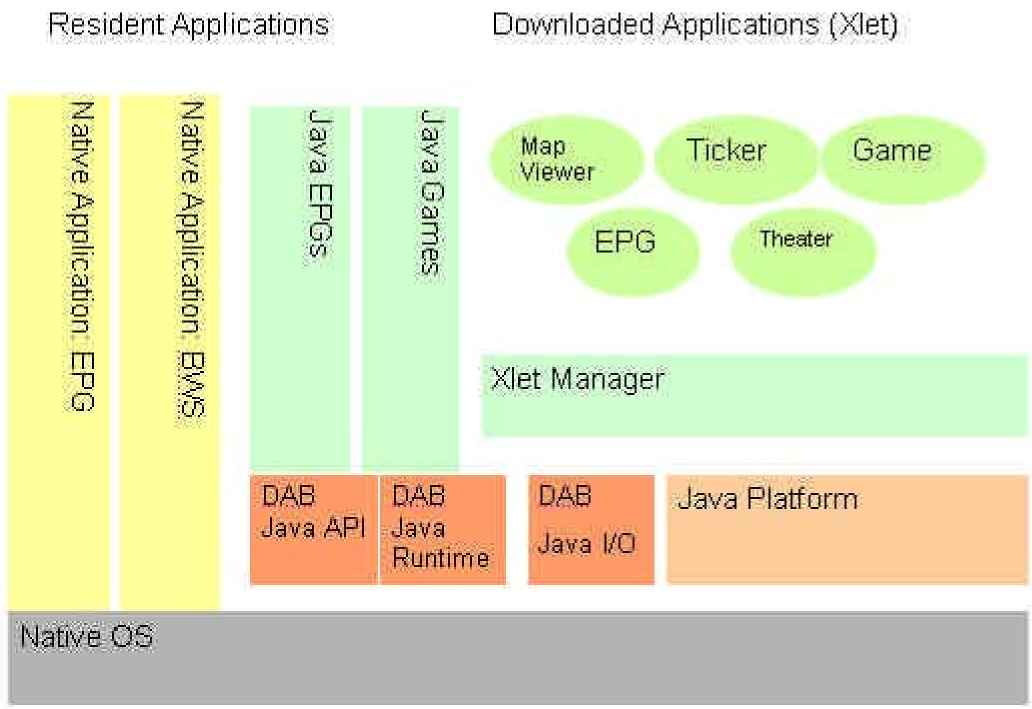
End-to-end reference implementations have been successfully developed to demonstrate the benefits and new possibilities of DAB Java. These implementations include an EPG application, a BWS application, a stock market ticker and some local-interactive games. The BBC has developed an interactive DAB Java – based application "Composer Biographies." Bosch has demonstrated an integration of DAB Java in an OSGI-based telematics system (GPS device).

[21] TS 102 818 v1.1.1 Digital Audio Broadcasting (DAB); XML Specification for DAB Electronic Programme Guide (EPG), ETSI

[22] TS 101 993 V1.1.1 (2002-03) Digital Audio Broadcasting (DAB);A Virtual Machine for DAB: DAB Java Specification, ETSI

Figure A.1 below shows the architecture of DAB Java.

Figure A.1



The development and implementation of DAB Java requires close cooperation of content, service, network providers and terminal manufacturers.

DAB Receiver Interfaces

In order to introduce new applications in the mature market with millions of DAB receivers deployed, it is essential to allow the legacy receivers to connect to the new application decoders via an agreed interface. To this end, The WorldDAB Forum has developed a specification for the Receiver Data Interface (RDI).[23] Nevertheless, as RDI has some technical limitations (e.g. flexibility, fixed bandwidth), it has been decided to develop a new interface. The WorldDAB Forum and the DRM Forum have agreed to cooperate in defining a generic physical USB interface for all digital radio receivers. Furthermore, a generic low level driver interface based on Digital Command Set for Receivers (DRCS) specification will be developed, taking into account of copy protection and digital rights management issues.

Conditional Access

The DAB system already includes a comprehensive conditional system (see Chapter 9 of EN 300 401). Further work is now underway to develop a simple, yet reliable system to be used in

[23] EN 50255 Digital Audio Broadcasting system; Specification of the Receiver Data Interface (RDI), CENELEC

commercial receivers using a common scrambling algorithm and a common receiver interface, however allowing the use of different commercial CA systems such as Simulcrypt and Multicrypt.

SBR Layer II

Spectrum Band Replication (SBR) is a process, proposed by Coding Technologies and now standardised within MPEG-4 Audio, designed to potentially improve spectrum efficiency of the DAB system by reducing the audio bit rate for the same quality, while retaining backwards compatibility. Some initial studies indicate that about 30% improvement could be achieved.

The EBU Project group B/AIM (Audio In Multimedia) is carrying studies on error sensitivity and compatibility with non-SBR receivers. Some preliminary results show that the inclusion of SBR in the DAB system does not significantly degrade the C/N performance of the DAB system, neither in terms of Threshold of Audibility (TOA) nor Point of Failure (POF).

Studies are continued on balancing the benefits and drawbacks of SBR. The matters to be addressed involve the increase of complexity (and thus cost) of the receiver and the related IPR issues. No decision has been taken by the WorldDAB Forum to date about the viability of using including SBR into the standard and recommending its incorporation into commercial receivers.

File caching in the receiver

The WorldDAB Forum has now established a specification for using an optional caching facility in the receiver. The user will benefit from a so-called "rewind radio," which will allow listening of the latest programme at any time. The caching device will also allow the user to use the DAB receiver as a PVR (Personal Versatile Recorder) device for time-shifted playout of audio events (with or without associated data). It should be pointed out the use of caching may change the way how people access and enjoy radio listening. It potentially widens the programming possibilities offered by the broadcaster but also introduces new technical and operational problems (copyright, EPG, etc).

In September 2003 RadioScape which specialises in digital radio software launched a new module called RS200L.[24] One of the features of this module is the inclusion of Rewind Radio that enables about ten minutes of audio to be stored on chip RAM. This can be used to listen to a news clip again or time shift by pausing and resuming the radio. The module has been designed using the DRE200 chip from Texas Instruments, which is probably one of the world's best selling receiver chips for the EU 147 standard. This chip has now been superseded by a new version, DRE310,[25] that can decode more than one channel simultaneously and includes time-shifted radio, announcement support, service linking (FM/DAB ensemble switching), TII (Transmitter Identification Information) and MP3/Windows Media Audio CD support.

TopNews

TopNews is a commercial name for Bosch/Blaupunkt's system which allows broadcasters (and multiplex providers) to download via a suitable DAB data channel (e.g. MOT, MSC packet mode) the news and other audio files or other objects coded in MP3 to the receiver.[26] The user is appropriately informed of the existence of these audio objects and could access them at their convenience. The broadcaster is responsible for contents and needs to update the audio file

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[24] http://www.worlddab.org/pressreleases/RADIOSCAPE-LAUNCHES-THE-RS200L.pdf
[25] http://www.worlddab.org/pressreleases/TI-uses-Radioscape-23-06-03.pdf
[26] WorldDAB TC 075 available from http://www.worlddab.org/tc_presentations/2

contents regularly. There is no need for return link to the service provider. This "audio anytime" system is particularly attractive for in-car applications.

IP datacasting in DAB

The DAB system is capable of carrying IP packets (datagrams) using IP/UDP protocol.[27] As these packets travel unidirectionally from a service provider to many users simultaneously, this is a form of IP Multicasting, e.g., pushing the same contents to several users concurrently. The IP datagrams are tunnelled through a DAB packet mode service component (SC). This is done by encapsulating the IP datagram in an MSC data group on packet mode transport level. It is not necessary to establish a connection between the transmitter and the user prior to the transmission of data.

For connection oriented point-to-point transport, TCP has to be used (rather than UDP). TCP requires an interaction channel for the return flow of acknowledgements.

Further work is necessary to be carried out similar to that performed by the DVB-IPI project in order to specify the discovery and selection of the data services by the user.

The Digital Video Broadcasting (DVB) Project has developed a data broadcasting standard describing an IPv4 and IPv6 datagrams encapsulation in MPEG-2 transport stream. This system is commonly called Multi-Protocol Encapsulation (MPE) or Data Piping[28] and includes dynamic address resolution, multicast group membership and other supporting procedures and protocols. The overhead due to encapsulation is reasonably low, e.g., below 3%.

IP datacasting is an interesting option for the DAB systems required to work with IP-enabled devices such as mobile phones and PDAs. The IP layer could be used as a common communications layer between the two systems. IP datacasting over DAB will bring the data content such as moving pictures, audio, web pages, computer programmes and software upgrades reliably to each user (or a group of users) and will thus expand significantly market opportunities of DAB. IP datacasting will pave the way towards the personalisation of broadcast services.

TPEG transport in DAB

It is well known to all broadcasters that radio is an ideal (and the cheapest) medium to inform travellers about the road conditions and traffic jams – provided that such information is timely and relevant, in the correct location. Currently analogue FM radio uses a well-established RDS-TMC (Traffic Message System) system. However, the TMC is essentially limited to inter-urban road events and every decoder must have a location database to interpret any message received.

TPEG was developed by the EBU to overcome these limitations. TPEG delivers very rich location referencing information with every message, so that receivers do not need a location database. Thus, navigation systems which are now becoming a standard commodity in the car can "machine read" the location content and localise an event directly onto the map display. A text-only device (such as a PDA) is able to present locally found names such as a railway station name and a platform number directly to an end user as a text message. Such a message can be rendered in the language of choice of the end user. TPEG can filter the information to avoid receiver overload, so that end users can select massages on any number of criteria, such as the type of location, mode of public transport, direction of travel, event, etc.

[27] ES 201 735 V1.1.1 Digital Audio Broadcasting (DAB); Internet Protocol (IP) Datagram Tunnelling
[28] EN 301 192 V1.3.1 (2003-05) Digital Video Broadcasting (DVB); DVB specification for data broadcasting

TPEG can be transported within the DAB system in the Transparent Data Channel (TDC) in a stream-like format; bytes come out in the same order they go in.[29] The TDC Specification allows TPEG data to be carried in three modes: packet mode, stream mode and X-PAD. Nevertheless, this approach which is specified in the present version of the DAB standard, involves several problems in terms of reception reliability and interpretation. It has therefore been proposed to transport TPEG as one of the multimedia applications in the MOT data channel. This would imply the following main advantages: MOT is already implemented in most receivers and enables efficient object compression, power saving and delta updates and has much lower overhead than TDC.

Advanced demodulation technique for COFDM

The Communications Research Centre Canada (CRC) developed an advanced COFDM demodulation technique[30] which reduces the effect of the Doppler effect and therefore increases the maximum speed, allowing vehicle speeds up to 140 km/s while achieving a target bit error rate (BER) of 10-4 . Canadian DAB broadcasters use L-Band (1452 to 1492 MHz) and would like to use Transmission Mode IV instead of Mode II, because the former allows for a larger separation distance between on-channel re-transmitters than in the case of Mode II. However, Mode IV in L-Band limits the speed to less than 100 km/h, so this new technique could help. Further studies are required to investigate whether this technique could be useful for VHF bands and whether the chip manufacturers could accommodate it readily into their chip design.

Technical Standards

International Standards

ETSI Standards[31]

Eureka 147 standards are formalised by ETSI and are available for download. The current list of ETSI standards relating to Eureka 147 are in Table A.3. The main ETSI standard for Eureka 147 is EN 300 401.

Table A.3: ETSI Standards relating to Eureka 147

Number	Title
EN 300 401 V1.3.3 (May 2001)	Digital Audio Broadcasting (DAB); DAB to mobile, portable and fixed receivers (THIRD EDITION)
EN 300 797 V1.1.1	Digital Audio Broadcasting (DAB); Distribution interfaces; Service Transport Interface (STI)
EN 300 798 V1.1.1	Digital Audio Broadcasting (DAB); Distribution interfaces; Digital baseband In-phase and Quadrature (DIQ) Interface
EN 301 234 V1.2.1	Digital Audio Broadcasting (DAB); Multimedia Object Transfer (MOT) protocol
EN 301 700 V1.1.1	Digital Audio Broadcasting (DAB); Service Referencing from FM-RDS;

[29] Guidelines for TPEG in DAB, B/TPEG Plenary Group 00/113 available from www.ebu.ch/bmc_btpeg.htm
[30] Thibault, Zhang, Boudreau, Taylor, Chouinard: Advanced Demodulation Technique for COFDM in Fast Fading Channels, IBC 2003 Proceedings, p. 416 to 422
[31] EBU BPN 062

Number	Title
	Definition and use of RDS-ODA
EN 302 077 V1.1.1	Electromagnetic compatibility and Radio spectrum Matters (ERM); Harmonised EN for Terrestrial Digital Audio Broadcast (TDAB) equipment used in the sound broadcasting service.
ES 201 735	Digital Audio Broadcasting (DAB); Internet Protocol Datagram Tunnelling
ES 201 736 V1.1.1	Digital Audio Broadcasting (DAB); Network Independent Protocols for Interactive Services
ES 201 737 V1.1.1	Digital Audio Broadcasting (DAB); DAB Interaction Channel through GSM / PSTN / ISDN / DECT
ETS 300 799	Digital Audio Broadcasting (DAB); Distribution interfaces; Ensemble Transport Interface (ETI)
TR 101 495 V1.1.1	Digital Audio Broadcasting (DAB); Guide to DAB Standards; Guidelines and Bibliography
TR 101 496-1 V.1.1.1	Digital Audio Broadcasting (DAB); Guidelines and Rules for Implementation and Operation
TR 101 496-2 V.1.1.2	Digital Audio Broadcasting (DAB); Guidelines and Rules for Implementation and Operation
TR 101 496-3 V.1.1.2	Digital Audio Broadcasting (DAB); Guidelines and Rules for Implementation and Operation
TR 101 497 V1.1.1	Digital Audio Broadcasting (DAB); Rules of Operation for the Multimedia Object Transfer Protocol
TS 101 498-1 V1.1.1	Digital Audio Broadcasting (DAB); Broadcast Website Application, Part 1:User Application Specification
TS 101 498-2 V1.1.1	Digital Audio Broadcasting (DAB); Broadcast Website Application, Part 2: Basic Profile Specification
TS 101 499 V1.1.1	Digital Audio Broadcasting (DAB); MOT Slide Show; User Application Specification
TS 101 735 V1.1.1	Digital Audio Broadcasting (DAB); Internet Protocol Datagram Tunnelling
TS 101 736 V1.1.1	Digital Audio Broadcasting (DAB); Network Independent Protocols for Interactive Services
TS 101 737 V1.1.1	Digital Audio Broadcasting (DAB); DAB Interaction Channel through GSM / PSTN / ISDN / DECT
TS 101 756 V1.1.1	Digital Audio Broadcasting (DAB); Registered Tables
TS 101 757 V1.1.1	Digital Audio Broadcasting (DAB); Conformance Testing for DAB Audio
TS 101 758 V2.1.1	Digital Audio Broadcasting (DAB); DAB Signal Strengths and Receiver Parameters
TS 101 759 V1.1.1	Digital Audio Broadcasting (DAB); DAB Data Broadcasting Transparent Data Channel
TS 101 860 V1.1.1	Digital Audio Broadcasting (DAB); Distribution Interfaces; Service Transport Interface (STI); STI Levels
TS 101 993 V1.1.1	Digital Audio Broadcasting (DAB); A Virtual Machine for DAB: DAB Java Specification
TS 102 818 V1.1.1	Digital Audio Broadcasting (DAB); XML Specification for DAB Electronic Program Guide (EPG)

Receiver Standards

European receiver standards have been developed by CENELEC, IEC and national standards bodies (e.g., UK). A list of relevant receiver standards is in Table A.4.

Table A.4: Receiver Standards for Eureka 147

Reference	Title
CENELEC EN 50255	Digital Audio Broadcasting system - Specification of the Receiver Data Interface (RDI)
CENELEC EN 50248	Characteristics of DAB receivers
CENELEC EN 50320	The DAB Command Set for receivers
IEC 62105	Digital Audio Broadcasting System - Specification of the Receiver Data Interface (RDI)
IEC 62104	Characteristics of DAB Receivers

ITU Publications and Recommendations

The International Telecommunications Union has a number of publications and Recommendations relating to Eureka 147 and digital radio in particular. The "DSB Handbook - Terrestrial and satellite DSB to vehicular, portable and fixed receivers in the VHF/UHF bands" is an aggregation of ITU input documents and data. Relevant recommendations are in Table A.5.

Table A.5: ITU Recommendations relevant to Eureka 147

Reference	Title
BS.1115	Low data rate audio coding
BS.774-2	Service requirements for DSB to vehicular, portable and fixed receivers using terrestrial transmitters in the VHF/UHF bands
BS.1114-3	Systems for terrestrial DSB to vehicular, portable and fixed receivers in the frequency range 30-3 000 MHz
BO.789-2	Service for DSB to vehicular portable and fixed receivers for broadcasting-satellite service (sound) in the frequency range 1 400-2 700 MHz
BO.1130-4	Systems for digital satellite broadcasting to vehicular, portable and fixed receivers in the bands allocated to BSS (sound) in the frequency range 1 400-2 700 MHz

APPENDIX B Relevant World Wide Websites

Advanced Television Systems Committee (ATSC)	www.atsc.org
AsiaDAB	www.asiadab.org
Asia-Pacific Broadcasting Union (ABU)	www.abu.org.my
Audio Engineering Society (AES)	www.aes.org
BBC (DAB)	www.bbc.co.uk/digitalradio
BBC (Research and Development)	www.bbc.co.uk/rd
BBC Training (Centre for Broadcasting Skills)	www.bbc.co.uk/woodnorton
BBC World Service (Radio)	www.bbc.co.uk/worldservice
Commonwealth Broadcasting Association (CBA)	www.cba.org.uk
Crown Castle International	www.crowncastle.com
Crown Castle UK	www.crowncastle.co.uk
DAB Canada	www.digitalradio.ca
Dalet	www.dalet.com
Digital Radio Mondiale (DRM)	www.drm.org
Digital Video Broadcasting (DVB)	www.dvb.org
European Broadcasting Union (EBU)	www.ebu.ch
Financial Times (Media and Telecoms.)	www.ftmedia.com
Ibiquity (HD Radio)	www.ibiquity.com
International Telecommunications Union	www.itu.int
Lucent Technologies (Lucent Digital Radio)	www.lucent.com
National Association of Broadcasters (US)	www.nab.org
National Association of Shortwave Broadcasters US (NASB)	www.shortwave.org
National Radio Systems Committee US (NRSC)	www.nrscstandards.org
National Transcommunications Ltd. (NTL)	www.ntlradio.com
North American Broadcasters Association (NABA)	www.nabanet.com
Office of Communications UK (Ofcom)	www.ofcom.org.uk
Radio Academy (UK)	www.radioacademy.org
Real Audio	www.real.com
Roke Manor Research (UK)	www.roke.co.uk
Sadie	www.sadie.com
Sirius Satellite Radio	www.siriusradio.com
Thales Broadcast	www.thales-bm.com

World Broadcasting Unions (WBU)	www.worldbroadcastingunions.org
World Radio Network	www.wrn.org
WorldDAB	www.worlddab.org
WorldSpace Radio	www.worldspace.com
XM Satellite Radio	www.xm.com

APPENDIX C Glossary of Acronyms

AAC	Advanced Audio Coding
AAS	Advanced Application Services
ADR	Astra Digital Radio
AM	Amplitude Modulation
API	Advanced Programming Interface
ATM	Asynchronous Transfer Mode
BBC	British Broadcasting Corporation
BER	Bit Error Rate
Bit	Binary digit
Bitrate	Rate of flow of bits per second
BSS(S)	Broadcast satellite services (Sound)
BWS	Broadcast Website
CA	Conditional Access
CBC	Canadian Broadcasting Corporation
CCETT	Centre Commun d'Etudes de Telediffusion et Telecommunication (Research Laboratories of France Telecom and Telediffusion de France)
CD	Compact Disc
CDMA	Code Division Multiple Access
CEG	Consumer Equipment Group
CELP	Code Excited Linear Prediction
CEMA	Consumer Electronics Manufacturers Association
CEPT	European Conference of Postal and Telecommunications Administrations
codec	Coder / Decoder
COFDM	Coded Orthogonal Frequency Division Multiplex
CP	Continual Pilot
CRC	Communications Research Centre Canada
CRTC	Canadian Radio-television and Telecommunications Commission
DAB	Digital Audio Broadcasting
DARS	Digital Audio Radio Service
DAT	Digital Audio Tape
DAW	Digital Audio Workstation
DMB	Digital Multimedia Broadcasting
DQPSK	Differential Quadrature Phase Shift Keying

DRB	Digital Radio Broadcasting
DRDB	Digital Radio Development Bureau
DRM	Digital Radio Mondiale
DRP	Digital Radio Promotion
DSB	Double Side Band
DSL	Digital Subscriber Line
DSR	Digital Satellite Radio
DTH	Direct to Home
DTS	Digital Theatre System
DTT	Digital Terrestrial Television
DVB	Digital Video Broadcasting
DVB-H	Digital Video Broadcasting – Handheld
DVB-T	Digital Video Broadcasting – Terrestrial
DXB	Digital Extended Broadcasting, a German-funded project
EBU	European Broadcasting Union
EIA	Electronic Industries Alliance (formerly Electronic Industries Association)
EMK	Electronic Media Kiosk
EPG	Electronic Program Guide
ETI	Ensemble Transport Interface
ETS	European Telecommunications Standard
ETSI	European Telecommunications Standards Institute
Eureka	European R and D programme
FAC	Fast Access Channel
FCC	Federal Communications Commission (US)
FIC	Fast Information Channel
FM	Frequency Modulation
FDMA	Frequency division multiple access
FHG	Fraunhofer Institute (Germany)
FIC	Fast Information channel
GPS	Global Positioning System
GSO	Geostationary (Satellite) Orbit
GSM	Global System for Mobile Communications
HEO	Highly Elliptical Orbit
HVXC	Harmonic Vector Excitation Coding

IEEE	Institute of Electrical and Electronics Engineers
IFPI	International Federation of the Phonographic Industry
iMP	Integrated Media Player
IP	Internet Protocol
ISDB-TSB	Integrated Services Digital Broadcasting – Terrestrial for Sound Broadcasting
ITU	International Telecommunications Union
ITU-R	ITU Radiocommunications Sector
IBAC	In-Band Adjacent Channel
IBOC	In-Band / On-Channel
JPEG	Joint Photographic Experts Group
kbps	1000 bits per second
LCD	Liquid Crystal Display
LF	Low Frequency
LW	Long wave
LEO	Low earth orbit (satellite)
MATS	Mobile Aeronautical Telemetry Services
MCI	Modular Control Interface
MD	Mini Disc
MDI	Multiplex Distribution Interface
MF	Medium Frequency
MHP	Multimedia Home Platform
MLC	Multi-Level Coding
MP3	MPEG Audio Layer 3 (see MPEG)
MPEG	Moving Pictures Expert Group
MPS	Main Program Service
MOT	Multi-media Object Transfer
MSC	Main Service Channel
MW	Medium wave
NAB	National Association of Broadcasters (US)
NHK	Nippon Hoso Kyokai (Japan Broadcasting Corporation)
NICAM 728	Near-Instantaneously Companded Audio Multiplex (728 is bit rate in kbps)
NRSC	National Radio Systems Committee (an industry sponsored technical standard setting body, co-sponsored by CEMA and NAB in the US)
OEM	Original Equipment Manufacturer
OFDM	Orthogonal Frequency Division Multiplexing

P2P	Peer-to-Peer Networking
PAD	Programme Associated Data
PC card	A plug in card for a Personal Computer, which allows it to receive DAB.
PDA	Personal Digital Assistant
PNG	Portable Network Graphics
POF	Point of Failure
PTY	Programme Type Codes
PVR	Personal Versatile Recorder
QAM	Quadrature Amplitude Modulation
QPSK	Quadrature Phase Shift Keying
RAM	Random Access Memory
RDS	Radio Data System
RDI	Receiver data Interface
RF	Radio Frequency
RSCI	Receiver Status and Control Interface
SBR	Spectral Band Replication
SCA	Subsidiary Communications Authorization
SDC	Service Description Channel
SDI	Service Distribution Interface
SFN	Single Frequency Network
S-DAB	Satellite DAB
SDARS	Satellite Digital Audio Radio Service
S-DMB	Satellite Digital Multimedia Broadcasting
SIS	Service Information Service
SLS	Slideshow
SMIL	Synchronized Multimedia Integration Language
SR	Sveriges Radio (Swedish Radio)
SSB	Single Side-Band
STL	Studio-to-Transmitter Link
SW	Short-wave
Simulcasting	Simultaneous transmission of a programme
T-DAB	Terrestrial DAB
TCM	Trellis coded Modulation
TDC	Transparent Data Channel

TDM	Time Division Multiplex
TDMA	Time Division Multiple Access
T-DMB	Terrestrial Digital Multimedia Broadcasting
TMC	Traffic Message System
TMCC	Transmission and Multiplexing Configuration Control
TOA	Threshold of Audibility
TPEG	Transport Protocol Experts Group
TTSL	Total Time Spent Listening
UEP	Unequal Error Protection
USB	Universal Serial Bus
VHF	Very high Frequency
VPN	Virtual Private Networks
VRT	Belgian Public Service Broadcaster
W(A)RC	World (Administrative) Radio Conference
WiFi	Wireless technology brand (coined by WiFi Alliance)
WiMAX	Worldwide Interoperability for Microwave Access
WMA	Windows Media Audio
WorldDAB	Organisation for promoting digital radio (DAB) based on the Eureka 147 system.

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