The Eureka 147 Digital Audio Broadcasting System Adapted to the U.S.

by

Nupur Gupta

Submitted to the Department of Electrical Engineering and Computer Science in Partial Fulfillment of the Requirements for the Degree of Master of Engineering in Electrical Engineering and Computer Science

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Abstract

This thesis is a study of the Eureka 147 Digital Audio Broadcasting (DAB) system developed by a European consortium. It begins with a brief introduction to DAB in the United States. It also gives a comprehensive technical overview of the Eureka 147 DAB system. This thesis proposes a solution to concerns expressed by the broadcasting industry regarding the acceptance of this system as the DAB standard for the U. S. These concerns focus on the present requirement that all broadcasters in each metropolitan area must have the same coverage area. This solution, using a Time Division Multiple Access technique in conjunction with the Transmitter Identification Information, will show that a Eureka 147 system implementation can be configured to address these primary coverage area concerns subject to some small loss in system capacity. The basis of this solution has been outlined along with some key considerations, such as synchronization, noise power and frequency re-use, that must undergo further analysis.

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Chapter 1

Introduction

1.1 Digital Audio Broadcasting

In the past decade there has been a technological development in the broadcasting industry that would bring high-quality stereo sound to the radio broadcasting environment. This development, commonly referred to as Digital Audio Broadcasting (DAB) or Digital Audio Radio (DAR), could revolutionize the way we think about radio reception. Although the primary objective of DAB is to improve the quality of sound received through broadcasting, developers are designing systems that may be able to offer other valuable services to listeners. Such services include transmitting of faxes, newspapers, electronic images, and other multimedia. Emergency warning and traffic messaging would also be included in the list of services that could be offered by such a system. For these reasons, digital radio may well be the "sound of the future."

There is some question as to why new services offered by DAB could not be offered by a new Frequency Division Multiple Access (FDMA) system. This has to due to desired spectrum utilization and propagation characteristics of a broadcasting environment. In order to transmit at high data rates, an FDMA system would have to utilize a relatively large amount of bandwidth per channel in order to include guard bands. On the other hand, a DAB system can be designed to transmit multiple carriers that are synchronized. This means that multiple programs can be compactly transmitted with minimal spectral guard bands but still be distinguishable from each other. Therefore, even though a DAB system would utilize a large bandwidth it would still exhibit spectrum efficiency. In addition, due to the congestion of the frequency band, it is unlikely that a new system could be implemented in reasonably low to mid-range frequencies. This imposes additional constraints on the transmitted signal. Since the objective is to provide high quality sound to all receiver types, both systems required channels to be separated by a guard interval in order to allow for interference problems caused by the propagation characteristics of the environment. However, the effects of this would be more noticeable in the spectrum efficiency of an FDMA system since only one program is transmitted per channel.

When evaluating the implications of a DAB service in the United States, we see that it could once again make radio broadcasting a pre-eminent medium for disseminating information. In the past, television has succeeded in becoming the leading information resource for most Americans. In addition, Compact Discs offers better sound quality than radio. Many view that AM and FM radio broadcasting no longer provide the advantages that they were once designed for. DAB has the capability to reverse the effects that advanced technology has had on radio broadcasting. It can once again bring advantages to supporting this industry. [16, p. 4]

Initially, DAB is seen as a method for improving the sound quality of audio programs. However, improvement of sound quality should not be viewed as the only advantage to accepting such a technology for the U.S. broadcasting environment. More importantly, DAB will allow the transmission of data that will aid the American consumer to use radio broadcasting not only for entertainment purposes but as an informative resource. In addition, commercial resource providers will be able to use data transmission in their advertising. Therefore, not only will radio broadcasting gain added support from its audience but also from its sponsors. The additional support from sponsors will facilitate in bringing the cost of supporting a DAB system down. With the costs lowered, speculation indicates that the broadcasting industry will eventually migrate to accepting DAB as the leading method of radio broadcasting. This is similar to how we view FM in regard to AM. This acceptance will facilitate research in providing additional services using DAB.

This thesis is an analysis of a proposed DAB system for the United States. This system is known as the Eureka 147 DAB system. Specifically, this thesis attempts to offer a solution to some of the concerns that broadcasters have raised in regards to accepting this system as the DAB standard for the United States. However, before proposing a solution to the issues that broadcasters have raised, this thesis gives a overview of the Eureka 147 DAB system and its implementation in various other parts of the world. Furthermore, this thesis will provide a summary of the technical standard developed for Eureka 147 as well as a short explanation of certain key procedures employed in Eureka 147.

1.2 Eureka 147

One system that has been fully developed and under implementation in much of Europe and Canada is known as Eureka 147. Eureka 147 was originally developed by a European consortium of companies and partners.The list of Eureka 147 partners has now grown to include companies from all around the world. A current list of partners can be found at URL http://www.kp.dlr.de/DAB/partner.htm.

The Eureka 147 DAB system is designed to provide better sound quality than AM or FM to the radio broadcasting environment. In addition, it has the capability to offer a wide range of services that will aid consumers in integrating their lives with DAB. These services will offer consumers a means of using the broadcasting environment beyond its current day scope. It will allow them to receive messages, data, pictures and even pay services. The developers hope to make the Eureka 147 become the accepted standard for DAB worldwide. The Joint Technical Committee (JTC) of the European Broadcasting Union (EBU) and the European Telecommunication Standards Institute (ETSI) have produced a European Telecommunication Standard (ETS) 300 401 titled "Radio broadcasting systems; Digital Audio Broadcasting (DAB) to mobile, portable and fixed receivers". This standard describes the procedures for constructing the transmitting signal for the Eureka 147 DAB system. The standard includes the coding algorithms for audio, multiplexing of audio programs and data services, channel coding and modulation procedures. [17, p. 11]

Eureka 147 uses the concept of combining programs to form one signal that is transmitted over a large bandwidth. This constitutes a DAB "channel", and each channel is 1.536 MHz wide.

Currently, for every 200 kHz in the FM band, there is one program available. In addition, FM channels generally do not use adjacent frequencies. However, for every 1.536 MHz in a Eureka DAB channel, there are up to six high-quality sound programs available. In addition, general data services can also be transmitted in this large bandwidth.

Eureka 147 is designed to be implemented as a Single Frequency Network (SFN). A SFN is a network whereby a large coverage area is achieved through the use of a group of relatively low power transmitters placed at different locations, but transmitting at the same frequency. This allows a radio station's coverage area to potentially be limitless. With low power transmitters, Eureka 147 can achieve the same coverage area as much larger high power transmitters that are conventionally used in the broadcasting industry today.

The transmitted DAB signal consists of Orthogonal Frequency Division Multiplex (OFDM) symbols. Three sets of OFDM symbols constitute the makeup of a DAB frame, the main transport mechanism for the system. Each of these sets of OFDM symbols is called a channel and each channel is specifically designed to serve a particular functionality for the Eureka 147 DAB network. Initially, the transmitting data is divided into blocks before being processed. These blocks are then coded and multiplexed together to form a DAB frame channel. The channels are then partitioned and modulated to form the three sets of OFDM symbols per frame channel. Further details of the semantics of the transmitting signal is provided in Chapter 3, "Technicalities of Eureka 147" on page 25 , where the ETS 300 401 is summarized. Chapter 4 provides clarification of the coding procedures employed by the Eureka 147 DAB system.

1.3 Developments in the United States

The U.S. is currently looking to adopt a DAB system. Proponents of DAB hope to have a standard adopted by the end of this year. Along with Eureka 147, there are a variety of systems that proposed to provide DAB. Some of these systems propose to provide a DAB service within the already existing AM and FM bands. They are called In Band On Channel (IBOC) or In Band Adjacent Channel (IBAC) DAB systems. These systems transmit their digital signal intermixed with channels in the AM/FM analog system. There are also other DAB systems that propose to provide a DAB services through a satellite based broadcasting network. Eureka 147 and these other systems are undergoing evaluation by the FCC to become the adopted standard for a DAB system in the United States. Limited documentation is available on these other DAB system proposals, and they are beyond the scope of this document. References have been provided at the end of this document for further inquiry into IBOC, IBAC and satellite based DAB systems.

There are number of reasons that are hindering the Eureka 147 system from being accepted as the U.S. DAB system. The broadcasting industry has expressed its concerns and feels that Eureka 147 may alter the industry's framework besides providing superior audio quality. One of the leading issues, that has caused much concern, is the use of SFNs in the DAB system. Extending a radio station's broadcast coverage area without limit is a new concept to the broadcasters of the U.S. While there is some concern with the potentially limitless coverage of SFNs there is more concern with the combining of signals from multiple broadcasters to produce a single transmitting signal that would have an equal coverage area for all of the combined signals. As one can see, this poses a problem for broadcasters that have invested much time, effort, and money into extending their coverage area to include a larger area than their leading competitors. In addition, smaller broadcaster do not have the resources nor the money to support audiences that are outside their current coverage areas. Some do not even feel that they provide services that they would want to extend to a larger community. Hence, such a system would require broadcasters to collectively agree on adjusting their coverage areas to become identical. Obviously, this is a very difficult task and has not received any enthusiasm from the broadcasting industry.

Furthermore, if Eureka 147 is adopted, the use of its SFN capability is inevitable. Since the Eureka 147 system is a new band system, it would most likely undergo implementation in the L-band. The applicable part of L-band ranges from 1452-1492 MHz. At such high frequencies, reception of the transmitted signal requires near line of sight conditions. In order to ensure that the receiver is able to detect the signal, it must have a nearly unobstructed path to the transmitter. This is difficult to do with just one transmitter for any coverage area besides the rural flat terrain environment. Hence, multiple transmitters will be used to cover areas where blockage of the transmitted signal is likely to occur.

However, there are variations to the Eureka 147 system that people in the industry have proposed. Unfortunately, there has not been an open discussion about these variations, and as a consequence, the broadcasting industry is uncertain of accepting the Eureka 147 system. In general, very few people have knowledge of how the system works and what can be done to accommodate the concerns of the broadcasting industry. This thesis attempts to give the reader a better understanding of the Eureka 147 DAB system. In addition, the author has provided a potential solution based on an analysis of the system. It is unclear at this point whether the Eureka 147 system will be adopted as the DAB system for the U.S. broadcasting environment. The author is only attempting to offer a possible solution in order address some the major concerns of the broadcasters. It is clear however, that the DAB technology should be adopted for the U.S. in the near future. Many countries have felt that it would be beneficial to their society and are already in the implementation stage of incorporating a Eureka 147 DAB system into their environment.

Chapter 2 has been provided to give the reader a better understanding of the developments of DAB around the world, as well as some insight into the difficulties that it has encountered in the United States. Chapter 3 is a summary of the ETS 300 401 standard on Eureka 147. Chapter 4 has been provided to assist in the understanding of the technical details of Eureka 147. Chapter 5 looks at a proposal that could be used as an implementation scheme for the Eureka 147 DAB system in the United States broadcasting environment.

Chapter 2

Development of Eureka 147 Around the World and In the U.S.

2.1 Background

The Eureka 147 DAB project was initiated in 1987 in order to address the need for a high-performance broadcasting system. The Eureka 147 developers originally consisted of a consortium of European partners from Germany, France, the Netherlands and the United Kingdom. [16, p. 14]. The list of partners has since then grown to include companies from all over the world. The International Union Technical Committee of the World Conference of Broadcasting Unions recommended Eureka 147 world-wide in 1992. Also in 1992, L-Band and S-Band frequencies were designated as DAB frequencies on a world-wide basis. In 1994, the ITU Recommendations BS. 1114 and BO. 1130 standardized Eureka 147. [10, p. 2] Since then it has been undergoing implementation in much of Europe. Within the last year, it was accepted as the DAB standard for Canada. In addition, it is being evaluated by the FCC to become the standard for DAB in the United States. This chapter is a summary of Eureka 147 DAB developments around the world and in the United States. Much of this is taken from the summary on Eureka 147 developments provided by the Eureka 147 Project Office.[10] A list of discussions on Radio-L, a news group dedicated to DAB developments and issues around the world, is maintained at URL http://www.magi.com/~moted/dr. This site primarily summarizes the major DAB developments in Canada. It has a list of links to where further information concerning DAB and Radio-L discussions can be found.

2.2 Europe

European countries are looking to introduce DAB in 1997. Many of them have initiated pilot projects that reach 20-50% coverage of the population. It is speculated that 70-80% of the European population will be able to receive DAB signals by the year 2000. Many of the countries see DAB as the replacement technology for analog broadcasting. They hope that DAB will one day be the only form of radio broadcasting replacing AM and FM broadcasting.

2.2.1 United Kingdom

The UK has allocated 12.5 MHz of radio spectrum for the Eureka 147 DAB system. The frequency range is from 217.5 to 230 MHz. This will accommodate seven DAB ensembles carrying up to a maximum of six audio programs. One ensemble has been reserved for BBC National Radio. BBC will use this to transmit radio stations Radio 1, 2, 3, 4 and 5. The final program will transmit services such as sporting events, Parliament proceedings and other live events. Another ensemble has been reserved for Independent National Radio. The remaining five ensembles are to be used for local, regional and BBC radio. Each area will be provided with one ensemble carrying a maximum of six services in addition to the national radio ensembles. For main metropolitan areas, two ensembles will be allocated to offer a total of twelve DAB services in addition to the two national radio ensembles.

In September of 1995, the BBC introduced DAB services to the United Kingdom. Currently they are operating five transmitters that cover 20% of the UK population. They are broadcasting their national service, Radio 1, 2, 3, 4, and 5. If plans progress as expected, DAB services should be available to 60% of the population by the early part of 1998 with the use of 27 transmitters. Additional information concerning the DAB developments in UK can be found at URL http://www.magi.com/~moted/dr/uk.

2.2.2 Germany

Germany is conducting a series of pilot projects that will introduce DAB. They are being conducted in Bavaria, Baden-Württemberg, North Rhine-Westphalia, Saxony, and Berlin. VHF channel 12 and L-band frequencies are being used to conduct these tests. In addition, over 10,000 receivers will available to those who are interested in participating in these tests. Germany has been the first to launch such tests where the consumer is consulted on DAB program services.

2.2.3 France

France is conducting DAB tests in Rennes and Paris. These are being conducted in Band I, 47-68 MHz, and L-Band. Radio France has established a Working Party on DAB. The Working Party will determine which programs are mostly likely suited for DAB transmission. In addition, France's main network provider has agreed with Radio France to provide DAB service to all of the major metropolitan areas and motorways as soon as receivers become available.

Ten programs are being broadcast in Paris using two transmitters. The majority of the programs offered have analog host stations in the FM band. The remaining two are new music programs that have been specially created for the DAB service. Further information can be found at URL http://www.kp.dlr.de/DAB/france1.htm.

2.2.4 Netherlands

The Netherlands are conducting a series of SFN tests. There are four transmitters operating at Channel 12 covering 40% of the population. These tests are administered by NOZEMA, the public and private network provider for the Netherlands. They are taking place in Haarlem, Hilversum and Rotterdam. In addition, the DAB system is utilizing very low coding rates in order to increase spectrum efficiency. Furthermore, various multimedia companies are expressing interest in providing additional services besides sound programs.

An extensive description of DAB developments in the Netherlands as well as DAB tests conducted in 1994 can be found at URL http://www.wp.com/hansbakhuizen/1_2dab.html. The Radio-L news group also maintains a URL http://www.magi.com/~moted/dr/nl concerning DAB developments in the Netherlands.

2.2.5 Other European Developments

There are various other DAB developments in Belgium, Denmark, Finland, Hungary, Italy, Norway, Poland, Sweden and Switzerland. Most countries have begun implementation of a DAB network by installing the first DAB transmitters. Some have conducted tests and have started or have plans to start a DAB service. For example, Sweden has installed six transmitters. They are in the process of providing a DAB service to 33% of their population. this is being done with the use of three SFNs. Additional information can be found by consulting the Radio-L web page.

2.3 Canada

The first demonstration of Eureka 147 was conducted in 1990. The Canadian government set up the Task Force on the Introduction of Digital Radio in 1992. In 1993, Digital Radio Research Inc. (DRRI) was formed by public and private broadcasters to own and operate experimental transmitters. After some extensive study, in August of 1995, Canada adopted the Eureka 147 DAB system as the Canadian system standard for digital radio services in part of the L-Band, 1452-1492 MHz.

Canada sees DAB as the only form of radio broadcasting suitable for the future. They feel that sometime in the next century, DAB will eventually replace any need for AM or FM broadcasting. Canada has been a strong developer of DAB. After conducting extensive research, Canada announced its necessity for a fourth mode in the original Eureka 147 DAB system. After some evaluation of Canada's proposal and the required changes it would bring to the standard, the Eureka 147 consortium agreed to the addition of this fourth mode. This a fairly new development and limited information is available on it. However, Canada's web site, listed at the beginning of this chapter, is fairly extensive and contains links to various other sites where information concerning DAB developments in Canada can be found.

2.4 United States

Currently Eureka 147 is undergoing evaluation by the Federal Communications Commission

(FCC) as a possible DAB standard for the United States. It is being evaluated along with some possible in-band solutions. In August 1995, the results from lab tests conducted on various DAB proposals for the U.S. were released. The lab tests were designed to test how well all the systems perform under extreme broadcasting conditions. The lab results indicated that Eureka 147 was able to perform under these extreme conditions. We must note that this was a quantitative analysis and no qualitative conclusions were drawn from the results. The organizations that administered these tests were not even made aware of the identity of the systems during the testing process.

Later this year, field tests will be conducted in the San Francisco bay area. This area was chosen for is varying terrain configuration as well as its distribution of population. Proponents of DAB are hopeful that the FCC will be able to adopt a standard for DAB based on this field test, and the previous year's lab results.

In addition to these tests that are being conducted, the broadcasting industry's opinion is also being taken into consideration by the FCC. For the most part, the broadcasters of this nation are in favor of an in-band solution. They feel that an in-band solution would offer them many more advantages than an out of band solution. The most important being that the in-band solution would allow them to keep their existing audiences by transmitting a digital signal for their station over their same coverage area. This is definitely an advantage for broadcasters who have invested heavily in their stations. However, it is not clear whether the in-band solutions can in fact provide the services that they claim. The results of this is still yet to be seen in the San Francisco field trials.

Chapter 3

Technicalities of Eureka 147

Background

This chapter is provided as a summary of the European Telecommunication Standard 300 401-Radio Broadcasting Systems; Digital Audio Broadcasting (DAB) to Mobile, Portable and Fixed Receivers. It is intended to give a brief description of the technical details surrounding the Eureka 147 DAB system. This chapter addresses the construction of the transmitting signal. Consequently, it is an overview of the services that can be provided by Eureka 147, their arrangement and placement within the signal, and the details concerning the structure of the final transmitted signal. Further clarification can be found in the references made throughout the chapter to the standard.

3.1 Introduction

The Eureka 147 Digital Audio Broadcasting (DAB) system was developed by a European consortium to deliver high quality digital radio broadcasting to mobile, portable, and fixed receivers. It has been designed to deliver data services as well as sound programs. Eureka 147 is a flexible system that takes into account the needs and availability of resources to any one particular service area. There are a variety of configuration and implementation techniques that are being explored and tested. The system has the capability to effectively be used as a pure terrestrial broadcasting system, a satellite broadcasting system with possible additional terrestrial gap fillers, a mixed terrestrial/satellite broadcasting system, or as a cable distribution system. The system also has the capability to operate at frequencies up to 3000 MHz. The most commonly used form of implementation to date is terrestrial based broadcasting.

The system has been developed to combat the effects of multipath that can traditionally be destructive to a received signal. The Eureka 147 system can take advantage of multipath echoes to contribute positively to the received signal. The system uses advanced coding techniques to protect the transmitted signal from errors and uses the advantages of spreading the signal in time and frequency to aid in making the signal robust against interference.

The system exercises the concept of Single Frequency Networks (SFN) to aid in efficient spectrum utilization. The signal is an interleaved frame of multiple programs that can be transmitted throughout a region on multiple transmitters all on the same frequency. This concept, although new to the United States, is a way of extending radio broadcasting networks without any limits.

The construction of the main signal is complex and knowledge of its construction is helpful in determining optimal solutions for implementation techniques. Figure 3.1 is a block diagram of the main signal construction. Typically, Eureka 147 uses multiple programs and/or data services to construct one ensemble that is transmitted in a particular channel frequency. A DAB frequency band will have multiple channels each with multiple programs. Each channel is 1.536 MHz wide. The receiver will be designed to decode each channel signal and then provide information as to what ensemble of stations or programs are available in each channel. Figure 3.1 illustrates how the programs are interleaved and multiplexed together to construct the main signal.

The information is transmitted in a frame structure which is made up of three groups of Orthogonal Frequency Division Multiplex (OFDM) symbols. The first group of OFDM symbols, labeled 1 in Figure 3.1, is called the Synchronization Channel. This part of the frame contains information that the receiver uses to initially synchronize on the transmitted signal. The second group of OFDM symbols, labeled 2 in Figure 3.1, is called the Fast Information Channel (FIC). The FIC is used to transport data that is not time interleaved and must be decoded immediately by the receiver. The third group of OFDM symbols, labeled 3 in Figure 3.1, is called the Main Service Channel (MSC). The MSC, among other things, contains the audio programs and is time interleaved to protect against channel distortion. A more detailed explanation of each of these channels and their corresponding contents is provided later in this chapter.

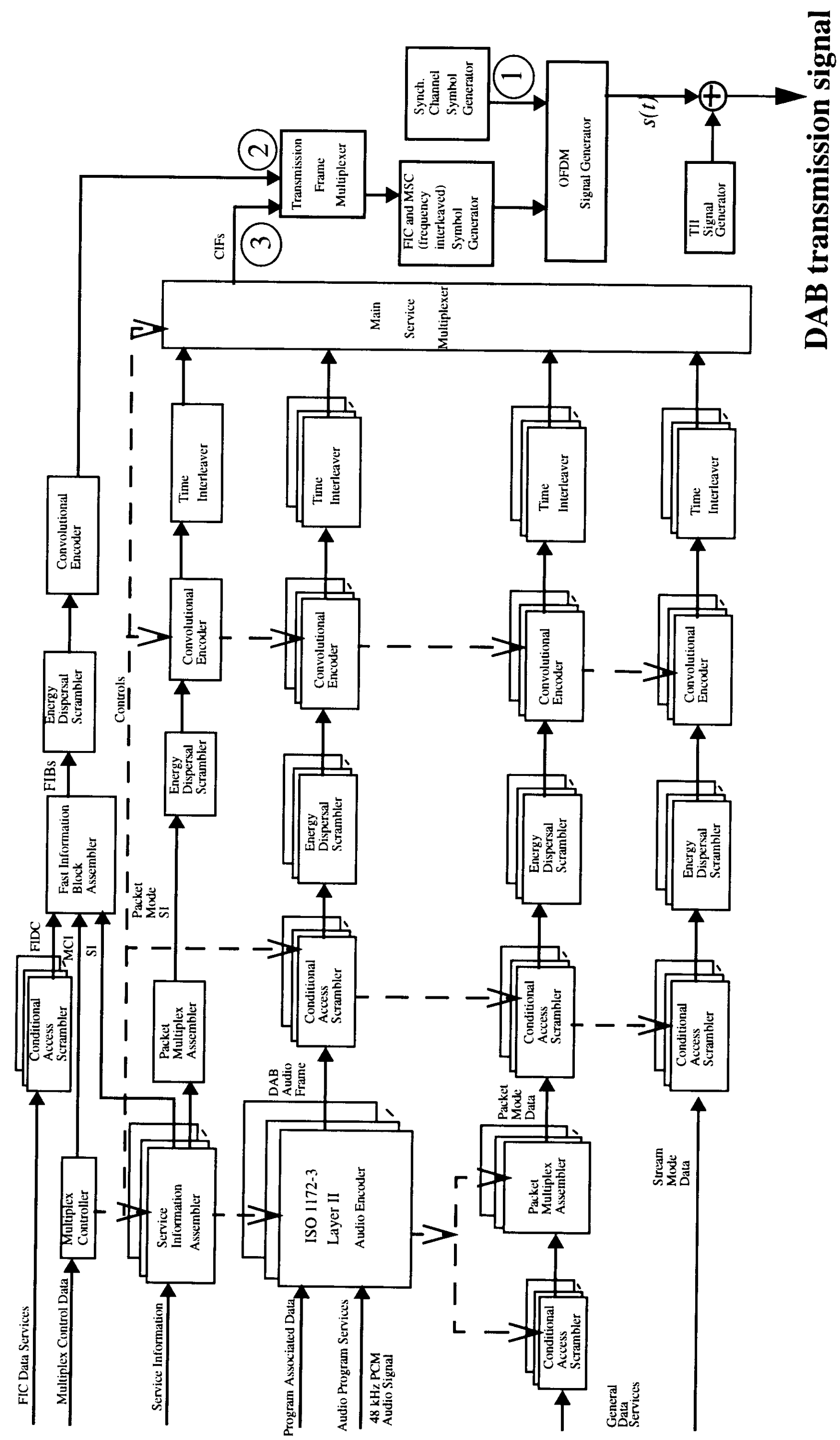

Figure 3.1: Conceptual DAB emission block diagram [17, p. 22]

Along with transmitted audio programs, Eureka 147 provides a medium to transmit other data services. Such data services includes Service Information (SI), Program Associated Data (PAD), Independent Data Services and Conditional Access (CA).

Service Information (SI)

SI data is used by the receiver to provide information about the ensemble. SI may also contain information about additional services that may be available in other ensembles as well as possible information about the AM/FM host stations that may be transmitting at additional frequencies. Specifically, the SI includes announcements, service component language, service linking, time and country identifier, program number, program type, service trigger, frequency information, information about other ensembles, FM services, transmitter identification, regional identification, local service area, ensemble label and service label.

Generally, the SI service is not time interleaved. The receiver must be able to decode the SI immediately to provide accurate information about the ensemble. Occasionally, when the SI information cannot be accommodated completely in the FIC, it is transported in the MSC under a heading of Auxiliary Information Channel (AIC). As a consequence, the AIC SI is not decoded as rapidly as the FIC SI. The coding of the SI will be discussed later when the coding of the FIC and AIC are addressed. In addition, subclause 8.1 of ETS 300 401 gives a more detailed explanation of the SI coding procedure.

Program Associated Data (PAD)

The PAD service is used to transmit information about programs within an ensemble. An example of this information may be the title and author of the audio program. The specific functions of the PAD data are dynamic range control, music/speech indication, command channel, product and article codes, program related text, and in-house information. The dynamic range control is used to adjust the receivers listening range. This allows the transmitter to indicate to the receiver about a possible noisy environment. The music/speech indication is used indicate if music or speech is being communicated. The command channel feature is used to dictate commands to the receiver's decoder. The product and article code feature is used to convey codes that may have accompanied the transmitted program. Specifically, these may be the Universal Product Code or European Article Number. The program related text is used to transmit text that is associated with the current program. This may be slogans for commercials or lyrics to a song. This feature may require more bit allocation than is provided by predetermined allowable PAD data length. Additional space can be allocated if necessary. The in-house information service is used within the broadcast chain and may contain commands relevant to the data transmitted.

The PAD information is embedded within the audio bit stream of the program transmitted in the MSC. This information is only relevant to the data that being transmitted in the next frame and is allocated two bytes at the end of the audio stream. These two bytes are called the Fixed Program Associated Data (F-PAD). Any additional required space is referred to as the X-PAD field. The amount of PAD information that can be embedded in the audio stream depends on the data rates of the program and how much the audio stream is willing to compromise on the quality of its program. Specifically, the F-PAD is located at the end of a DAB audio stream. Before these last two bytes, a Scale Factor Cyclic Redundancy Check (ScF-CRC) code is placed. The X-PAD region continues from the end of the audio frame to the beginning of the this CRC code. A maximum of four bytes of the X-PAD and the two bytes of the F-PAD have the same protection level as the ScF-CRC code. If the X-PAD extends beyond that, it has a lower protection level. Use of the PAD service is optional. If there is no PAD then the last two bytes of the audio frame are set to zero and consequently, it is assumed that there is no X-PAD service as well. Subclause 7.4 of the ETS 300 401 standard outlines a more detailed explanation of the coding procedure within the F-PAD and X-PAD.

Independent Data Service

General data services can be transmitted without the use of the PAD and SI mechanisms. When transmitted in the MSC, this data can be either transported in stream or packet mode. Stream mode data is segmented in 24 ms logical frames with data rates of n x 8 kbits/sec. Packetized data may have much lower capacities and are bundled in packet sub-multiplex. Coding procedures for independent data services are similar to the ones applied to audio services, PAD or additional SI.

When transmitted in the FIC, the independent data services are already defined. They are the Traffic Message Channel, Paging and Emergency Warning System. These services are specifically transported in what is called the Fast Information Data Channel (FIDC). The FIDC is defined by a special coding of the FIC and carries fast access information for the transmission frame. Detailed explanation of the coding and specifications of these services within the FIDC is given in subclause 8.2 of ETS 300 401.

Conditional Access (CA)

The Conditional Access data service is a system that places conditional access on the data services described above. It has three main functions: scrambling/descrambling, entitlement checking and entitlement management. The scrambling/descrambling function limits which users have access to the transmitted information. This function scrambles the proprietary service with a Control Word (CW). Multiple services can be scrambled with one CW or multiple CWs. The receiver is able to descramble the data if it is holding the appropriate CW. The entitlement checking function transmits the codes that are necessary to decode the message. The function checks to see if the necessary conditions required to access the service are available. The codes to decrypt the message are transmitted within the ensemble. The last function, entitlement management, is used to distribute entitlements to the receivers. This allows a receiver to decode a subscribed service. The services can be programs that a user may have pre-payed for or services that he or she wants to request on demand. These may be similar to the pay-per-view services that are offered by the cable industry.

This services will be transmitted using the independent data services structure.

Each service component has an indicator that determines whether the service component has Conditional Access or not. The service can be transmitted unscrambled, scrambled with a CW that is permanently coded into the receiver, or scrambled with the CW also transmitted within the service component. The messages that are flagged with an unscrambled identifier, are free to be accessed by all receivers and no subscription is needed. Services requiring a CW, are free to be accessed by receivers holding the CW. In addition, the receiver does not have to subscribe to the service. Both these modes are referred to as a "free access mode." Those services that have varying CWs are said to be in "controlled access mode." These services use the entitlement checking function to verify if a receiver has the necessary conditions to be able to decode the component. The CW is encrypted and transmitted along with the service. Details of the scrambling techniques used on the services can be found in subclause 9.1 of ETS 300 401.

When a service is in control access mode, it initially uses Electronic Checking Messages (ECM) to verify that the receiver has the necessary conditions to descramble the service. Then Entitlement Management Messages (EMM) are sent to transport new entitlements and management data to customers. These messages may be transported within the scrambled service or separately. Subclause 9.3 of ETS 300 401 gives the details of the ECM and EMM coding and configuration. Subclause 9.2 gives the details of the signaling and synchronization for the Conditional Access mechanism. Clause 9 of ETS 300 401 is dedicated to the details of providing Conditional Access to data service components.

3.2 Frame Construction

The DAB signal consists of transmission frames that contain the audio program and service information. Each transmission frame consists of three sets of Orthogonal Frequency Division Multiplex (OFDM) symbols. The transmission frame structure can be seen in Figure 3.2.

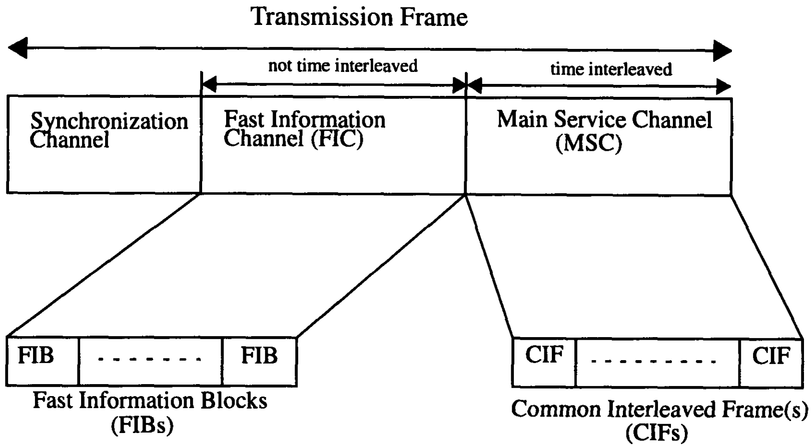

Figure 3.2: Transmission mode independent description of the FIC and MSC [17, p.30]

The three sets of the OFDM Symbols are called the Synchronization Channel, Fast Information Channel (FIC) and the Main Service Channel (MSC). The FIC is composed of Fast Information Blocks (FIBs), and the MSC is composed of Common Interleaved Frames(CIFs). The transmission frame contains the ensemble of data to be transmitted in a particular channel frequency.

An ensemble may have one or more services. There may be one or several service components within a service. These components may be audio, data, traffic messages, and other data services. Several services may also share service components. Each of these service components are either transmitted in a sub-channel within the MSC or in the FIC.

The organization of these sub-channels within the MSC is defined by the Multiplex Configuration Information (MCI). The MCI is transmitted in the FIC. The MCI serves five functions for the DAB ensemble. It defines the organization of the sub-channels in terms of their position and size in the CIF and their error protection, lists the services available in the ensemble, establishes the links between service and service components and signals a multiplex re-configuration if needed. Details of the MCI can be found in subclauses 6.1, 6.2 and 6.3 of ETS 300 401.

Eureka 147 has four modes of operation. They are labeled Transmission Mode I, II, III and IV. The fourth mode of operation is under development. Hence, specifications and details of the fourth mode were not available to be listed here. The transmission modes determine the transmission parameters such as frame length and symbol duration. These transmission modes allow the system to operate from 30MHz to 3 GHz. The parameters are determined to insure that the signal does not suffer from the effects of Doppler shift or delay spread. The signal is divided into smaller segments that are then modulated using π/4–Differential Quadrature Phase Shift Keying (D-QPSK). This spreads the signal and allows the signal to be free from errors that can be caused by delay spread. Intersymbol interference is avoided by the use of a guard interval for each symbol, whose length is determined by the transmission mode.

Each frame has total duration of 96 ms for Transmission Mode I and 24 ms for Transmission Mode II and III. Table 3.1 lists the frame length and the number of FIBs and CIFs per transmission frame. Table 3.2 lists the frame parameters for each of the transmission modes.

Transmission Mode	Duration of transmission frame	Number of FIBs per transmission frame	Number of CIFs per transmission frame
I	96 ms	12	4
II	24ms	3	1
III	24ms	4	1

Table 3.1: General transport characteristics of the transport frame [17, p. 30]

Parameter	Transmission Mode I	Transmission Mode II	Transmission Mode III
L	76	76	153
K	1,536	384	192
T_F	196,608 T or 96 ms	49,152T or 24 ms	49,152T or 24 ms
T_{NULL}	2,656T ~ 1.297ms	664T ~ 324µ s	345T ~ 168µ s
T_S	2,552T ~ 1.246 ms	638T ~ 312µ s	319T ~ 156µ s
T_U	2,048T or 1ms	512T or 250µ s	256T or 125µ s
Δ	504T ~ 246µ s	126T ~ 62µ s	63T ~ 31µ s

Table 3.2: Definition of the parameters for transmission modes I, II, III [17, p. 160]

The period T = 1/2048000 seconds. The various parameters are defined as follows.

L is the number of OFDM symbols per transmission frame excluding the Null Symbol
K is the number of transmitted carriers
T_F is the transmission frame duration
T_{NULL} is the Null symbol duration
T_S is the duration of OFDM symbols of indices l = 1, 2, 3,..., L;
T_U is the inverse of the carrier spacing
Δ is the duration of the time interval called guard interval;

The signal is defined by the following formula:

$$s(t) = Re\left\{e^{2j\pi f_c t} \sum_{m=-\infty}^{\infty} \sum_{l=0}^{L} \sum_{k=-K/2}^{K/2} z_{m,k,l} \cdot g_{k,l}(t - mT_F - T_{NULL} - (l-t)T_S)\right\}$$

with,

$$g_{k,l}(t) = \begin{cases} 0 & for\ l = 0 \\ e^{2l\pi kt - \Delta/T_U} \cdot Rect(t/T_S) & for\ l = 1, 2, \ldots, L \end{cases}$$

and $T_S = T_U + \Delta$.

Where

$z_{m,l,k}$ is the complex D-PQSK symbol associated with carrier k of OFDM symbol l during transmission frame m. For $k = 0$, $z_{m,l,k} = 0$, so that the central carrier is not transmitted.

f_c is the central frequency of the signal. The possible values of f_c are given in clause 15 of ETS 300 401.

As illustrated in Figure 3.1, there are multiple blocks following the Main Service Multiplex that generate the main signal, $s(t)$. A more detailed explanation of the Transmission frame multiplexer, FIC and MSC symbol generator, synchronization channel symbol generator and OFDM signal generator is shown in Figure 3.3. Once the symbols for the Synchronization, Fast Information and Main Service Channel are constructed, they are multiplexed together as shown in Figure 3.3 in the OFDM Signal Generator. Then the symbols, $z_{(m),l,k}$, are used to construct the main signal, $s(t)$, using the above formula in the OFDM symbol generator.

3.3 Synchronization Channel

The Synchronization Channel contains information to perform frame synchronization, auto frequency control, channel state estimation, and transmitter identification. It contains the first two OFDM symbols of the frame. The first symbol is always the Null symbol. Its duration is determined by T_{NULL}. The second symbol in the Synchronization Channel is called the Phase Reference Symbol.

Null Symbol

The first symbol of each frame is called the Null Symbol. The signal $s(t) = 0$ for $t = 0...T_{NULL}$. The null symbol generator is located in the synchronization channel symbol generator. It is illustrated in Figure 3.3.

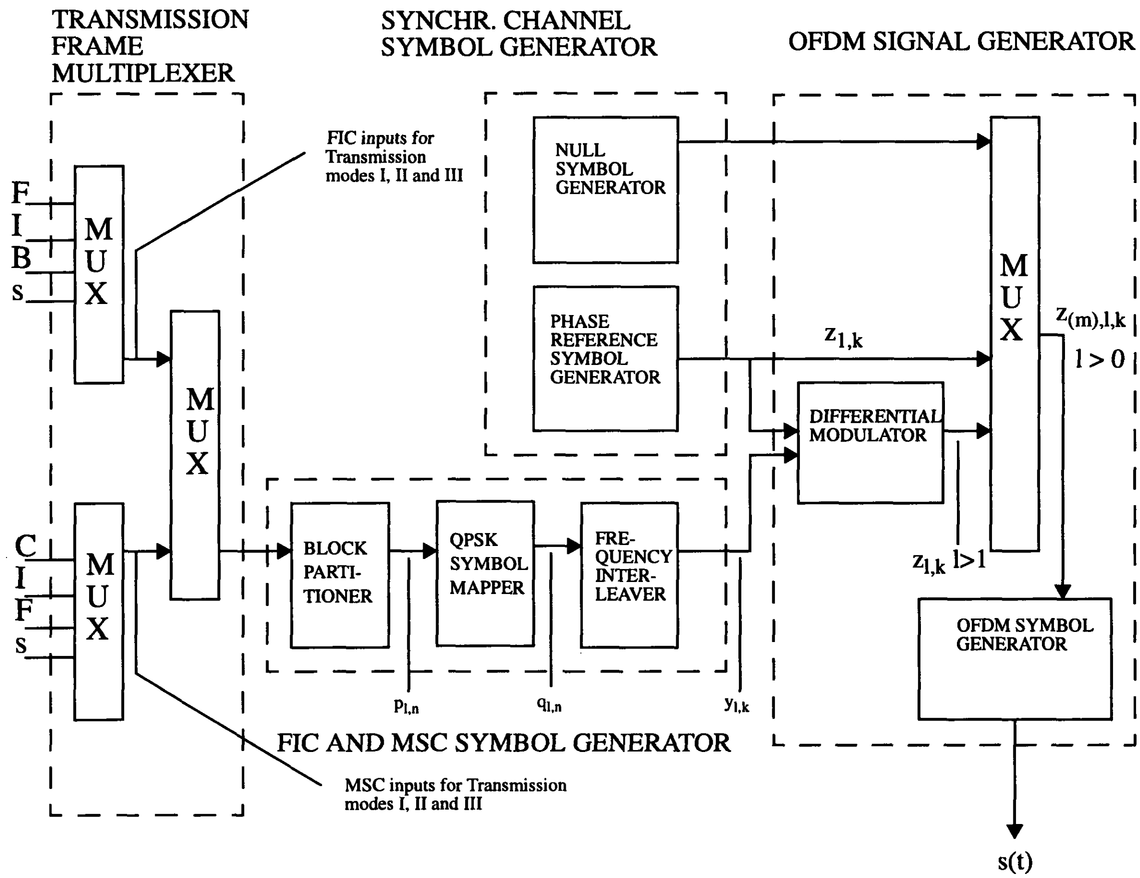

Figure 3.3: Generation of the main signal [17, p. 161]

Phase Reference Symbol

This is the second OFDM symbol for the frame. It forms the reference for the next OFDM symbol transmitted in the FIC. Due to differential modulation, a symbol uses the previous symbol to obtain its phase reference. The phase reference symbol generator is also located in the synchronization channel symbol generator as shown in Figure 3.3. The phase reference symbol has the following form.

$$z_{1,k} = \begin{cases} e^{j\varphi_k} & for \quad -\frac{K}{2} \le k < 0 \quad and \quad 0 \le k \le \frac{K}{2} \\ 0 & for \quad k = 0 \end{cases}$$

where

$$\varphi_k = \frac{\pi}{2}(h_{i,k-k'} + n)$$

The parameters i, k', n, and $h_{i,j}$ have been specified by the following tables. Table 3.3 is for transmission mode I, Table 3.4 is for transmission mode 2, and Table 3.5 is for transmission mode III. Table 3.6 shows the values for $h_{i,j}$. These values are constant for all three modes of operation.

k in the range of		*k'*	*i*	*n*
min	max			
-768	-737	-768	0	1
-736	-705	-736	1	2
-704	-673	-704	2	0
-672	-641	-672	3	1
-640	-609	-640	0	3
-608	-577	-608	1	2
-576	-545	-576	2	2
-544	-513	-544	3	3
-512	-481	-512	0	2
-480	-449	-480	1	1
-448	-417	-448	2	2
-416	-385	-416	3	3
-384	-353	-384	0	1
-352	-321	-352	1	2
-320	-289	-320	2	3
-288	-257	-288	3	3
-256	-225	-256	0	2
-224	-193	-224	1	2
-192	-161	-192	2	2
-160	-129	-160	3	1
-128	-97	-128	0	1
-96	-65	-96	1	3
-64	-33	-64	2	1
-32	-1	-32	3	2
1	32	1	0	3
33	64	33	3	1
65	96	65	2	1
97	128	97	1	1
129	160	129	0	2
161	192	161	3	2
193	224	193	2	1
225	256	225	1	0
257	288	257	0	2
289	320	289	3	2
321	352	321	2	3
353	384	353	1	3
385	416	385	0	0
417	448	417	3	2
449	480	449	2	1
481	512	481	1	3
513	544	513	0	3
545	576	545	3	3
577	608	577	2	3
609	640	609	1	0
641	672	641	0	3
673	704	673	3	0
705	736	705	2	1
737	768	737	1	1

Table 3.3: Relation between the indices *i*, *k'* and *n* and the carrier index *k* for transmission mode I [17, p. 163]

k in the range of		*k'*	*i*	*n*
min	max			
-192	-161	-192	0	2
-160	-129	-160	1	3
-128	-97	-128	2	2
-96	-65	-96	3	2
-64	-33	-64	0	1
-32	-1	-32	1	2

k in the range of		*k'*	*i*	*n*
min	max			
1	32	1	2	0
33	64	33	1	2
65	96	65	0	2
97	128	97	3	1
129	160	129	2	0
161	192	161	1	3

Table 3.4: Relation between the indices *i, k'* and *n* and the carrier index *k* for transmission mode II [17, p.163]

k in the range of		*k'*	*i*	*n*
min	max			
-96	-65	-96	0	2
-64	-33	-64	1	3
-32	-1	-32	2	0

k in the range of		*k'*	*i*	*n*
min	max			
1	32	1	3	2
33	64	33	2	2
65	96	65	1	2

Table 3.5: Relation between the indices *i, k'* and *n* and the carrier index *k* for transmission mode III [17,p. 163]

j	0	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15
$h_{0,j}$	0	2	0	0	0	0	1	1	2	0	0	0	2	2	1	1
$h_{1,j}$	0	3	2	3	0	1	3	0	2	1	2	3	2	3	3	0
$h_{2,j}$	0	0	0	2	0	2	1	3	2	2	0	2	2	0	1	3
$h_{3,j}$	0	1	2	1	0	3	3	2	2	3	2	1	2	1	3	2

j	16	17	18	19	20	21	22	23	24	25	26	27	28	29	30	31
$h_{0,j}$	0	2	0	0	0	0	1	1	2	0	0	0	2	2	1	1
$h_{1,j}$	0	3	2	3	0	1	3	0	2	1	2	3	2	3	3	0
$h_{2,j}$	0	0	0	2	0	2	1	3	2	2	0	2	2	0	1	3
$h_{3,j}$	0	1	2	1	0	3	3	2	2	3	2	1	2	1	3	2

Table 3.6: Time-Frequency-Phase parameter *h* values [17, p. 164]

The above information pertaining to the Synchronization Channel can also be found in ETS 300 401, subclause 14.3.

3.4 Fast Information Channel

The second set of OFDM symbols that are transmitted in a frame are part of the Fast Information Channel (FIC). The FIC is made up of Fast Information Blocks (FIBs). Before continuing on to the FIC symbol generation and OFDM signal generation due to the FIBs, the construction of the FIBs will be discussed.

Fast Information Block (FIB)

The number of FIBs in an FIC is determined by the transmission mode and can be found in Table 3.1. The FIB, which is made up of 256 bits, is further subdivided into Fast Information Groups (FIG), an end marker, padding and a cyclic redundancy check (CRC). The structure is shown in Figure 3.4. The CRC word is generated based on the polynomial $G(x) = x^{16} + x^{12} + x^5 + 1$. The procedure for generating the CRC can be found in ETS 300 401 Annex E.

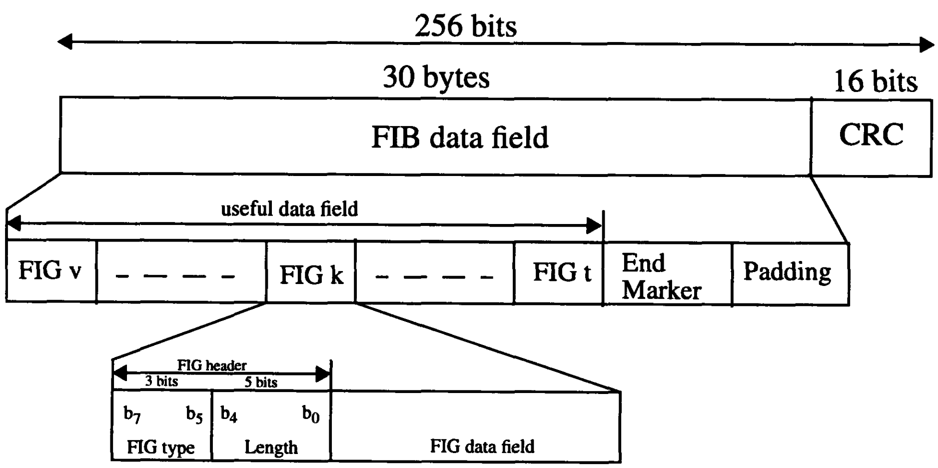

Figure 3.4: Structure of the FIB [17, p. 31]

The end marker is composed of eight bits that are all ones. It is considered a special FIG. The padding is a string of zeros that will complete the 30 bytes of the FIB data field. In the beginning of each FIG there are eight bits that designate its type and length. Three bits are used to designate the FIG type and five bits are used to designate its length. The length ranges from 1-29 but values 0, 30, and 31 have been reserved for future use. However, FIG length 31 (11111) when used with FIG type 7 (111) is designated as the end marker with no FIG data field. The remaining bits in a FIG are the FIG data field. The FIG structure is also illustrated in Figure 3.4.

The FIB adheres to this structure unless the FIB useful data field is 30 bytes long. In this case there is no end marker or padding provided for the field. In the case when the useful data field is 29 bytes long, the padding but not the end marker is eliminated. Table 3.7 lists the different FIG types that are defined in the ETS 300 401 standard. Those that are marked Reserved, will be defined in the future when additional uses for the FIG are determined.

FIG Type Number	FIG Type	FIG Application
0	000	Multiplex Configuration Information (MCI) and part of the SI
1	001	Labels, etc. (part of the SI)
2	010	Reserved
3	011	Reserved
4	100	Reserved
5	101	FIC Data Channel
6	110	Conditional Access
7	111	In house (expect for Length 31)

Table 3.7: List of FIG Types [17, p. 32]

FIG types 0, 1, 5, 6 are discussed below.

FIG Type 0 -- Multiplex Configuration Information(MCI) and part of the Service Information (SI)

The FIG type 0 is used carry the Multiplex Configuration Information and some basic services for the Service Information mechanism. The FIG type 0 data field is shown in Figure 3.5.

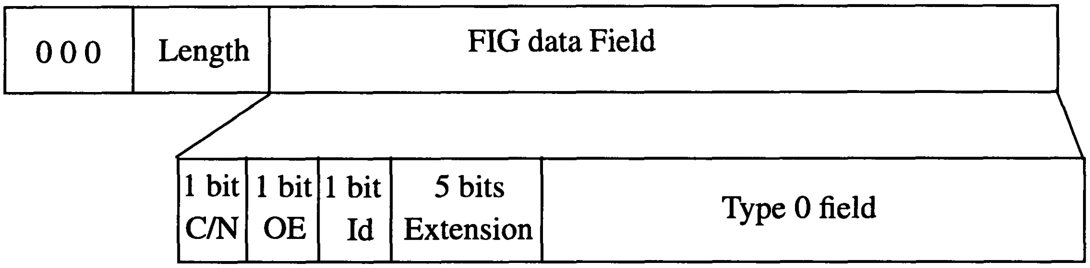

Figure 3.5: Structure of the FIG type 0 data field [17, p. 33]

The C/N bit is used to determine whether the information being transmitted is for the current multiplex configuration or the next multiplex configuration. A 0 is used to convey data for the current configuration and a 1 is used to convey data for the next configuration. The OE bit is used to indicate if the data is related to this ensemble or another ensemble. A 0 is used for this ensemble and a 1 is used for another ensemble. The Id bit is used to convey wether the Service Identifiers are in 16 or 32 bit format. A 0 is used to indicate a 16 bit format and a 1 is used to indicate a 32 bit format. The 5 bit extension field is used to indicate which extension of the FIG type 0 is being used. There are 32 extensions possible, however, not all have been defined. The following is a table for those extensions that are defined. Also listed are the relevant sections in the ETS 300 401 standard that discuss the FIG type 0 extension coding in detail.

Extension	Function	Definition	Section of ETS 300 401 with further clarification
0	Ensemble Information	contains SI and control mechanisms common to all services in ensemble	Subclause 6.4
1	Sub-Channel Organization	defines the position and size of the sub-channels in the Common Interleaved Frame and the error protection employed	Subclause 6.2

Table 3.8: Extensions to FIG Type 0 data field

Extension	Function	Definition	Section of ETS 300 401 with further clarification
2	Basic service and service component definition	defines the basic service organization	Subclause 6.3.1
3	Service component in Packet mode with or without CA	gives additional information about the service component description in Packet Mode	Subclause 6.3.2
4	Service component with CA in stream mode or FIC	gives additional information about the service component in Stream mode or in the FIC	Subclause 6.3.3
5	Service Component Language	defines the language of the service component	Subclause 8.1.2
6	Service Linking Information	provides service linking information for use when services carry the same program	Subclause 8.1.15
10	Time and Country Identifier	defines date, time, local time offset, extended country code and the International Table	Subclause 8.1.3
11	Region Definition	defines geographical area	Subclause 8.1.16.1
12	Program Type Preview	provide a preview of Program Type codes of programs which are planned to be broadcast in the future	Subclause 8.1.5.3
16	Program Number	provide date and time at which a program begins	Subclause 8.1.4
17	Program Type Coding	specifies the Program Type (PTy) code and category	Subclause 8.1.5.1
18	Announcement Support	assigns, to a service, the types of announcements by which this service may be interrupted and links to other services which share the same interruption privileges.	Subclause 8.1.6.1
19	Announcement Switching	provides the dynamic signal to allow a vectored interruption of the service by another carrying announcement	Subclause 8.1.6.2

Table 3.8: Extensions to FIG Type 0 data field

Extension	Function	Definition	Section of ETS 300 401 with further clarification
20	Service Trigger	provides a general mechanism to signal that a new service will be emitted	Subclause 8.1.7
21	Frequency Information	provide lists of frequencies of part or full of current and other ensembles and FM services which can be received in and out of the coverage area of the ensemble.	Subclause 8.1.8
22	Transmitter Identification Information (TII)	provides a cross reference between transmitter identifiers and the geographic locations and relative time delays of the transmitters.	Subclause 8.1.9
23	Local Service Area	Identifies the Service Area	Subclause 8.1.17
24	Services (other ensembles)	used to identify the services carried in other DAB ensembles	Subclause 8.1.10.2
25	Announcement Support (other ensembles)	provides announcement support for other ensembles	Subclause 8.1.10.5.1
26	Announcement Switching (other ensembles)	provides announcement switching for other ensembles	Subclause 8.1.10.5.2
27	Announcement Support (FM services)	provides announcement support for FM services	Subclause 8.1.11.2.1
28	Announcement Switching (FM Services)	provides announcement switching for FM services	Subclause 8.1.11.2.2
29	Satellite Handover	Information contains a control mechanism for the handover from the descending to ascending satellite broadcasting a DAB ensemble, provides the value of the CIF count at which the handover occurs	Subclause 8.2.18.2
30	Satellite Database	provides the information needed to assist service continuation at satellite handover or when switching to a satellite is performed	Subclause 8.1.18.1

Table 3.8: Extensions to FIG Type 0 data field

Extension	Function	Definition	Section of ETS 300 401 with further clarification
31	FIC Re-direction	used to signal which data features, coded in FIG Type 0 and 1 are carried in the Auxiliary Information Channel	Subclause 8.1.12

Table 3.8: Extensions to FIG Type 0 data field

FIG Type 1--Labels, etc. (part of the SI)

The FIG type 1 is used to signal labels for display and other information defining labels. Figure 3.6 shows the structure of the FIG Type 1 field. The 4 bits of the Charset are used to identify a character set to qualify the character information contained in the FIG type 1 field. Currently this field is only defined for Latin, Cyrillic, Greek, Arabic and Hebrew character sets. The remaining codes will be defined in the future.

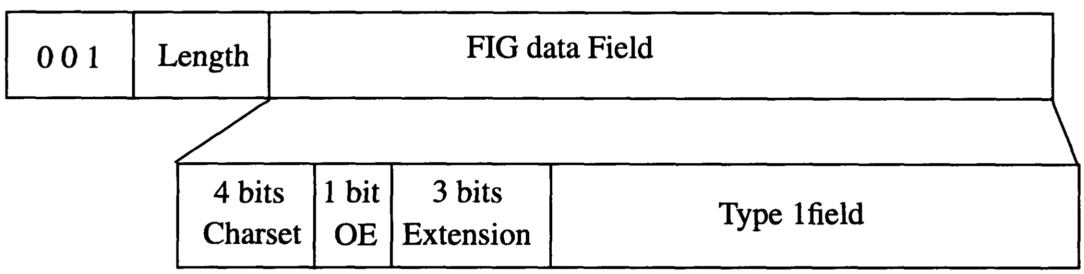

Figure 3.6: Structure of the FIG type 1 data field [17, p. 34]

The OE flag is used in the same fashion as FIG type 0 data field. A 0 is used to indicate that the information pertains to the current ensemble and a 1 is used to indicate that the information pertains to the next ensemble. The Extension field is used to designate between the different extensions available for FIG Type 1. The following table summarizes the extensions that are defined to date by the ETS 300 401 standard. Also listed are the relevant sections in the ETS 300 401 standard that discuss the FIG type 1 extension coding in detail.

Extension	Function	Definition	Section of ETS 300 401 with further clarification
0	Ensemble Label	defines the ensemble label	Subclause 8.1.13
1	Service Label	defines the service label	Subclause 8.1.14
2	Program Type Downloading	used to dynamically define the meaning of the PTy codes, used to establish link between different codes.	Subclause 8.1.5.2
3	Region Label	defines the region label	Subclause 8.1.16.2

Table 3.9: Extension for FIG Type 1 field

FIG Type 5--Fast Information Data Channel

FIG Type 5 is used to code the Fast Information Data Channel (FIDC). The FIDC is used to transport non-audio related data services. The following figure depicts the structure of the FIG Type 5 field. Coding of the D1 and D2 bits is determined by the Extension bits. The Type Component Identifier (TCId) is used to identify one of eight different service components which may be carried using the same extension number.

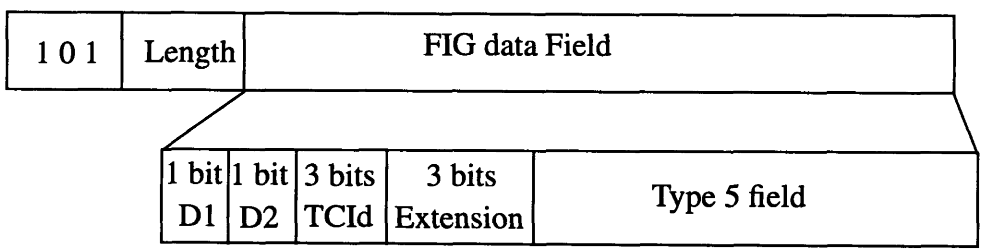

Figure 3.7: Structure of the FIG type 5 data field [17, p. 35]

There are eight different extensions possible with FIG type 5 data field. Only three extensions have been defined and are listed in Table 3.10. Other extensions are reserved for future use. Also listed are the relevant sections in the ETS 300 401 standard that discuss the FIG type 5 extension coding in detail.

Extension	Function	Definition	Section of ETS 300 401 with further clarification
0	Paging	pointer mechanism is used to indicate where the paging information maybe be carried in the MSC	Subclause 8.2.1
1	Traffic Message Channel (TMC)	provides the Traffic Message	Subclause 8.2.2
2	Emergency Waring Systems (EWS)	provides the Emergency Waring Message	Subclause 8.2.3

Table 3.10: Extension for FIG Type 5 field

FIG Type 6--Control and Management Information About a Scrambled Service Component

FIG Type 6 is used to control and manage information about a scrambled service. These services are CA messages. Specifically, Entitlement Checking Message (ECM) or Entitlement Management Messages (EMM) are sent in a FIG Type 6 field. The structure of FIG type 6 is depicted in the figure below. The first/last bits are used to indicate how the CA messages are managed if they have to be split into more than one FIG Type 6 field. A 00 is used to indicate that the FIG is an intermediate FIG of a series of FIGs. A 01 is used to indicate that the FIG is the last FIG of the series. A 10 is used to indicate that FIG is the first of a series. A 11 is used to indicate that there is one and only one FIG for the CA message.

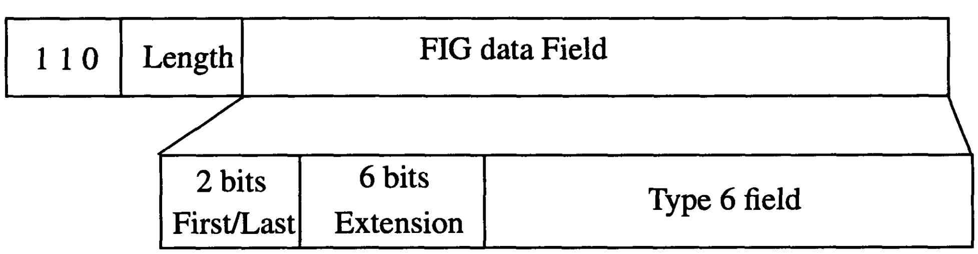

Figure 3.8: Structure of the FIG type 6 data field [17, p. 35]

The six bit extension field is used to identify one of 64 interpretations of the FIG type 6 field. An extension equal to the FIC ECM Identifier and not equal to zero implies that an ECM is being sent. An extension equal to 0 indicates that an EMM is being sent. See subclause 9.3 for details of the coding of EMM and ECM messages.

FIC Generation -- Energy Dispersal Scrambler

Once the FIBs have been configured, they are put through an "Energy Dispersal Scrambler." The signal is added modulo-2 to pseudo-random binary sequence (PRBS). This is generated by a feedback shift register with taps represented by the polynomial $P(X) = X^9 + X^5 + 1$. For transmission modes I and II, three FIBs at a time, each corresponding to a different CIF, are juxtaposed together to form a vector that is 768 bits long. This vector is then scrambled with the PRBS, with the first bit being scrambled with the PRBS bit of index 0. For transmission mode III, four FIBs are juxtaposed together to form vector of length 1024. This vector is then scrambled with the PRBS, with the first bit being scrambled with the PRBS bit of index 0. Clause 11 defines the details of the Energy Scrambler.

FIC Generation -- Convolutional Encoder

After the FIBs have been processed through the Energy Scrambler, they are convolutionally encoded. The channel encoding is based on punctured convolutional coding. This allows for both equal and unequal error protection. The convolutional code has a constraint length of 7. The definition of the mother code is given in subclause 11.1. Subclause 11.1.2 defines the puncturing procedure to be applied to the signal. For transmission mode I and II, the 768 bit output vector from the energy scrambler shall be convolutionally encoded as defined by subclause 11.1.1. This will generate a 3072 bit vector. These 3072 bits will then be punctured as defined in subclause 11.1.2 using a puncturing index of 16 and 15, which corresponds to a rate of approximately 1/3. The resulting vector has length 2304. For transmission mode III, the 1024 bit output vector from the energy scrambler shall be convolutionally encoded as defined in subclause 11.1.1 to produce a vector of length 4096. This vector will then be punctured using the procedure defined in subclause 11.1.2 with puncturing index 16 and 15. This also corresponds to a rate of approximately 1/3. The resulting vector has length 3072.

FIC Generation -- Block Partitioning

Once the FIBs have been convolutionally encoded, they are partitioned before being mapped to QPSK symbols. The details of the partitioning are provided in subclause 14.4.1. For transmission mode I, the four sets of convolutionally encoded vectors shall be multiplexed together to form a vector that is then partitioned into three blocks. These blocks will then be transmitted on three OFDM symbols. The four sets of convolutionally encoded vectors of length 2304 are juxtaposed together to form a vector of length 9216 and then partitioned into vectors of length 3072. These vectors will be OFDM symbols 2, 3 and 4. For transmission mode II, the convolutionally encoded vector of length 2304 shall be partitioned into three vectors of length 768. These vectors will be OFDM symbols 2, 3 and 4. For transmission mode III, the convolutionally encoded vector of length 3072, shall be partitioned into vector lengths of 384. This will produce 8 OFDM symbols numbered from 2 to 9. The resulting 2K-bit vector for each symbol from the block partitioner shall be referred to as $p_{l,n}\big|_{n=0}^{2K-1}$ for $l=2, 3, 4,\ldots,L$.

FIC Generation -- QPSK Symbol Mapper

All OFDM symbols shall be shall be mapped on the K complex QPSK symbols $q_{l,n}$ according to the following relation.

$$q_{l,n} = \frac{1}{\sqrt{2}}[(1-2p_{l,n}) + j(1-2p_{l,n+K}) \qquad for \qquad n = 0, 1, 2, \ldots, K-1.$$

Subclause 14.5 of ETS 300 401 discusses the details of the QPSK symbol mapper.

FIC Generation -- Frequency Interleaver

The QPSK symbols, $q_{l,n}$, are then frequency interleaved before proceeding to the OFDM Signal

Generator. Subclause 14.6 defines how the index n is mapped to the carrier index k ($-K/2 \le k < 0$ and $0 < k \le K/2$). Where $k=F(n)$ and $F(n)$ is defined for each transmission mode. The QPSK symbols shall be re-ordered according to the following relation:

$$y_{l,k} = q_{l,n} \qquad for \qquad l = 2, 3, 4, \ldots, L$$

Once the symbols have been frequency interleaved, they continue to the OFDM signal generator.

FIC Generation -- Differential Modulator

Part of the OFDM signal generator is the differential modulator. All symbols with the exception of the null symbol are differentially modulated before being multiplexed together and proceeding to the OFDM symbol generator. The differential modulation is defined by the following rule:

$$z_{l,k} = z_{l-1,k} \cdot y_{l,k} \qquad for \qquad l = 2, 3, 4, \ldots, L$$

and $-\frac{K}{2} \le k \le \frac{K}{2}$

Each of the symbols are modulated using a $\pi/4$-D-QPSK. Once all FIC symbols have been generated, they proceed to the OFDM symbol generator multiplex, where they are multiplexed with the null and phase reference symbols of the Synchronization channel, and the symbols for the Main Service Channel. The final signal, $s(t)$, is generated in the OFDM symbol generator.

3.5 Main Service Channel (MSC)

The MSC makes up the major portion of a DAB transmission frame. It carries audio and data services that may have many service components. The MSC is made up of Common Interleaved Frames (CIFs). Table 3.1, "General transport characteristics of the transport frame [17, p. 30]," on page 33 lists the number of CIFs in a transmission frame depending on the transmission mode. Each CIF is 55,296 bits long. CIFs are further subdivided into Capacity Units (CUs). Each CU is 64 bits long. An integral number of CUs are grouped together to constitute the basic transport unit of the MSC, called a sub-channel. As discussed earlier, each sub-channel carries service components of audio or data services that are carried in a transmission frame. The sub-channel organization is determined by the MCI, which is coded in the FIC.

Figure 3.1 shows how the MSC is constructed. Audio and PAD services are encoded before proceeding to the Conditional Access Scrambler. Only those services that are subjected to a CA restraint are scrambled. The signal is then processed through a energy dispersal scrambler, convolutional encoder and time interleaver before continuing on to the Main Service Multiplexer. General data services follow a similar procedure for packet and stream mode data. Additional SI can also be coded in the MSC. It is coded in sub-channel 63 which is known as the Auxiliary Information Channel (AIC). Once all the services have been multiplexed, they continue on to the transmission frame multiplexer, MSC symbol generator and finally the OFDM signal generator, where they follow a similar procedure as the one described for the FIC. The final signal is multiplexed together with the Synchronization Channel and FIC to complete the final signal, *s(t)*.

MSC Generation -- Audio Coding

Audio program services are first processed through an ISO 1172-3 Layer II Audio Encoder. A simplified block diagram of the encoder is shown in Figure 3.9.

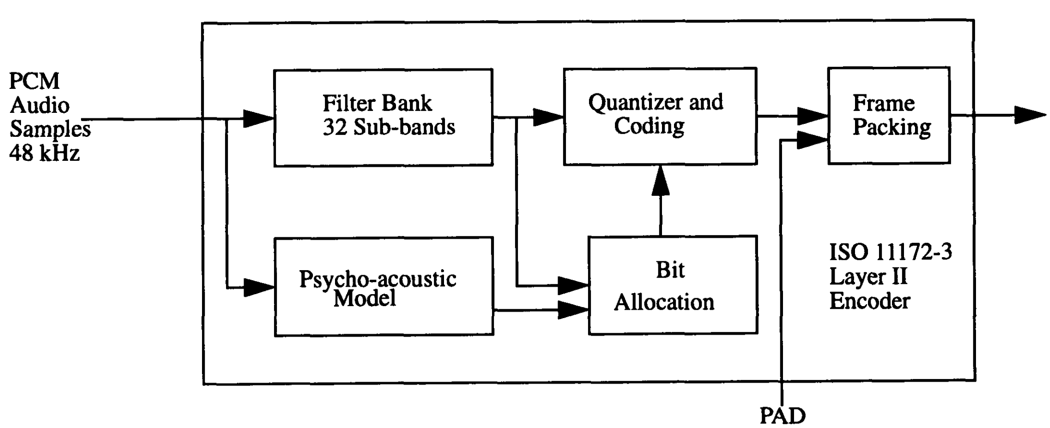

Figure 3.9: Simplified block diagram of the DAB Audio Encoder [17, p. 53]

The PCM audio samples are fed into the filter bank, which creates a signal that is a sampled version of the original signal. The filter bank splits the original signal into 32 equally spaced sub-bands. The sub-bands are also normalized by a scale factor. Details of the filter band and scale factor calculation can be found in subclause 7.1.1 to 7.1.4 and annex C, Clause C.1 of ETS 300 401.

The PCM audio samples are also fed into the Pyscho-acoustic model. This models the characteristics of human hearing and perception of sound to determine compressibility of the original signal. The information from the Psycho-acoustic model is used to determine the just noticeable noise level for each of the sub-band output samples from the filter bank. The output of the model is a Signal to Mask Ratio that is fed into the bit allocation block. Details of the psycho-acoustic model can be found in subclause 7.1.5 and Annex C, Clause C.2 of ETS 300 401.

The bit allocation block determines which of the sub-bands should not be transmitted using the signal to mask ratio and the 32 sub-band samples from the filter bank. It calculates the required bit allocation from the psycho-acoustic model and sub-band samples and finally produces a fixed bit rate to the quantizer and coding block. Details of the bit allocation procedure can be found in subclause 7.1.6 and annex C, clause C.3 of ETS 300 401.

The quantizer and coding block uses the fixed bit rate determined by the bit allocation procedure to limit the actual signal that will be transmitted from the 32 sub-band samples. These quantized samples will then be coded before continuing to the frame packing procedure. Details of the quantization and coding can be found in subclauses 7.1.7 and 7.1.8 of ETS 300 401.

The frame packing block assembles the audio bit stream from the quantizer with header information, CRC words for error detection and PAD. The resulting output is called an audio frame. Details of the frame packing can be found in subclause 7.1.9 of ETS 300 401. Details of the audio coding and variations thereof can be found in Clause 7. Coding of the PAD can also be found in subclause 7.4 of ETS 300 401.

MSC Generation -- General Data Services and Service Information

There are two different transport modes for the MSC general data services and SI. They are stream and packet mode. In stream mode, a service application is allowed to accept and deliver data transparently from source to destination. The data has a fixed rate which is a multiple of 8 kbits/sec. Only one service component is allowed to be carried on a sub-channel. Data is also transmitted at rates of multiples of 8 kbits/sec for packet mode transport. In addition, they have four fixed packet lengths: 24, 48, 72 or 96 bytes. Padding will be used in packets to ensure that the data rate is a multiple of 8 kbits/sec. Packets are identified by their address. Packets that have the same address must maintain their sequence in a sub-channel, and packets with different addresses can be transmitted in any order in a sub-channel. One or more packets with the same address shall be grouped together in a data group which will contain a data group header and data group CRC. The data group header, among other things, shall identify if the packetized data is general data and/or CA, or a file transfer. The Multiplex Configuration Information (MCI) will contain the links between a service component and packet address. The details of coding packetized data is found in subclause 5.3.2 to 5.3.4 of ETS 300 401. General data services shall be either transported in stream mode or packet mode.

SI that is redirected from the FIC using FIG type 0 extension 31, shall be transmitted in the MSC. The SI that is redirected is carried in the AIC. The AIC is formed using sub-channel 63 and packet address 1,023. However, although AIC uses a packet address, it will have a different construction than normal packetized data. FIGs, instead of packets, shall be transmitted in data groups. In addition, different FIG types can be transmitted within the same data group. The data group will still have a header and data group CRC. The maximum length of the data field shall be 512 bytes. Details of the transport of the SI in the AIC can be found in subclause 5.4 of ETS 300 401.

MSC Generation -- Conditional Access Scrambler

Once the audio bit stream has been encoded or the data services have been either coded in stream mode or packet mode, the signal continues on to a conditional access scrambler. With each service component, there is a conditional access flag which indicates whether or not the service component uses the CA mechanisms. If the flag is set to use the CA mechanisms, the audio and data are added modulo 2 to a PRBS and then scrambled based on the procedures outlined in subclause 9.1.4.2 and 9.1.4.3 or ETS 300 401. Clause 9 of ETS 300 401 discusses the details of the CA mechanism.

MSC Generation -- Energy Dispersal

The audio and data then proceed to an energy dispersal scrambler. This is similar to the procedure described in "FIC Generation -- Energy Dispersal Scrambler" on page 47 . The audio or data services shall be scrambled with a PRBS such that the first bit of each logical frame for a given sub-channel shall be added modulo 2 to the PRBS bit of index 0. The PRBS shall be defined by $P(X) = X^9 + X^5 + 1$.

MSC Generation -- Convolutional Encoder

The convolutional coding is similar to the one described for the FIC. The mother code is the same as the one applied to the FIC. Details of the mother code can be found in subclause 11.1.1. The resulting vector shall be applied to the puncturing procedure as described in subclause 11.3 for sub-channels conveying an audio service component and sub-channels conveying data service components. The details of the puncturing procedure are specified in terms of the protection profiles and protection levels. The average code rate can vary from 1/4 to 3/4 for data transmitted in the MSC. A code rate of 1/4 has the highest protection level because 3 bits are used to code 1 source bit of information.

MSC Generation -- Time Interleaving

The output of the convolutional encoder shall be time interleaved before continuing onto the Main Service Multiplexer as shown in Figure 3.1. The procedure for time interleaving is defined in Clause 12 of ETS 300 401. The time interleaving rule takes into account the bit rate and protection level of the transmitted signal. The interleaver produces a bit stream that is equal to the input bit stream in length. In the case of a multiplex re-configuration, the output bit stream may be longer than the input bit stream. In addition, the input is time interleaved over 16 frames.

MSC Generation -- Main Service Multiplexer

Each output from the time interleavers is multiplexed together in the main service multiplexer. The output of the multiplexer are CIFs. Outputs from the time interleavers are multiples of 64 bits and hence sub-channels. Specifically, each CU is 64 bits, and every sub-channel is an integral number of CUs. The outputs of the time interleavers shall be multiplexed together to form CIFs with multiple sub-channels. Clause 13 describes the procedure for determining CIFs that are inputted into the transmission frame multiplexer.

MSC Generation -- Transmission Frame Multiplexer

As shown in Figure 3.3, one or multiple CIFs are fed into the transmission frame multiplexer before continuing onto the block partitioner. For transmission mode I, the four CIFs shall be multiplexed together to form a vector whose length is 4x55,295 bits. For transmission mode II and III, the input to the block partitioner shall be the one CIF that is transmitted of length 55, 295.

MSC Generation -- Block Partitioning

The input vector to the block partitioner shall be divided into consecutive blocks. Each block will contain the bits transmitted in the MSC OFDM symbols. For transmission mode I, the input vector containing four consecutive CIFs shall be subdivided into 72 blocks for OFDM symbols l= 5, 6, 7,..., 76. Each block will be of length 3072 bits. Details of the block partitioning can be found in subclause 14.4.2.1 for transmission mode I. For transmission mode II, the input vector containing one CIF shall be subdivided into 72 blocks for OFDM symbols l=5, 6, 7,...,76. Each block will be of length 768. Details of the block partitioning can be found in subclause 14.4.2.2 for transmission mode II. For transmission mode III, the input vector containing one CIF shall be divided into 144 consecutive blocks for OFDM symbols l=10, 11, 12,...,153. Each block will be of length 384. Details of the block partitioning can be found in subclause 14.4.2.3 for transmission mode III. Each block shall be designated as $p_{l,n}$ for $n=0...383$.

MSC Generation -- QPSK Symbol Mapper

For each block, $p_{l,n}$, the vector shall be mapped on the K complex QPSK symbols as defined in "FIC Generation -- QPSK Symbol Mapper" on page 48 . The resulting output vector shall be designated $q_{l,n}$.

MSC Generation -- Frequency Interleaving, Differential Modulator, and OFDM Symbol Generator

Frequency interleaving, differential modulation and OFDM symbol generation for the MSC is done identically to the procedures described for the FIC. Please refer to those sections for further clarification.

3.6 DAB Transmission Signal

The Eureka 147 system has a Transmitted Identification Information (TII) mechanism that can be added to the output of the OFDM symbol generator to constitute the final DAB transmission signal. This is shown in Figure 3.1. The TII is conveyed in the Synchronization channel. This information conveys the identification of each transmitter in a DAB network. The TII signal is transmitted in the Null symbol of the Synchronization Channel. The TII signal is specified for all three transmission modes. It can be helpful in determining the geographical location of the receiver. The details of the signal construction are given in subclause 14.8 of ETS 300 401. The

TII feature is optional.

The DAB transmission signal consists of a succession of consecutive transmission frames of 96 ms duration for transmission mode I, and 24 ms duration for transmission modes II and III.

Chapter 4

Understanding Eureka 147

4.1 Background

This chapter is provided to give the reader a better understanding of Eureka 147 DAB system. Specifically, we address questions concerning why certain procedures where employed in this system. We begin with a brief summary of the Eureka 147 DAB requirements. The chapter then continues with an explanation of the OFDM symbols and transmission modes. This will show how the OFDM multicarrier scheme and the transmission modes have been used to facilitate recovery of a transmitted signal that has suffered multipath fading. In addition, this chapter will explain how procedures such time interleaving and convolutional encoding contribute to fulfilling the DAB requirements. We conclude this chapter with some Eureka 147 DAB signal facts.

4.2 DAB Requirements

During the development of the Eureka 147 DAB system, the European Broadcasting Union (EBU) in joint conjunction with the Eureka 147 project team determined a set of requirements that a DAB system should fulfill. These requirements where high quality sound, ability to transmit value added services, low bit rate capability, local, regional and national service availability, service information, ability to access blocks of data, mobile, portable, and fixed reception, immunity to multipath and interference, universality, spectrum efficiency, low complexity, and various operational requirements. With these requirements in mind, the project team and EBU were able to develop a system that utilized a variety of coding techniques to support them. One of the primary concerns was the development of a modulation technique that would support such stringent requirements. [15, p. 3]

4.3 OFDM Modulation

After some extensive research into various modulation techniques, the EBU and Eureka 147 Project Team felt that Coded Orthogonal Frequency Division Multiplex (COFDM) would be the modulation technique employed by the DAB system in development. Techniques such as spread spectrum were also explored. However, due to low spectrum utilization, the possibilities of using spread spectrum was abandoned. [15, p. 13] "COFDM was developed in France within the framework of the Eureka 147 (DAB) Project to meet the exacting requirements of high bit-rate transmission to vehicular, portable, and fixed receiver."[15, p. 7]

COFDM, commonly referred to as OFDM, is a multicarrier scheme that takes the total signal and subdivides it into bit streams each with a lower bit rate. This allows for symbols to have a longer duration than the expected delay spread for the transmission channel. In addition interleaving and coding procedures are used to mitigate the effects of frequency selective fading. This combined process produces a signal that is less sensitive to multipath propagation.

This OFDM concept lies on the two-dimensional distribution of modulated symbols. In time, the symbol spacing allows for guard interval that protects against delay spread. In frequency, the symbols are overlapping one another such that each symbol's peak amplitude occurs at a zero crossing for all other symbols. [11, Vol. III, p. 5001] Hence, the system characteristically has high spectrum efficiency because the OFDM symbols only require guard intervals in the time domain.

The Eureka 147 DAB signal is transmitted in a frame structure. Each frame is divided into a integral number of modulated carriers dependent on the transmission mode. These carriers are modulated using Differential Quadrature Phase Shift Keying (D-QPSK). "The modulated carriers are derived from the orthogonal base of the discrete Fourier Transform by using a vector of complex coefficients defining the I and Q components of each element of the base." [15, p. 13] This procedure is done by the block partitioner and QPSK symbol mapper. The differential modulation is applied after the frequency interleaving. It is applied to facilitate bit recovery at the receiver.

Each OFDM carrier contains two bits of Gray-coded QPSK data. [11, Vol. III, p. 5003] Details of the convolutional encoding have been provided later in this chapter. Therefore, determining the parameters for the convolutional coding, modulation technique, frequency interleaving, time interleaving, guard interval, and bandwidth consequently defines a COFDM signaling procedure.

4.4 Transmission Modes

The Eureka 147 system has the capability of operating in four different transmission modes. These modes are known as Transmission Mode I,II, III, and IV. However, details of mode IV will not be discussed in this document. Further information can be found by consulting the references provided at the end of the document. The transmission modes allow the Eureka 147 to operate at a wide range of frequencies from 30 MHz to 3 GHz. Transmission mode I can operate at frequencies from 30 MHz to 375 MHz. Transmission mode II can operate at frequencies up to 1.5 GHz. Transmission mode III can operate at frequencies to 3 GHz. [11, Vol. I, p. 1006] These modes have been designed to allow variation in implementation for the DAB network. The system can be implemented using terrestrial transmitters, satellite broadcasting, or as a cable distribution network. In addition these modes are designed to compensate for the delay spread and Doppler spread that can occur in a mobile radio environment.

In a mobile radio environment, a transmitted signal is likely to undergo severe degradation due to multipath propagation. Multipath propagation occurs when reflections of the transmitted signal arrive at the receiver dispersed with respect to each other in time and space. When these multipath echoes combine, it produces a received signal that has a longer duration than excepted. This phenomenon is known as delay spread. [24, p. 18] With respect to Eureka 147, the OFDM symbols could spread in time. This would result in overlapping OFDM symbols and therefore causing intersymbol interference. To account for this phenomenon, each OFDM symbol has a guard interval of duration Δ as given by Table 3.2. For frequencies above 30 MHz, research shows that the delay spread is independent of the operating frequency. [24, p. 43] Eureka 147 has been designed with guard intervals that are consistent with this research. [15, p. 42] In addition, multipath may cause the received power of some of the carriers to be reduced due to destructive interference. Frequency interleaving and coding counters this problem.

The delay spread phenomenon also determines the maximum SFN transmitter separation. In an SFN, identical programs are transmitted on the same frequency from different transmitters. As a consequence, the transmitters are limited in their separation. For transmission mode I, II and III, the maximum transmitter separation for an SFN network is 96 km, 24 km, 12 km respectively. Transmission mode I is appropriate for an SFN network, since it allows for the maximum transmitter separation. Transmission mode II is appropriate for local radio broadcasting, requiring one or two transmitters. Transmission mode III is appropriate for cable of satellite based broadcasting. [11, Vol. I, p. 1006]

Doppler spread also contributes to signal degradation. Doppler spread occurs when there is a spreading in frequency of the received signal due to motion. The frequency spread has to be well below the carrier spacing, $1/T_U$, for economic receiver design. The carrier spacing is 1000 Hz, 4000 Hz, 8000 Hz for transmission modes I, II, III respectively. The maximum Doppler frequency occurs at V/λ away from the carrier frequency, where V is the velocity of the moving vehicle and λ is the wavelength of the carrier frequency. [24, p. 29] The Eureka 147 system would experience a maximum Doppler frequency of 200 Hz from the maximum operating frequency of 375 MHz for transmission mode I, 800 Hz from the maximum operating frequency of 1.5 GHz for transmission mode II, and 1600 Hz from the maximum operating frequency of 3 GHz for transmission mode III for a mobile vehicle travelling at 160 km per hour. This indicates that the Eureka 147 system is designed to accommodate Doppler spread even in these extreme conditions.

These features combine to allow for a multipath echo with maximum delay of 300 μs for transmission mode I, 75 μs for transmission mode II, and 37.5 μs for transmission mode III without causing intersymbol interference. [13, p. 4]

4.5 Time Interleaving

Before the OFDM symbols are generated, parts of the signal are processed through a time interleaver. Specifically, the Main Service Channel (MSC) is time interleaved, and the Synchronization Channel and Fast Information Channel (FIC) are not time interleaved.

The time interleaving process covers 16 frames of information. However, this accounts for a 384 ms processing delay at the receiver. This delay is nominal when compared to the advantages that time interleaving gives. Time interleaving allows the signal to go through severe degradation, including spectrally flat fading where most of the OFDM symbols are degraded in a frame. [11, Vol. III, p. 5010] For example, if a multipath fade severely degrade a complete frame, the deinterleaving process would reveal that the fade only degraded parts of 16 frames of information. The error correcting code would be able to correct these dispersed errors, but would not be able to correct a frame full of errors that was not time interleaved.

The synchronization channel and the FIC are not time interleaved. The time phase reference symbol in the synchronization channel is used to sync on the first symbol for the FIC. The 384 ms delay to process the Synchronization channel would not allow for the receiver to obtain lock on the signal. The FIC is not coded for similar reasons. The FIC contains the Service Information (SI) and Multiplex Configuration Information (MCI). The SI is used to transport information concerning the services featured in the MSC. The MCI is used to transport information concerning the organization of the sub-channels in the MSC. It is also used to indicate a multiplex re-configuration to the receiver. As a consequence, the SI and MCI must be processed without any delay. However, the FIC has a high protection ratio of 1/3 to allow for severe signal degradation. This means that for 1 source encoded bit, there are 3 output bits after convolutional encoding. The time interleaving process has been briefly explained in Chapter 3.

4.6 Convolutional Encoding

Before the signal proceeds to the time interleaver or OFDM symbol generator, it is convolutional encoded to allow for forward error correction. The convolutional encoding is a process whereby redundancy is added to the transmitted data. This is another preventative technique against multipath signal degradation.

The convolutional encoder uses equal error protection for the data and unequal error protection for the music. Since the FIC is not time interleaved, it must have a high average protection ratio of about 1/3. The average protection ratio is defined as the number of source encoded bits to the number of encoded bits after convolutional encoding. The MSC can have a varying protection ratio that is dependent of the type and bit rate of data being transmitted. The protection ratio also varies with the type of medium used to transmit the information. If the Eureka 147 network is being implemented as a cable distribution network, then it would require less protection on its data sources than a network being implemented as a terrestrial transmitter network.

The convolutional coding uses a "mother" code that produces 4 bits for every input bit. The mother code is based on convolutional encoder of constraint length seven. This means that it has a memory of 6 bits. Each bit along with 6 previous bits are used to produce the binary parity symbols. The details of the convolutional mother code can be found in subclause 11.1.1 of ETS 300 401. As a consequence, the resulting codeword vector is four times the length of the original input vector.

A puncturing procedure is then used to remove some of the bits introduced by the mother code. For the FIC, the puncturing procedure removes roughly 1/4 of these bits thus producing an output vector of protection ratio 1/3. For the MSC the puncturing procedure produces an output vector based on the protection profile for the transmitting signal. The protection profile is a set of allowable protection levels based on the bitrate of the transmitting signal. A protection level of 1 indicates the highest protection and a protection level of 5 indicates the lowest protection level. These varying protection profiles will output data with different protection ratios from the convolutional encoder. [11, Vol. III, p. 5001]

4.7 DAB Transmission Signal Facts

Each DAB channel has a bandwidth of 1.536 MHz. Six audio programs can be transmitted in DAB channel with high-quality stereo sound. [11, Vol. I, p. 1002] Synchronization on the DAB channel can be obtained in less than 1 sec. [13, p. 16]

The DAB ensemble has a gross data rate of 2432 kbits for transmission modes I and II and 2448 kbits/sec for transmission mode III. For the FIC in transmission mode I and II, the gross bit rate is 96 kbits/sec resulting in a net capacity rate of 32 kbits/sec due to the 1/3 error protection code rate. For the FIC in transmission mode III, the gross bit rate is 128 kbits/sec resulting in a net capacity rate of 42,667 bits/sec due to the 1/3 error protection code rate. [11, Vol. I, p. 2013]

The MSC has a gross bit rate of 2304 kbits/sec. Since the MSC can have varying protection profiles, the net capacity rate will also vary. Hence the net capacity rate can vary from 600 kbits/sec, for protection level 1, to 1.73 Mbits/sec, for protection level 5. [11, Vol. III, p. 2012-2013]

Audio services have the option to be encoded at different bit rates. For a monophonic channel, the encoded bit rate can be: 32, 48, 56, 64, 80, 96, 112, 128, 160, or 192 kbits/sec. For a stereophonic channel or a dual channel, the encoded bit rate can be twice that of a monophonic channel. The encoding rate is left to the discretion of the broadcasters. In general, sound quality and desired number of programs offered in an ensemble will determine the required bit rate. [11, Vol. I p. 1004]

The joint Eureka-147 DAB WG1/EBU Task Force has written "Eureka 147: Digital Audio Broadcasting, Definition of Ensemble Transport Interface" and "Eureka 147: Digital Audio Broadcasting, Guidelines for Implementation and Operation, Vol. I, II, III" to help facilitate the implementation and understanding of the Eureka 147 DAB system. These documents provide extensive details concerning the system and should be consulted if further understanding of the system is desired.

Chapter 5

Eureka 147 for the United States

5.1 Background

As discussed earlier, the Eureka 147 DAB system has had difficulty gaining acceptance by the broadcasting industry of the United States. Much of the opposition to this system originates from issues concerning coverage area. The broadcasters in the U.S. would like to adopt a standard that would give them the flexibility of determining their coverage area. Specifically, they are in favor of a DAB system that would maintain the current status quo of an AM or FM channel's coverage area. Broadcasters have spent much time and effort in providing a service that is tailored for their audience, and they will not accept a system that will require multiple broadcasters to collectively agree on one coverage area. This chapter provides a possible implementation scheme for the Eureka 147 DAB system that would allow broadcasters the flexibility to determine their individual coverage. It is based on the concept of Time Division Multiple Access (TDMA). This method may provide the solution for DAB that broadcasters are looking for.

5.2 The Problem

The Eureka 147 DAB system is being widely implemented as a SFN system in most of the European countries. A SFN system allows the same program to be transmitted without regional limits. In a Eureka 147 SFN system, multiple programs are transmitted from multiple transmitters but on the same frequency. All of the programs for that frequency have the same coverage area. This method is very appealing for the European broadcasting environment where broadcasters operate on a national level. This would allow their programs to be transmitted on the same frequency throughout the country. Furthermore, this would maintain their status quo concerning their audience since most programs are available throughout the country but on different frequencies. The

European radio broadcasting network is similar to the U.S. television network system, where networks such as ABC, NBC, CBS, and FOX are available throughout the country but on different TV channels. A SFN implementation does not decrease listenership for European broadcasters, it makes accessing programs easier.

Although the SFN concept is appealing to the European nations, the U.S. broadcasters feel differently. A SFN implementation would force broadcasters to share coverage areas. However, since the Eureka 147 system is being considered for implementation in the L-Band, multiple transmitters will have to be used to mimic the same coverage area as one or two transmitters would have at a lower frequency. At high frequencies, broadcast communication systems are subjected to line of sight conditions. This means that a receiver must have a nearly unobstructed path to the transmitter in order to detect the signal. Hence, to ensure that a large percentage of receivers have such paths, multiple transmitters must be placed around the region of coverage to account for areas that experience shadowing.

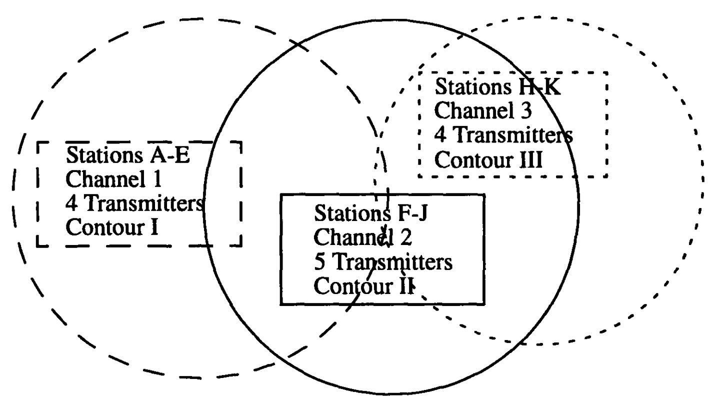

Figure 5.1: Typical SFN implementation for Eureka 147 DAB system

Figure 5.1 illustrates a Eureka 147 SFN system for a particular area. We see that any one station cannot have a smaller coverage area than any one of the contours. The coverage area, for a DAB ensemble, would be determined by the combined coverage areas for the set of transmitters of that ensemble. For example, station A would not be able to service only the lower left portion of Contour I if it wanted to do so. Instead it is forced to service all of Contour I. Furthermore, if station E in Contour I wanted to extend its coverage area, it couldn't unless stations A-D agreed to add additional transmitters. This is the problem that most broadcasters see with the Eureka 147 DAB system. They are aware that Eureka 147 will be able to provide the technical advantages of DAB, but they are concerned of the political and economic issues that they would have face if the FCC were to accept it as the standard for DAB in the U.S.

5.3 The TDMA Solution

Many who have studied the Eureka 147 system, have asked the question, whether it is possible to modify SFN implementation to account for tailored coverage area. Using TDMA, there are methods that will allow broadcasters to designate their coverage area. The following is a description of one such method.

The main signal for Eureka 147 is transmitted using a frame structure. Each DAB frame consists of high-quality sound programs multiplexed together to form the main signal. Each transmitter in a SFN transmits this same multiplex over its broadcast area for a particular channel frequency. Generally in a SFN, the OFDM signal is generated at one site and sent to the transmitters through a ground or satellite link. Upon receiving the signal, the transmitters broadcast the signal over the air. This allows a receiver to detect a strong signal even on the edge of the combined coverage area of all the transmitters. As a consequence, this method requires all programs in that multiplex to share the same coverage area.

The TDMA solution uses this same SFN structure but imposes some further constraints to allow for individual coverage areas. These constraints are determined by the number of programs, and their desired coverage area. In the TDMA solution, each audio program is designated a time interval in the frame. For example, if a channel had five programs, the frame would be subdivided into 5 time intervals where the each station would transmit its audio program. This would be a fixed time interval within the MSC of the frame and would not vary depending on the program type. Then for each transmitter in a SFN, only those programs that desired that coverage area would transmit their signal. However, each transmitter must be capable generating the full spectral OFDM signal because each transmitter still needs to be able to transmit the multiplex for some programs. This would require OFDM signal generators at each transmitter as opposed to distributing the OFDM signal through a satellite link to each of the transmitters.

In addition, this method would also require a synchronization network which would assure the synchronization of the transmitters. This may prove very difficult to do. Essentially, the signals that are sent from the various transmitters must arrive at the reciever within the guard interval designated for that transmission mode. Since we are looking at implementing this system at L-Band frequencies, we would be operating in transmission mode II. Transmission mode II requires that the maximum delay a multipath echo can experience is 75 μs before becoming destructive to the received signal. Therefore, to ensure that this time delay is not violated, we would require that all received signals arrive within this interval of each other. This is severely constrains our synchronization error at the transmitters.

The Joint Eureka 147 DAB WG1/EBU Task Force has written "Eureka Project 147, Digital Audio Broadcasting System, Definition of the Ensemble Transport Interface." In this document, the specification for distributing a multiplexed signal to different transmitters each with their own OFDM generator is discussed. This TDMA solution must comply with the restrictions listed in this document. We also know that synchronization for transmitters that are fed the signal in this manner must be time synchronized within 10% of the guard interval. In addition they must also be frequency synchronized within 1% of the carrier spacing. [11, Vol. III, p. 4011] Since the TDMA solution would provide signals to the reciever that were not always identical but subsets of each other, it is necessary that at the very least the above specifications be observed. However, further analysis of this issue must be administered before accepting these as the synchronization constraints. It must also be noted that time and frequency reference can be obtained with the use of the Global Positioning System (GPS) or synchronizing pulses derived from a satellite TV channel. [11, Vol. III, p. 4014]

Those signals that are not broadcast from that particular coverage area would require special handling. The details of which will be discussed later in this chapter. The signal is then multiplexed and processed through an OFDM signal generator at each transmitter before being sent. This sets up the coverage map, shown in Figure 5.2, for a particular channel frequency.

Although this solution seems to be very simple, there are some other details that must be taken into account in order to insure reception of the signal. The first addresses the question that one may bring up concerning TDMA. Specifically, why must the stations be designated to a particular time allotment in the frame? Why is not possible to simply allow each transmitter to transmit a DAB signal that is a multiplex of the stations that want to be included in that coverage area? The problem is that of co-channel interference. Specifically, two different multiplexes can not transmitted

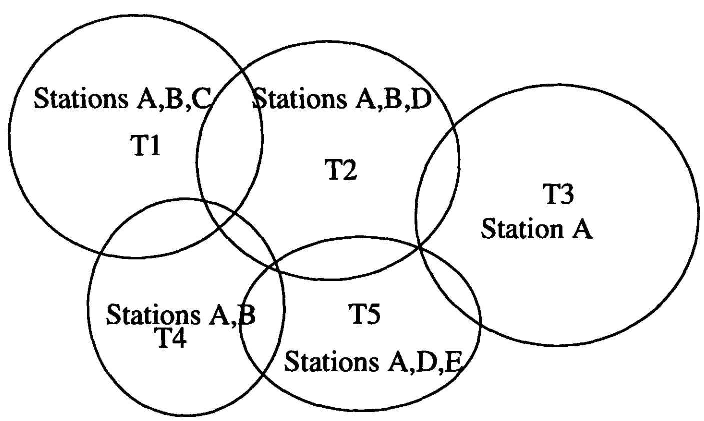

Figure 5.2: SFN network using TDMA

on the same channel frequency within a certain distance of each other. For example, in Figure 5.2, lets assume that stations A-E were not assigned their corresponding time slots 1-5. Instead each transmitter, T1-T5, transmitted a signal that contained its corresponding stations. Hence, for T1, stations A, B, and C could be transmitted in slots 1, 2 and 3 in the DAB signal, and for T2, stations A,B, and D could be transmitted in the slots 1,2 and 3 in the DAB signal. It is clear to see from the figure, that in the region where the coverage area for T1 and T2 slightly overlap, the receiver would not be able to decide which signal to decode and would mute the program. This is because two different signals were transmitted in slot 3 and were transmitted on the same frequency within close proximity of each other.

On the other hand if stations A,B, and C were transmitted in slots 1, 2 and 3 for T1 and stations A,B and D were transmitted in slots 1,2 and 4 for T2, there would not be a co-channel interference problem. A receiver in the overlapping region would be receiving two signals that did not differ from each other where data was being transmitted. It would receive stations A and B from both transmitters and receive station C from T1 and D from T2, and C and D would not overlap in any part of the multiplex.

The second detail that must be taken into account when designing this suggested implementation technique is the Service Information (SI) transmitted in the frame. The SI provides the receiver data concerning the available programs in the DAB ensemble. In addition, the Multiplex Configuration Information (MCI) is also transmitted in the FIC and this allows the receiver to decode the data properly. If the SI and MCI are different for each transmitter, the receiver will also see differences in the transmitted signal and therefore, not be able to decode the signal.

This gives rise to the question that if the SI and MCI indicate to the receiver that a station is available in the ensemble when in fact it is only available on certain transmitters, how will that affect what is displayed to the consumer? This issue can be resolved with the use of the Transmitter Identification Information (TII) option.

The TII option is used for unambiguous identification of each transmitter. This information is transmitted during the null symbol of the Synchronization Channel. Each transmitter sends a unique signal that identifies it. This is done by designating each transmitter with a subset of the total OFDM carriers that are sent. The TII signal uses a set of 384 carriers for Transmission mode I and II, and 192 carriers for transmission mode III. The carriers are divided into 24 combs of carrier pairs. These carrier pairs are switched on four times, each sending four carriers, in transmission mode I, one time, each sending four pairs, in transmission mode II and one time, each sending two pairs, in transmission mode III. [11, Vol. III, p. 5012] The receiver is designed to decode this carrier configuration and be able to decide its approximate geographical location with respect to the SFN. Currently, this feature is used for the Traffic Message Channel, Frequency Information, announcements, and the local area service feature outlined by Eureka 147. This local area service feature must not be confused with the local broadcast service that this TDMA solution would provide. The one outlined by the Eureka 147 developers uses this feature to transmit local services within a SFN, but each local service is transmitted in the same exact part of the DAB ensemble and must be separated from each other by a significant distance in order to avoid co-channel interference.

The FIC information can be configured to indicate which stations are available on which transmitters. Upon decoding the TII, the receiver would know its availability of transmitters. Cross referencing this with the FIC information would indicate what stations where actually receivable. Therefore, even though the MCI and may indicate that stations A-E are available in the ensemble, this additional information could be used to indicate to the consumer that a particular station is not available but may be available near by. There is also another possible advantage to using this option. After cross referencing the transmitter identifiers with stations, the reciever could decrease the amount of time it takes to decode the signal by ignoring the signal that is being received in the time slots where stations would not be transmitting for the nearby transmitters.

The final and most important detail concerns programs that are not transmitted for a particular coverage area but are featured in that ensemble. In any particular time slot in a DAB frame where data is not being transmitted, we must account for the possibility of noise transmission. A transmitter will feature programs that are not available in an adjacent transmitter. We must specify what is sent during the time interval where a transmitter is not sending data. Transmitting null data or digital zeros can be destructive to the system since even digital zeros will have a noise power associated with them. Where coverage areas overlap, a receiver would have difficulty decoding stations where the received signal is a sum of the signal from the transmitter featuring the program and the transmitter outputting digital zeros which results in a noisy signal.

Therefore, the optimal solution would be to be able to output no power during the time interval where a program is not featured on that transmitter. "One consequence of the multi-carrier technique, with the statistical nature of the carrier phases, is a relatively high peak-to-mean ratio of the signal amplitude in the time domain." [11, Vol. III, p. 5001] From this we can infer that the transmitter must have a wide dynamic range of power levels. If the transmitter has the capability to turn itself off, this would comply with our optimal solution to the problem.

However, this is an assumption that may not hold true. There are other options that we can explore to lessen the noise level on the received signal. One option is to transmit at the minimal power that a transmitter is capable of. This may or may not be enough depending on the dynamic range of the transmitter. If this solution does not alleviate the problem, we may be able to employ a transmitting procedure similar to the one used by the TII option. The TII option shows us that the transmitter has the capability to switch on certain carriers in order to convey its transmitter id. Hence, it is possible that we can transmit only some of the carriers that are transmitted for that particular program in the designated time interval. Research has shown in theory that if every tenth carrier is bursted with this minimal power, the signal to noise ratio is degraded about a 1-1.5 dB. This is for a system that employs rate 1/2 coding and constraint length 7. [4, p. 3]

In order to address the coverage area issue with a TDMA solution, this particular problem will have to be solved. In fact, it is unlikely that any TDMA solution can be implemented without addressing this problem first. We are confident though that since the spectrum has a characteristically high peak to mean ratio and the system employs the transmitting techniques of TII, a reasonable solution can be found.

Using TDMA in conjunction with the TII option would solve many of the synchronization problems that would arise if each station were to transmit only their part of the frame with a specific part of the FIC and Synchronization Channel. The key point is that to the receiver, the signals should not appear to be different. They can be subsets of each other but any variation in the signal besides having parts of the signal non-existent could severely hinder the signal from being detectable. In this method, it would not be feasible to transmit another station in the same time slot but within the same overall coverage area. Hence, this TDMA solution does limit frequency re-use within one region.

However, there are some further modifications we can make in order to possibly allow frequency re-use with this TDMA solution. Figure 5.3 illustrates the scenario we are interested in modelling. We can see from the figure that the DAB ensemble consists of 5 time slots. Each time slot is designated to a particular station. However, in our scenario we have two very small stations that would like to be included in this DAB ensemble. This poses a problem since there is only one time slot available and two stations requesting to transmit during it.

Before considering the co-channel interference problem, we must first account for the header information. Specifically, how can we indicate to the receiver that both stations are available but only one at a time. This can be done in the same fashion as before using the TII. The difference is that we must now redefine the SI and MCI in order to allow both stations to co-exists within the same ensemble. This way, all transmitters can send the same header indicating that stations A-D are available in slots 1-4 and station E and F are available in slot 5. Then with the use of the TII, we can cross reference the transmitter id with the stations that are available. Once this is complete, the appropriate SI and MCI can be displayed and processed.

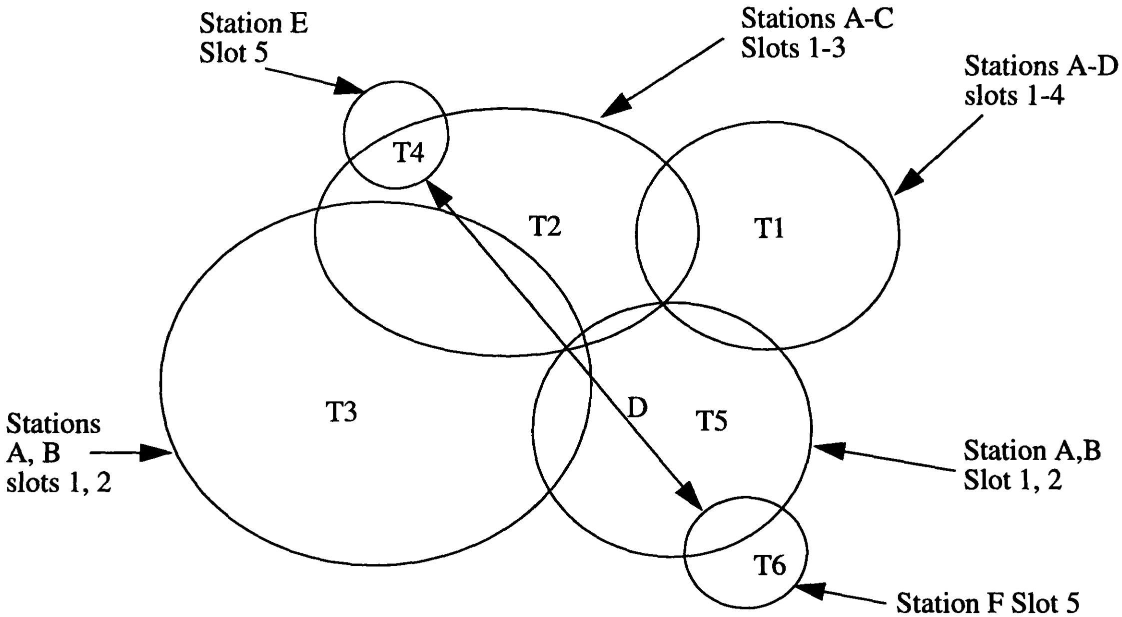

Figure 5.3: A Frequency Re-use Scenario

This scheme will give an extra degree of freedom when planning out networks for metropolitan areas where frequency allocation is a limiting factor. The difficulty in this system arises with the co-channel interference problem. Specifically how large does the distance D have to be between the T4 and T6 in order for the two signals not to interfere with each other. We have already accounted for problems that the receiver would have encountered from T1, T2, T3 and T5 indicating that different stations were available in slot 5 with respect to each other. This problem is being studied with the local service option by Eureka 147. It is commonly referred to as the local windows. "The main disadvantage is that this approach only admits islands of coverage for the local services due to mutual interference. Therefore, there will be an area between the different local services using the same localized symbols in which it is not possible to receive any of the them. Since the coverage area is limited by interference an increase in power will not improve the coverage." [11, Vol. III, p. 5027] Essentially, the problem would occur somewhere in T2 or T5's coverage area where the received signal from T4 and T6 could have equal strength. One option to avoid this problem is to avoid decoding slot 5 while in this region where the signal would not be detectable. Since the TII option allows us to determine which transmitters are visible, the receiver may be able to configure a state transition map whereby one of the states may be that if the transmitter id from both T4 and T6 is visible, then to simply ignore the transmitted signal in slot 5 since it will be a conflicting one. This would ensure the reception the of the remainder of the frame.

Another option that would help decrease the dead-band area for slot 5 would require the use of low power transmitters. Placing an additional constraint on the output power level for transmitters would facilitate frequency re-use. A interfering signal must be at least 8dB below the desired signal in order for accurate reception of the desired signal and 20 dB below for comfort. [26, p. 330] Therefore, the use of low power transmitter would facilitate frequency re-use. For example, if station E used two low power transmitters for its desired coverage area instead of one high power transmitter, then the interfering signal strength on station F's signal would be lessened even though station E still attained its desired coverage area. Consequently, this would decrease the required distance D between stations that are operating on the same frequency in the same time slot. This, however, would require stations to agree to the use of multiple low power transmitters in order to avoid interference on other stations that may also be transmitting in their slot.

In addition, directional antennas can be used to facilitate frequency re-use. For example, station E may use directional antennas to ensure the signal is directed away from station F. This way, the received interfering signal strength on station F's desired signal will be less than if a omni-directional antenna was used.

The distance D will vary from station to station depending on the desired coverage area and the type of transmitters used to obtain it. In addition, we must take into account who the service is being provided for. Automobile antennas generally do not have any directivity associated with them. On the other hand, household antennas may. The chances of a directional antenna detecting a weak signal are greater than that of an omni-directional antenna for any pre-determined direction. Therefore, if a service is being provided for an household environment with a steerable directional antenna, the distance D can be reduced.

Frequency re-use is a very complex problem and must be investigated further before any conclusions can be drawn as to its feasibility. We must also account for receiver complexity when trying to develop solutions to the frequency re-use problem.

When considering other possible TDMA implementations, it is not feasible to allow one transmitter to transmit the header information. This would not allow the use of the TII option. Indicating that a program is available when it isn't would cause confusion to the consumer. One method that has also been introduced, is the idea of not only dividing up the MSC into time slots but dividing the complete frame into 4-6 slots depending on the number of programs available for that ensemble. In this configuration, each slot would be a miniature frame that would be part of the whole frame. It would have its own FIC. This way, the FICs that the receiver was able to decode would indicate what programs where available in the DAB ensemble. This also seems to be a possible solution to the problem and is also being researched further.

As we can see, there are multiple variations to the TDMA solution that may allow for tailored coverage area. Each solution requires its own set of constraints that limits its flexibility. However, there are multiple advantages to implementing such a system. Using TDMA with the TII option would allow stations to tailor their coverage area to their needs. It would also allow small and large stations to share the same frequency channel without adjusting either one's coverage area. Furthermore, stations could increase their coverage area by adding one of the transmitters in their SFN or adding a new transmitter to the SFN if needed. Again, this can be done independently by a station or as collectively with other stations. The FCC could also designate classes to stations as it does not with the FM band. Furthermore, stations could share the cost of additional transmitters.

It is very important to note that major changes to the standard will not be necessary. A few FIG extensions will need to be defined in order to use the TII id to cross reference the available stations. This is an important advantage because this will allow the US to accept a standard that is consistent with most of the countries who are undergoing DAB implementation. In addition, the developers of Eureka 147 would most likely not raise any objections to such minimal changes to the standard. For the TDMA system that uses the TII option with allowable local windows, we must be able to redefine the MCI and the SI in order to allocate different perimeters for different stations featured in the same time slot.

We see that there are multiple advantages to adopting this suggested implementation technique. More importantly, it gives the broadcaster an option to look at. This could possibly facilitate the acceptance of a DAB standard for the U.S. broadcasting environment.

Chapter 6

Conclusion

The Eureka 147 DAB system can offer technological advancements to the U.S. broadcasting environment. It has the capability to not only offer high-quality sound programs but also general data services. It is a system that has been adopted by much of Europe and other countries such as Canada. This system has a well defined standard that has the capability and flexibility to be implemented in many different broadcasting environments.

The United States has had difficulty accepting this system because of combining of signals to create one ensemble transmitted over a larger bandwidth. This would require all those transmitting in the ensemble to have identical coverage areas. Combine this with the an SFN implementation, which is inevitable at L-Band frequencies, this system becomes unacceptable for the broadcasters of the U.S.

This unfavorable response has caused proponents of DAB to look into a variety of other possible DAB solutions. Some of these solutions propose to provide DAB services within the already existing AM and FM band. However, these systems have yet to prove that they are capable of providing this service without disturbance to the already existing AM and FM band. The results of their ability will be seen at the conclusion of the San Francisco field tests.

Before abandoning the Eureka 147 systems, broadcasters should look into the variety of implementation techniques that can be applied to Eureka 147 in order to address some of their concerns. It is understandable that they may feel that such a system would destroy an infrastructure that has taken many years to develop. However, this system does offer them many services that would benefit their businesses in the future. They are encouraged to study and analyze the system. This will allow them to fully understand what Eureka 147 is and how it is designed.

This thesis hopes to give broadcasters and engineers an opportunity to understand Eureka 147 better. It provides a summary of the technical details as well as a summary of the features of this system. In addition, the author is offering a possible implementation scheme of Eureka 147 that would address major concerns about coverage area made by the broadcaster.

This implementation technique employs the concepts of TDMA. It allows each broadcaster to tailor their DAB coverage area without any serious redefining of the standard for Eureka 147. For the most part, this technique will only require a few more definitions of FIG extensions. It also uses services that are already offered in the system. Specifically, it would utilize the TII option. This option assigns an id tag to each transmitter. The receiver would be able to use the id tag to cross reference which programs are being sent from that transmitter in a particular ensemble. This will allow the transmitters to send identical headers even though the receiver will be able to distinguish what stations are available for service in any one region. Variations of this basic scheme may allow the system to transmit multiple stations from the same ensemble in the same time slot provided they are not in close proximity of each other. In addition, with the use of the TII, the system would be able to avoid creating islands of coverage whereby certain areas in the coverage region for the entire ensemble are not able to receive the signal.

All of this requires careful planning and further investigation. The key issues of synchronization, noise power and identical headers have been pointed out and discussed to some extent. The industry is encourages not to shut the door on Eureka 147. It is a capable system that may actually provide the solution that they are looking for.

From a performance point of view, there is a vast amount of research being conducted in Europe as well as several pilot projects that are being administered. It is in the benefit of the United States to study the results of that research and understand what are the necessary constraints in order to avoid interference when trying to develop systems that are localized.

This will facilitate the design process of any TDMA system that may be considered. Any solution must be able to utilize the strengths of Eureka 147 to its fullest capability. These strength include the ability to use multipath echoes constructively, availability of wide range of services, and its advanced coding procedures to allow for a variable number of programs from ensemble to ensemble. The TDMA solution uses all these strengths.

Broadcasters and officials are urged understand the implications of all the systems that are under investigation. This will facilitate them in making a decision that will be beneficial to the broadcasting industry in the future and not just in the present.

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