INTRODUCTION

Digital audio broadcasting, DAB, is the most fundamental advancement in radio technology since that introduction of FM stereo radio. It gives listeners interference — free reception of CD quality sound, easy to use radios, and the potential for wider listening choice through many additional stations and services.

DAB is a reliable multi service digital broadcasting system for reception by mobile, portable and fixed receivers with a simple, non-directional antenna. It can be operated at any frequency from 30 MHz to 3GHz for mobile reception (higher for fixed reception) and may be used on terrestrial, satellite, hybrid (satellite with complementary terrestrial) and cable broadcast networks.

DAB system is a rugged, high spectrum and power efficient sound and data broadcasting system. It uses advanced digital audio compression techniques (MPEG 1 Audio layer II and MPEG 2 Audio Layer II) to achieve a spectrum efficiency equivalent to or higher than that of conventional FM radio. The efficiency of use of spectrum is increased by a special feature called Single. Frequency Network (SFN). A broadcast network can be extended virtually without limit a operating all transmitters on the same radio frequency.

EVOLUTION OF DAB

DAB has been under development since 1981 of the Institute Fur Rundfunktechnik (IRT) and since 1987 as part of a European Research Project (EUREKA-147).

- In 1987 the Eureka-147 consoritium was founded. It's aim was to develop and define the digital broadcast system, which later became known as DAB.
- In 1988 the first equipment was assembled for mobile demonstration at the Geneva WARC conference.
- By 1990, a small number of test receivers was manufactured. They has a size of 120 dm3
- In 1992, the frequencies of the L and S — band were allocated to DAB on a world wide basis.
- From mid 1993 the third generation receivers, widely used for test purposes had a size of about 25 dm3, were developed.
- The fourth generation JESSI DAB based test receivers had a size of about 3 dm3.

1995 the first consumer — type DAB receivers, developed for use in pilot projects, were presented at the IFA in Berlin.

In short

1992 — 1995 — field trial period.

1996 — 1997 — introduction period

98 onwards — terrestrial services in full swing

For DAB via satellite 1996 — 2001 is planned as experimental stage 2002 — 2003 introduction period.

DIGITAL AUDIO DATA

The conversion of analog audio data to the digital domain begins by sampling the audio input in regular, discrete intervals of time and quantizing the sampled values into a discrete number of evenly spaced levels. The digital audio data consists of a sequence of binary values representing the number of quantizer levels for each audio sample This method of representing each sample with an independent code word is called pulse code modulation (PCM).

The digital representation of audio data offers many advantages.

- High noise immunity
- Stability
- Reproducibility
- Allows the efficient implementation of many audio processing functions (i.e. mixing, filtering, equalization) though the digital computer.

According to the Shannon’s theory, a time sampled signal can faith represent signal up to half the sampling rate. The max audible frequency for humans is 20 KHz. Therefore the typical sampling rate is 48 KHz. (i.e. more than twice the signal frequency).

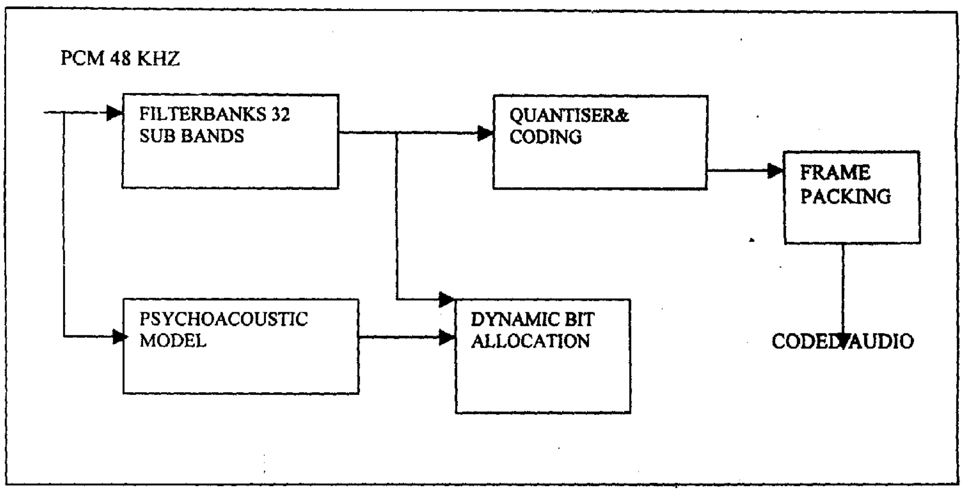
DIGITAL AUDIO COMPRESSION

Digital audio compression allows the efficient storage and transmission of audio data. While quantizing, the number of quantizer levels is typically a power of 2 to make full use of a fixed no: of bits per audio sample to represent the quantized values. With uniform quantizer step spacing, each additional bit has the potential of increasing the signal to noise ratio. The typical number of bits per sample used for digital audio is 8, 16, 32, 64. The audio data on a compact disc (2 channels of audio samp1. at 44.1 KHz with 32 bits per sample) requires a data rate of 32x2x44xl000(megabits per second. Ti) transfer this uncompressed data requires a large data transfer rate and a larger bandwidth. Therefore audio data need to be compressed for efficient storage and transmission.

COMPRESSION TECHNIQUES

The MPEG (Motion Picture Experts Group) audio compression algorithm is an International Standardization Organization (ISO) standard for high fidelity audio compression. The high performance of this compression algorithm is due to the exploitation of auditory masking. This masking is a perceptual weakness of the ear that occurs whenever the presence of a strong audio signal in spectral neighborhood of weaker audio signals makes it imperceptible. This noise-masking phenomenon has been observed and corroborated through a variety of psycho acoustic experiments. Due to the specific behaviour of the inner ear, the human auditory system perceives only a small part of the complex audio spectrum. Only those parts of the spectrum located above the masking threshold of a given sound contribute to its perception, where as any acoustic action occurring at the same time but with less intensity and thus situated under the masking threshold will not be heard because it is masked by the main sound event.

To extract the perceptible part of the audio signal the spectrum is split into 32 equally spaced sub-bands. In each sub-band the signal is quanitised in such away that the quantising noise matches the masking threshold. This coding system for high quantity audio signals is known as MUSICAM (masking pattern adapted universal sub- band integrated coding and multiplexing)



MUSICAM DAB CODER

The input audio stream passes through a filter bank that divides the input into multiple sub-bands. The input audio stream simultaneously passed though a psycho acoustic model that determines the signal-to mask ratio of each sub-band. The bit allocation block uses the signal-to mask ratios to decide how to apportion the total no: of code bits available for the quantization of the sub-signals to minimize the audibility of the quantization noise. Finally, the last block takes the representation of the quantized audio samples and formats the data into a decodable bit stream.

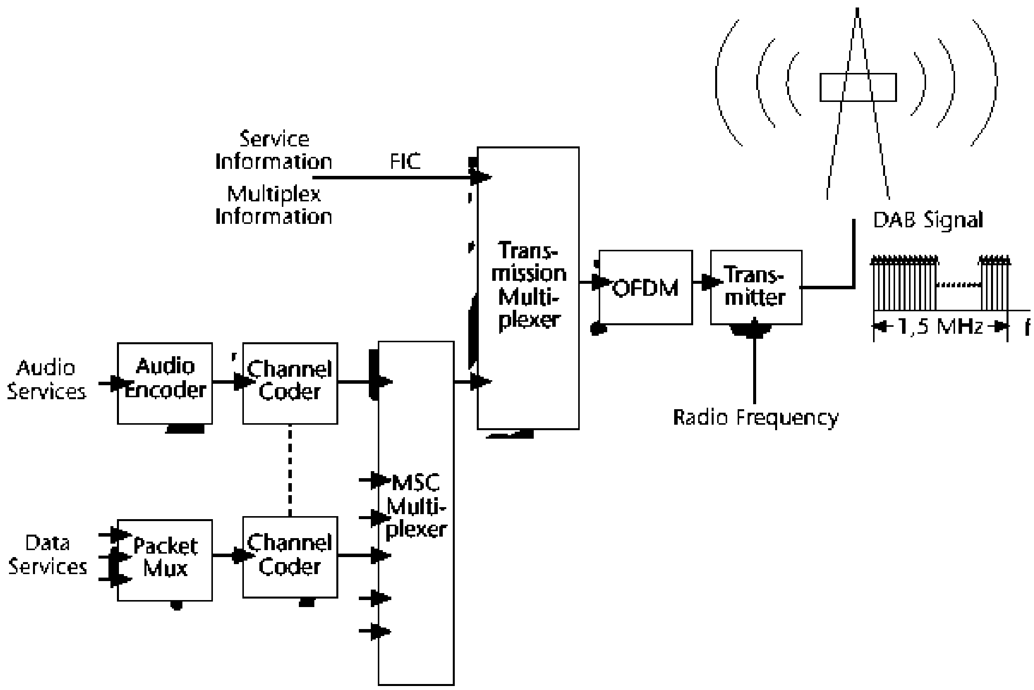
The 32 constant width filter bands reflect the ear's critical bands. With MUSICAM, high quality audio can be perceived with data rates down to 200

Kbs per stereo channel compared to 2,800 Kbs of CDs that use an uncompressed technique.

OUT LINE OF THE DAB SYSTEM

GENERATION OF DAB SIGNAL

The figure shows that block diagram of a conceptual DAB signal generator.



Conceptual DAB Signal Generator

Each service signal is coded individually at source level, error protected and time interleaved in the channel codes. Then the services are multiplexed in the Main Service Channel(MSC), according to a predetermined , but adjustable, multiplex configuration. The multiplexer output is combined with multiplex control and service information, which travel in the Fast Information Channel (FIC) to form the transmission frames in the transmission multiplexer. Finally, Orthogonal Frequency Division Multiplexing (OFDM) is applied to shape the DAB signal which consists of a large number of carriers. The signal is then transposed to the appropriate radio frequency band, amplified and transmitted. The broadcasting frequency for digital audio varies from 30 MHz —3 GHz.

TRANSMISSION FRAME

In order to facilitate receiver synchronization, the transmitted signal 'is designed according to a frame structure with a fixed sequence of symbols. Each transmission frame (See Fig. 3) begins with a null symbols for course synchronization (when no RF signal is transmitted), followed by a phase reference symbol for differential demodulation. The next symbols are reserved for the FIC and the remaining symbols provide the MSC. The total frame duration is 96 ms, 48 ms or 24 ms depending on the transmission mode. Each service within the MSC is allocated a fixed time slot in the frame.

TRANSMISSION FRAME MODE OFDM SYMBOLS

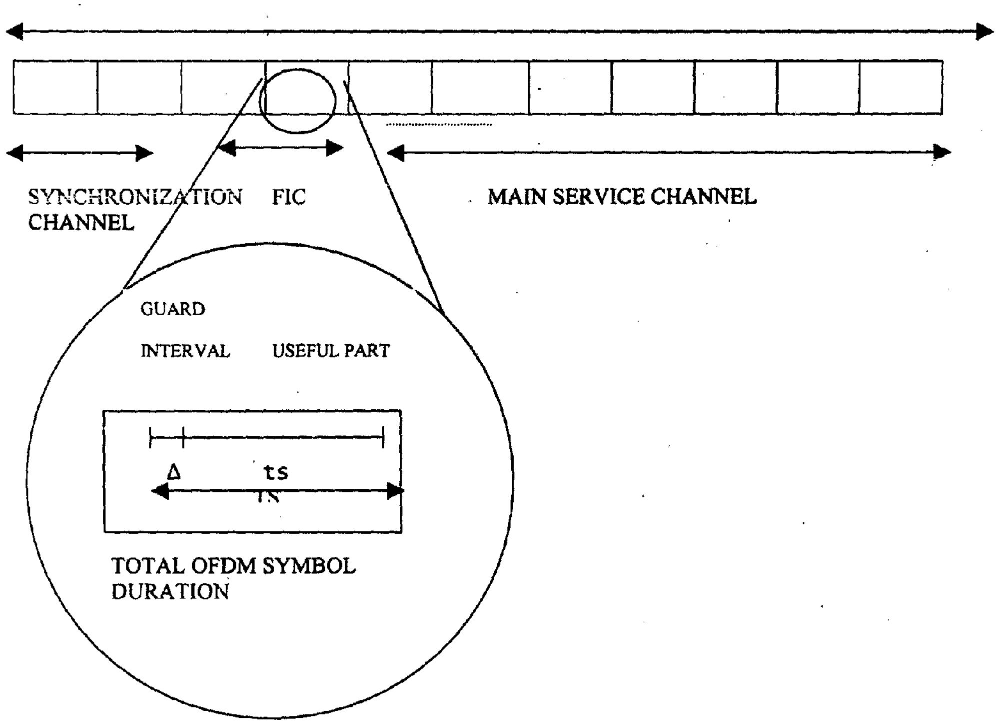


Fig. 3. Transmission frame

MODULATION WITH COFDM AND TRANSMISSION MODES

The DAB system uses a multi carrier scheme known as Coded Orthogonal Frequency Division Multiplexing. This scheme meets the requirements of high bit-rate digital broadcasting to mobile, portable, and fixed receivers, especially in multi-path environment.

The multi-path propagation is likely to produce echoes in reception. The COFDM is a transmission technique by which the complete ensemble (multiplex) is transmitted via several hundred (or even several thousand) closely-spaced RF carriers which occupy a total bandwidth of approx 1.5 MHz, the so-called frequency block. Due to the low data of each RF carrier, any delayed reflections of signal due to multipath propagation will add to the direct signal already received and thus allow interference free reception under conditions of multipath propagation.

Before the transmission, the information is divided into a large number of bit- streams with low bit-rates. These are then used to modulate individual orthogonal carriers in such a way that the corresponding symbol duration becomes larger than the delay spread of the transmission channels (Differential quadrature phase shift keying). By inserting temporary guard interval between successive symbols,channel selectivity and multipath propagation will not cause inter symbol interference.

SINGLE FREQUENCY NETWORK CAPABILITY OF THE COFDM

With analogue broadcasting especially when it comes to mobile receivers such as car radio-reception is often disturbed by aggravating interference in the form of distortion, noise or total failure. The losses also occur due to signal shadowing. Therefore more than one transmitter may be needed to avoid signal shadowing. To avoid interference from neighboring transmitters different carrier frequencies are used for the same FM/AM program. This can lead to spectrum overloading, especially, in densely populated areas with a high number of stations.

In Single Frequency Network (SFN) all transmitters are emitting the same station in the same frequency. The receiver cannot distinguish whether the received signal is a reflected one or comes from a second transmitter. The DAB allows the combination of blocks of stations on single DAB channel of 1.5 MHz band width, without leading to interference. In conjunction with a SFN, a block of at least six stations per country can be broadcasted via the same DAB channel. By using one or more additional DAB channels, it is possible to provide further blocks of stations for regional and local programs. Thus SFN provides superior frequency economy.

The system provides 4 transmission mode options which allows a wide range of transmission frequencies between 30 MHZ and 3 GHZ and network configuration. For the normal frequency ranges, the transmission modes have been designed to suffer neither from Doppler spread nor from delay spread, both inherent mobile receptions with multipath echoes.

The table below gives the temporal guard interval duration. The nominal max transmitter separation and frequency range for mobile reception for the different modes.

System Parameter	I	II	III	IV
Frame duration	96 ms	24 ms	24 ms	48ms
Null symbol duration	1297 µs	324 µs	168µs	648µs
Guard interval duration	246 µs	62 µs	31 µs	123µs
Nominal maximum transmitter separation for SFN	96 KM	24 KM	12 KM	48KM
Nominal frequency range (For mobile reception)	<=375 MHz	<=1.5 GHz	<=3 GHz	<=1.5GHz
Speed 1 coverage Trade-Off	No	No	No	Yes
Useful Symbol Duration	1 Ms	250 µs	125µs	500 µs
Total Symbol Duration	1246 µs	312 µs	156µs	623 µs
Number of radiated carriers	1536	384	192	768

The table shows that the higher the frequencies, the shorter the guard intervals available hence the smaller the max non-destructive echo delay. Mode I is most suitable for a terrestrial single frequency in the VHF range, because it allows the greatest transmitter separation. Mode will preferably be used for medium - scale SFN in L-band and for local radio applications that require one terrestrial transmitter large transmitter spacing can be accommodated by inserting artificial at the transmitters and by using directional transmission antennas.

Mode III is most appropriate for cables, satellite and complimentary terrestrial transmission since it can he operated at all frequencies up to 3 GHz.

ADDITIONAL SERVICES

1 .PROGRAMME ASSOCIATED DATA

Each audio programme contains Programme Associated Data (PAD) with a variable capacity (mm 667 bits/s upto 65 kbps) which is used to convey information together with the sound programme. The PAD channel is incorporated at the end of the DAB/ISO audio frame. The typical examples of PAD applications are dynamic range control information, a dynamic label to display programmed titles or lyrics speech/music indication and text with graphic features.

2. INDEPENDENT DATA SERVICES

In addition to PAD, general data may be transmitted as a separate service. This may be either in the form of a continuous stream segmented into 24 ms logical frames with a data rate of n x 8 kbps (n x 32 kbps) for some code rates) or in packet mode, where individual packet data services may have much lower capacities and are bundled in a packet sub-multiplex. A third way to carry independent data services is a part of the Fast Information Channel (FIC). The typical independent data services are

- Traffic message channel
- Correction data for differential GPS
- Paging
- Electronic newspaper

3.**CONDITIONAL ACCESS**

Every service can be fitted with conditional access if desired. The Conditional Access (CA) system includes 3 main functions.

- Scrambling/descrambling
- Entitlement checking

- Entitlement management

The scrambling/descrambling function makes the service incomprehensible to unauthorized users. Entitlement checking consists of broadcasting the conditions required to access a service, together with encrypted secret codes to enable descrambling for authorized receivers. The entitlement management function distributes entitlements to receivers. This facility brings out the concept of pay radio. It also has a lot of defence applications.

4. SERVICE INFORMATION

The following elements of Service Information (SI) can be made available to the listener for programme selection and for operation control of receivers.

- Basic programme-service label 9i.e. the name of a programme service
- Programme type label (e.g. news, sports, music, etc.)
- Dynamic text label(programme title,lyrics,names)
- Programme language
- Time and date, for display or recorder control
- Switching to traffic reports, news flashes. or announcements on other services.
- Cross reference to the same service being transmitted in another DAB ensemble or via AM or FM and to other services.
- Transmitter identification information (e.g. for geographical selection of information)

Essential items of service Information that are used for programme selection are carried in the FIC of the transmission frame.

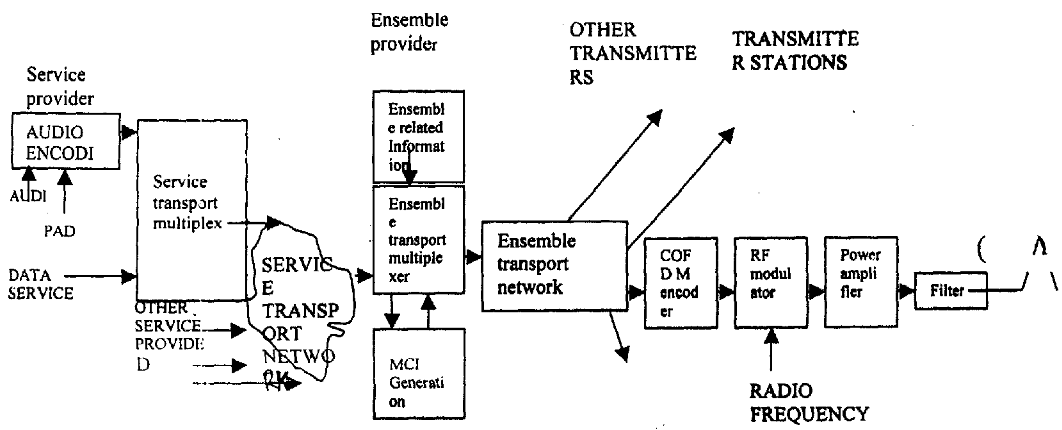
MAIN SERVICE MULTIPLEX

The encoded and interleaved data is fed to the Main Service Multiplex where every 24 ms the data is gathered in sequence. The combined bit stream output from the multiplexer is know as the Main Service Channel (MSC) and has a gross capacity of 2-3mbps.

The DAB system allows the Main Service Multiplex to be reconfigured from time to time. The precise information about the contents of the Main Service Multiplex is carried by the Fast Information Channel to communicate to the receiver how to access the services. This information is known as the Multiplex Configuration Information (MCI). When multiplex configuration is about to change, the new information, together with the timing of the change, is transported via MCI and details in advance what changes are going to take place.

IMPLEMENTATION OF TERRESTRIAL DAB NETWORKS

The specification of the DAB signal (i.e. system parameters discussed earlier) gives full details of the characteristics of a signal which is to be remitted from the transmitters in the form of a DAB ensemble. A conceptual DAB distribution network is shown below:



1. The service provider creates and manages the data that is to become a service in a DAB ensemble.
2. The data provided by service provider is passed to the ensemble provider via the service transport network.
3. The ensemble provider manages the capacity of the complete ensemble. Typically, information about services will be received from many different service provides. This information will then be assembled into a set of data representing the complete DAB ensemble. The ensemble description is passed to the transmitter stations where the DAB ensemble is generated and radiated. The interface between the ensemble provider and the transmission network is known as the ensemble transport interface. It allows the efficient distribution of signals from the DAB ensemble

multiplexer to the COFDM generators of the transmission network, which is most likely a single frequency network.

CHANNEL CODING AND TIME INTERLEAVING

The data representing each of the programme services is subjected to energy dispersal scrambling, convolutional coding and time interleaving. For energy dispersal scrambling a pseudo-random bit sequence is added to the data in order to randomize the shape of the DAB signal and thus efficiently use power amplifiers. The convolutional encoding process involves adding redundancy to the data in order to help the receivór detect and better eliminate transmission errors than others and accordingly 1he amount of redundancy added is reduced for these. This method is known as unequal error protection.

SATELLITE DAB

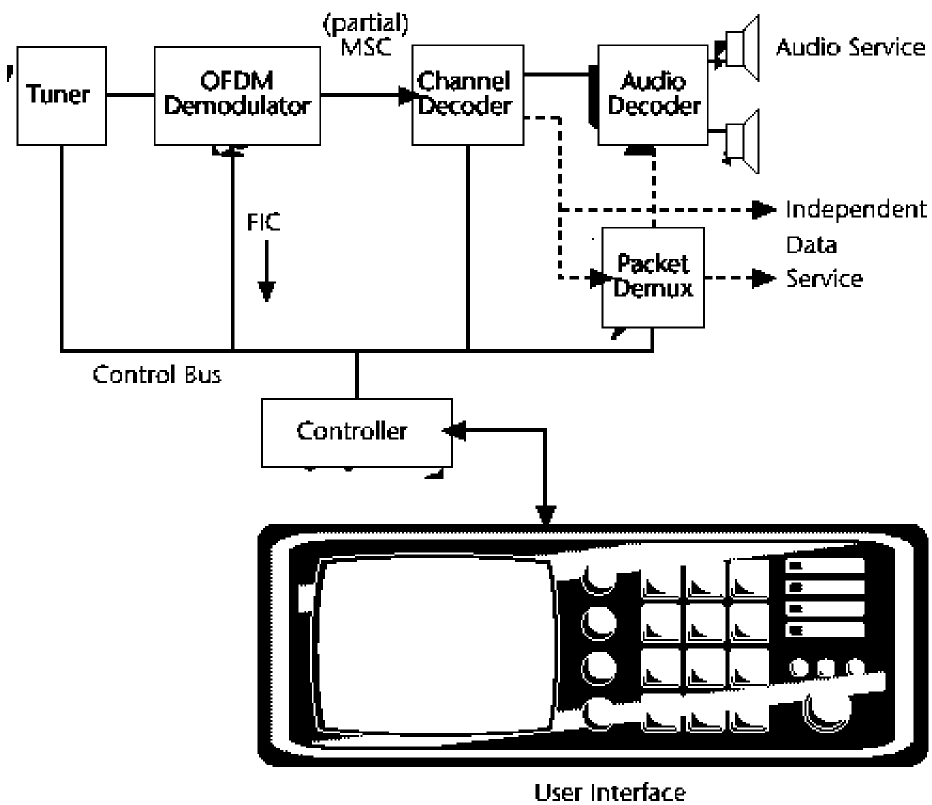
Besides terrestrial transmission the DAB system is suitable for satellite as well as for hybrid/mixed terrestrial/satellite broadcasting, using a simple omni-directional receiving antenna. Satellites will receive the data generated by uplink stations ,amplify this data and send it back through special spot beams not only to fixed, but also to mobile and portable receivers, complementary terrestrial transmitters may be necessary, e.g. in big cities with high-rise buildings. In contrast to conventional TV satellites where radio programmes can only be picked up with the help of special receivers, and dishes have to be installed. The DAB satellite system will have the same modulation/coding system parameters as the terrestrial system. Thus, the same receiver and antenna can be used both for terrestrial and satellite DAB.

Field tests on satellite DAB have been conducted recently — one in Australia, the other in Mexico. Although both test satellites were not specially designed for multi- carrier systems such as the EUREKA-147 DAB system, but for mobile phone service, satellite transmission of DAB signals proved technically feasible. With satellite DAB it will be possible to cover areas much larger than those covered by terrestrial broadcast stations. A geostationary

(GEO) satellite system could cover low latitude areas such as most parts of Africa,central and South American, India, Indonesia etc.

RECEPTION OF DAB SIGNAL

The figure below shows a conceptual DAB receiver. The DAB ensemble(multiplex) is selected in the analogue tuner , the digitized output of which is fed to the OFDM demodulator and channel decoder to eliminate transmission errors. The information contained in the FTC is passed to user interface for service selection and in use to setup the receiver appropriately. The MSC data is further processed in an audio decoder to produce the and audio signal or in a data decoder(Packet Deumux) as appropriate.



Concept of DAB Reception

To achieve low cost and excellent performance a high integration of receiver components into specific integrated circuits is necessary. DAB c sets have to support a variety of receivers , from the affordable portable radio to the state of the art receiver for multimedia services. Advanced single chip DAB system controllers • and data decoders are essentially, which will decisively influence costs and performance of ‘DAB receivers for consumer purpose.

ADVANTAGES OF DAB

Bandwidth requirements are less compared to the analog counterpart. This has been brought about by the efficient compression techniques.

- Better quality audio can be obtained.
- Digital system requires only low power than regular radio signals.
- Error correction is a part of the digital system
- Multipath interference which is the main problem of analog FM is reduced or almost avoided.
- High spectrum efficiency due to single frequency networks. This is made possible by a new and efficient method of modulation: COFDM
- Significant data casting capacity.
- Additional data services

CONCLUSION

More and more countries across the world are switching on to DAB. Their plan is to gradually terminate the existing AM and FM channels, say by 2008 and to use that spectrum for some other purposes. Any how DAB is going to be the Sound of the future. It is in the path of the growth and development. It is going to replace the present methods, even through it may take time. .Its clear that DAB in its infancy has the potential to completely change the way that radio is perceived. Its efficiency in bandwidth , providing greater use of available spectrum, combined with data handling characteristics of the concept provides a sea change in our use of radio as a medium. The ability to inform, entertain, advertise and trade has never seen a more versatile vehicle.

REFERENCES

1 TECHNICAL JOURNAL-ITIS FRANCE -JAN 2000
"MAGIC FEATURES OF THE COFDM"
-BY GERARD FARIA,

2. WWW.GOOGLE.COM

3. DIGITAL TECHNICAL JOURNAL VOL-5 NO.2. "1999DIGITAL AUDIO COMPRESSION"

4. WWW. WORLDDAB.COM

5. WWW.DIR.DE/DAB/

ABSTRACT

Digital radio broadcasting is a technique which gives listeners interference free reception of high quality sound, easy to use radio, and the potential for wider listening choice through many additional station and services. Current analogue FM radio broadcasting systems in VHF band cannot satisfy demands of future ,such as excellent sound quality, large number of stations and small portable receivers and no quality impairment due to multipath propagation. Digital audio technology has set technical quality standards, which are far beyond those available to radio broadcasting transmitted over the analog FM system. One of its lucrative features is improved mobile radio reception. DAB has the potential to make radio as we know it now as antiquated as the crystal models of a bygone age.

CONTENTS

ACKNOWLEDGEMENT

I extend my sincere thanks to **Prof. P.V.Abdul Hameed,** Head of the Department for providing me with the guidance and facilities for the Seminar.

I express my sincere gratitude to Seminar coordinator **Mr. Berly C.J**, Staff in charge, for their cooperation and guidance for preparing and presenting this seminar.

I also extend my sincere thanks to all other faculty members of Electronics and Communication Department and my friends for their support and encouragement.

ABDU RAHEEM ABDU RAZAK