A Critical appraisal of Digital Audio Broadcasting

Gurvinder S. Baicher and Ogbonnaya Anicho

Department of Engineering and Professional Studies
Newport Business School, Allt-Yr-Yn Avenue, Newport, South Wales NP20 5DA, UK
g.s.baicher@newport.ac.uk

***Abstract* — Audio Broadcasting describes the technology that uses digital modulation techniques for transmission on radio frequencies. The most commonly referenced form of digital radio is the digital audio broadcasting (DAB) although it also includes digital TV broadcasting, two-way wireless radio using digital signals and Radio Communications via the internet. This paper specifically examines digital audio broadcasting; its development, modulation, coding and time interleaving, synchronisation, frame structure etc. The paper also looks at the specific drawbacks of DAB especially in terms of reception quality in certain conditions and transmission costs. A comparison is drawn with the existing AM/FM broadcast. It will also consider developments in the new DAB+ format and draws a considered conclusion on the status quo and future of this form of audio broadcast nationally in UK and internationally.**

***Index Terms* — Digital Audio Broadcast, History of DAB, Eureka 147, COFDM, DAB Audio Codec**

I. Introduction

Digital Audio Broadcasting (DAB) is a digital technology for broadcasting audio programs in digital format. The use of radio broadcasting as an electronic medium for mass dissemination of information has been widely used and mainly based on the analogue AM and FM radio broadcasting standards. The AM and FM standards has over the century achieved both technological and operational maturity with about 2 billion receivers worldwide and thousands of radio stations offering services varying in speech and musical contents within and across continents [1]. However the breakthrough in digital technology techniques promised the potential of delivering better quality and more efficient services through an all-digital platform. The new Digital Audio Broadcasting technology was expected to broadcast signals in digital format delivering CD-quality sound, significant spectrum efficiency and a host of other value added services using terrestrial transmitters.

Development of Digital Audio Standards – Historical Perspective

There have been several attempts at developing a robust and widely accepted digital audio standard by different groups and organisations over time. An earlier implementation of digital audio broadcasting attempted the use of satellite delivery (10 to 12GHz band region). The Digital Satellite Radio (DSR) and Astral Digital Radio (ASR) are examples of Satellite implementations of audio radio but were not successful. Compression techniques used, and poor reception by mobile users affected the performance and acceptance of the systems [2], [3]. Further attempts at developing digital systems for radio broadcasting also resulted in the emergence of NICAM 728 (Near-Instantaneously Companded Audio Multiplex) system developed by the BBC. The technology was later implemented by the BBC for stereo television sound in the VHF/UHF band in the 80s [4].

It can be easily explained that no conceived digital system earlier mentioned could easily replace conventional systems especially for mobile reception in order to enable smooth transition to full digital platforms for audio broadcasting. The basic ITU-R recommendations (BS.774, currently BS.774-2) specified service requirements for digital sound broadcasting to vehicular, portable and fixed receivers using terrestrial transmitters in the VHF/UHF bands [5]. An acceptable digital system was expected to meet detailed ITU-R recommendations and commercial feasibility requirements.

Some difficulties already experienced with current FM and AM technologies ranging from spectrum efficiency, multipath reception, shadowing, passive echoes, mobile reception, modulation schemes etc. would be taken care of by this digital standard. Considering the enormous flexibility, robustness and performance indices of digital systems, industry watchers, analysts and many concerned parties theorized the expected benefits of digital audio broadcast. The eventual emergence of Eureka 147 DAB was considered a groundbreaking digital standard which was expected to change the history of digital audio broadcasting. However in retrospect, the expectations and current realities of the implementation of DAB reveal some disappointing discrepancies.

II. History of DAB

Eureka 147 DAB system

The development of a digital audio standard with appealing potentials began with a European consortium in 1987 as a project called 'Digital Audio Broadcasting' DAB also collectively known as Eureka 147 project [2], [6]. About 19 organisations from France, Germany, the Netherlands and the UK initiated this project though membership has since been extended well beyond those boundaries. The main aim of the project was to deliver a digital system with perfect mobile reception, CD quality audio, efficient frequency spectrum usage, low power transmission among others. Extensive research work was carried out to develop a digital audio system that will be widely accepted. Therefore decisions on schemes for transmission, audio coding and other key technical elements involved extensive tests, simulations and standardization. The Eureka initiative showed very strong level of standardization having been passed by the ITU-R and approved by ETSI. It is estimated to have more than 50 corresponding international standards and related documents to its credit. In 1999, Eureka 147 was merged with the promoting organisation known as World-DAB, (currently known as World-DMB forum). DAB gained very wide acceptance globally, though with expected glitches. There are several other digital audio protocols or systems available other than Eureka 147 DAB e.g. IBOC, IBAC etc. The acronym DAB is used both to identify the generic technology of digital audio broadcasting, and specific technical standards like Eureka 147. However, the Eureka 147 DAB is the widely accepted standard adopted for DAB implementation in Europe and beyond. This paper specifically focuses on the Eureka DAB which has been widely deployed in the UK. However the success of this technology in terms of operational and technical experience will be examined.

III. Overview of Analogue FM and AM

This section will critically assess the well established AM and FM systems used widely in broadcasting audio services since the 1920s. Also attention will be focused on major drawbacks of analogue FM and AM revealing technical issues which gave rise to the need of developing and implementing all-digital broadcasting standards [1].

AM – Noise and Interference Issues

AM systems operating at 530 – 1700 kHz are susceptible to interference caused by power system distribution, florescent lighting and many other unintentional radiators. It is also susceptible to adjacent channel or co-channel interference which affects its performance. This peculiar susceptibility is mainly due to the frequency of operation and inherent issues with amplitude modulation techniques [7]. For example, most of the noise is amplitude based and it is common knowledge that AM receivers are sensitive to it. However, the simplicity of AM receivers and coverage is a positive attribute from a broadcasting and business point of view.

FM – Multipath Propagation Issues

FM systems broadcasting at 88-108MHz frequencies are easily affected by multipath propagation and abrupt signal fading, degrading FM reception, especially to mobile receivers. For moving vehicles, the strength of the signal can vary dramatically thus leading to poor reception and quality of reception [1].

Spectrum and Bandwidth issues

Strictly speaking spectrum efficiency is a key criterion for measuring any transmission technology given the limited available bandwidth of the RF spectrum. The transmission of FM especially takes up a lot of bandwidth. A typical Wideband FM (WBFM) signal transmission requires 200 kHz bandwidth and a frequency reuse factor of about 15 to avoid cross talk. The prospect that DAB will be more spectrally efficient was a major selling point for the digital system.

Transmission Power

Analogue transmissions often require high power for transmission, although this is generally controlled by regulation. The ERP (effective radiated power) is usually specified; FM broadcast transmitters range from 10W to 50kW in output power. However, when compared with DAB, FM transmission power is quite low and actually its performance is a lot better in certain conditions; for example with DAB when the signal strength drops then the error correcting code of the older version is not efficient enough to effectively decode the signal.

Superior Audio of FM systems

Despite the many setbacks outlined above for AM and FM systems, one very key advantage which cannot be overlooked is the superior audio quality of WBFM systems. The superior audio of FM systems ensured that it captured the broadcast market especially in the music genre, which was quite a lucrative sector. DAB designers had in mind to surpass the benchmark set by FM systems, by pitching for CD quality audio as a minimum for the DAB system. However this vision was not realised as will be seen later in this paper.

IV. Analysis of Eureka 147 DAB

The key technical concepts of DAB are critically analyzed below, to further understand principles such as error coding, modulation and transmission methods used in implementing DAB.

Audio Coding

Audio coding involves reducing the bit rate of audio signals which can be achieved variously by removing the redundancy of the audio signal using statistical correlation or reducing irrelevancy through psycho-acoustic phenomena. The human ear has a hearing bandwidth of up to 20 kHz and a dynamic range of about 140dB, which must be matched with high quality audio signals for high fidelity experience. The spectral and temporal masking effect of the human auditory organ is exploited, as some sound events are masked by others [6]. This performance is necessary to achieve high fidelity necessary for good quality audio experience of DAB systems. The main audio modes supported by Eureka 147 are mono-mode (one-channel), stereo mode (two- channel), dual channel and joint stereo-mode.

DAB Audio Codec

For implementation of DAB, the MPEG Audio layer II (MP 2) coding technique was used. This coding technique technically requires a standard rate of 192 Kbps, a minimum for good sound quality [6]. The audio codec used in DAB could not really provide the quality and also support bandwidth consequently the quality of the audio did not match the near CD quality expected of the system. In a bid to implement more stations, broadcasters use a coding rate of 128 kbps which further degraded the quality of the audio. The tradeoff between sound quality and bandwidth became a major index for DAB implementation. It must be also stated that the decision of using 128 kbps in some countries happen to be the minimum stipulated by the regulatory authorities e.g. OfCom in the UK [8].

DAB Transmission Coding and Multiplexing

In DAB transmission, audio based and data based multimedia services are multiplexed into a single data stream known as an ensemble for transmission. This ensemble is frame based and has three main parts [6];

1. Synchronization Channel: Enables the receiver to synchronize and decode the received signal by supplying the transmission frequency and timing information.
2. Fast Information Channel: Informs the composition of the multiplex towards signal extraction and decoding information.
3. Main Service Channel: This can be described as the payload of the DAB signal as it contains the main data/video or audio service content.

Diagram Fig. 1 below shows the frame-based BBC DAB ensemble.

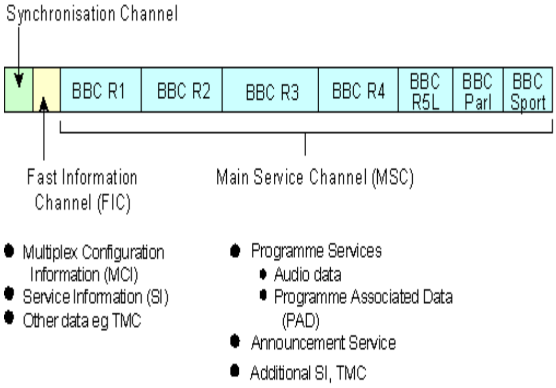

Fig. 1 DAB multiplex frame structure [6]

Frequency Management

The tradeoff between sound quality and bandwidth adversely affected the advantage of improved spectral efficiency of DAB. However, when quality is traded for bandwidth, DAB then has superior spectral efficiency when compared to AM or FM systems. It enables the combination of blocks of stations on a single 1.5MHz channel without interference. A block of at least six stations per country can be broadcast nationwide via a single DAB channel in conjunction with a single-frequency network. This enables more radio stations to be accommodated without congesting the radio waves unlike in FM systems. For instance, a typical FM transmitter has a bandwidth of 0.2MHz, with a spacing of 0.4MHz for nearby transmitters, which means a total of about 2.2MHz will be needed for a network covering a country [6], [8]. But a DAB network using a single frequency network will need about 1.5MHz for 10 services covering an entire country i.e. about 15 times more efficient than FM. The use of COFDM is a major reason for the frequency superiority and interference immunity of DAB systems.

COFDM

Coded Orthogonal Frequency Division Multiplexing (COFDM) is an orthogonal, multi-carrier system with divides the information to be transmitted into thousands of sub-carriers (Frequency division multiplexing) for transmission at lower bit rates [9].

Orthogonality

The concept of orthogonality is quite useful in DAB, as the need for excessive filtering is managed ingeniously. Mathematically, two signals are orthogonal if their dot product is zero, i.e. if the two signals are multiplied together and integrated over some time interval, if the result is zero then they are orthogonal over that time interval. This actually represents a demodulation operation of a carrier by multiplying it by a carrier of the same frequency [10].

Guard Intervals

In COFDM, the negative effect of multipath delays and impairment from echoes is eliminated by introducing Guard Intervals [9]. Mathematically, the integration period of the signals (at the receiver) should not span two symbols, the length of the guard interval should normally match the level of multipath expected. In DAB systems the effects of multipath is mitigated by the use of this technique, a good selling point for digital platform over analogue FM and AM. See guard interval described in the diagram in Fig. 2 below.

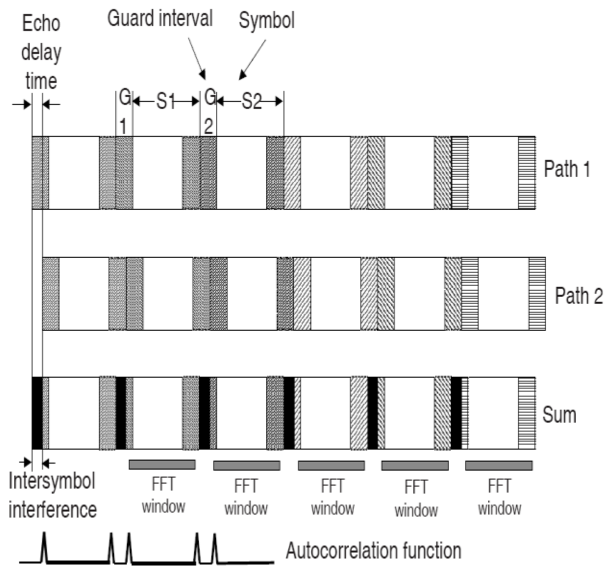

Fig. 2 Multipath Reception in COFDM [11]

DQPSK and π/4 DQPSK

Differential Quadrature Phase Shift Keying is a variation of QPSK modulation technique in which the symbol information is encoded as the phase change from one symbol period to the next (variation in phase) rather than as an absolute phase. At the receiver the phase changes has to be detected rather than the absolute value of the phase, which avoids the need for a synchronized local carrier [12], [13]. In DAB the variant of DQPSK used is the π/4 DQPSK, which simply implies that the phase transitions are limited to multiples of π/4 (i.e. ±π/4 or ± 3π/4). The π/4 DQPSK modulation format uses two QPSK constellations offset by 45 degrees (π/4 radians) and transitions occur from one constellation to the other. The sub-carriers generated from the COFDM operation are modulated using π/4 DQPSK. The use of π/4 DQPSK is preferred because it uses differential encoding which permits differential detection at the receiver, though coherent detection techniques perform better, the complexity that it attracts (especially under fading conditions) is a disadvantage and avoided in DAB systems.

Error Coding

The use of error coding enables the correction of all or some of the bits received. By ensuring that the errors are largely random it becomes easier to analyse statistically, this is readily achieved by spreading the transmitted signal across all carriers and interleaving it with time. DAB employs punctured convolutional coding with uses unequal error protection (UEP) which simply means that the bit streams more prone to error are provided with more protection. This practice technically introduces a grey area between periods of strong reception and no reception, a major flaw as listeners suddenly experience 'bubbling mud' sound [8], [10]. Convolutional forward error correction coding, combined with frequency and time interleaving are used to provide protection in DAB systems which use COFDM. However the new DAB standard (DAB+) uses a more superior coding which incorporates Reed Solomon error coding.

Convolutional Coding

In OFDM flat fading which results from using so many narrow band subcarriers is a major setback but this is taken care of in COFDM by the use of convolutional coding technique. Convolutional coding is a specific type of forward error correction (FEC) technique used to add redundancy to the signal transmitted. This technique enables the receiver to be able to correct any bits received in error [2], [10], [13]. The error correction decoder used in COFDM is the Viterbi algorithm, a typical soft decision approach. This is typically considered as combining convolutional coding and soft decision decoding, giving higher performance in terms of improved BER. COFDM's use in DAB enables added error detection capabilities.

The use of forward error correction coding is very important in order to deliver an acceptable bit error ratio (BER) at a significantly low signal-to-noise ratio (SNR) [9]. The process involves adding some carefully designed redundant channel coding information at the transmitter, which will provide the receiver with additional information and redundancy to assist in the decision-making process. The greater the additional information added the greater the resilience of the system (gigawave), which is achieved by using convolutional coding with soft decision decoding. By including the concept of channel-state information in the generation of soft decisions establishes the unique performance of COFDM in the presence of frequency-selective fading and interference [14].

Viterbi decoding

Convolutional encoding is implemented with viterbi decoding in channels were the transmitted signal is corrupted mainly by additive white gaussian noise (AWGN). Viterbi decoding is a method of recognising, at the receiver, the pseudorandom sequence added at the transmitter by the convolutional encoder. The viterbi decoder has the ability to recognize the distinctive pattern imposed on the data by the sequence even in the presence of errors and outputs a decision based on the best match found.

Hard Decision Decoding

In hard decision decoding the viterbi decoding algorithm just finds the best match to the incoming data stream with reference to the threshold voltage levels for either 1 or 0 [10]. It does not consider the effect of any interference or noise on the signal amplitude at the instance of decision-making.

Soft Decision Decoding

Soft decision coding adds a clever analysis for the confidence of the threshold matching process in the viterbi decoder [10], [13]. A lot of intelligence is added to the matching process, a kind of weighting function when analyzing the best match to the data. The incoming data is not just sliced into 1's and 0's, but is converted into a three-bit number representing its size (basically a three-bit ADC process).

Channel State Information (CSI)

The concept of CSI permits a level of confidence to be given to each of the multiple COFDM carriers. This enables the Viterbi decoder to use the CSI information to lower the soft-decision confidence levels for noisy carriers [10].

Time and Frequency Interleaving

Frequency interleaving is used when echoes are received with a rather shorter duration than normally expected which may put notches in the channel frequency response affecting a number of adjacent carriers. By spreading out or interleaving the carrier data with respect to frequency, FEC may be able to recover the data. In time interleaving as echoes get longer (i.e. in flat fades, Doppler shifts or short term complete loss of signal) then most or all the carriers will be affected for a period [10], [14]. However, if sequential data is spread over a number of carriers with respect to time, then FEC may well be able to recover the data.

Power Inefficiency of DAB

Another significant setback of DAB is that it has poor transmission power efficiency. This power inefficiency ultimately results in high transmission costs as a lot of power is required in the circuitry for converting digital encoded signal to the audio content [8]. This is in sharp contrast to analogue FM system where conversion cost is lower due to analogue audio. In terms of power, DAB is about 25-35% efficient, while FM is in the region of about 90%. Clearly a lot has to be done technically to reduce these overhead which is a key cost center for DAB implementation. Also, the DAB receivers require greater power for its operation due to the digital signal processing required to process the received digital signal to its analogue output format. This places greater overhead of power on the receiver thus leading to shorter battery life, as compared with FM receivers.

V. Summary of Challenges and Drawbacks

Audio Compression

After reviewing many materials, it is evident that the major technical setback of Eureka 147 DAB is the audio compression coding used in DAB.

The use of the MPEG 2 audio codec remains a major flaw which would have been easily avoided through more rigorous technical and management procedures. However DAB+ uses a superior audio coding scheme, MPEG-4 HE AAC v2, which performs better but how easy it will be for current investors in DAB to switch to DAB+ becomes a very key issue. The strategic and financial impact is indeed quite substantial to be overlooked.

DAB Receivers and compatibility to DAB+

Another issue widely discussed is the forward compatibility of DAB to DAB+, this setback could gradually subside, as new rollouts of DAB, actually implement DAB+, and the technology is still new, making it possible to phase out old DAB receivers with minimal impact.

Coverage

Though the coverage of DAB is growing, however when compared to FM and AM coverage, then the picture becomes clearer. Technically DAB is superior and quite robust but acceptance and implementation has been seriously hampered by bureaucratic issues and business interests.

DAB+

This is the latest version of DAB, rolled out in 2007 to address some technical glitches in the original DAB.

Some of the features added to DAB+ are;

- Latest MPEG-4 audio codec delivers exceptional performance efficiency.
- More stations can be broadcast on a multiplex, greater station choice for consumers available.
- Higher frequency efficiency of radio spectrum than with conventional DAB.
- Lower transmission costs for digital stations than with conventional DAB.
- New receivers are backwards compatible with existing MPEG Audio Layer II broadcasts, including scrolling text and multimedia services.

VI. Conclusion

The technical expectations and business prospects envisaged by DAB designers fell short of minimum acceptable targets. In the paper several set-backs and challenges of the original DAB technology was discussed. It is evident therefore, that the rollout and implementation of the DAB standard was at best technically deficient and at worst a failed business case. Considering the problems outlined above, the introduction of DAB+ addressed major concerns but the expectation of any acceptable digital standard must satisfy the minimum technical and business requirements. The lesson learnt in the challenges experienced by the DAB implementation should be used as a template to adopt a better standard. One recommendation will be to adopt a more global approach if the expectation of phasing out analogue AM and FM audio broadcast will ever be achieved. This global approach must therefore involve more cooperation from governments, broadcasters and all stake holders.

References

[1] F. Kozamernik, *Digital Audio Broadcasting – Radio now and for the future*, EBU Technical Review, Autumn 1995.

[2] W. Hoeg and T. Lauterbach T, (ed), *Digital Audio Broadcasting: Principles and Applications of DAB, DAB+ and DMB*, England: John Wiley & Sons Limited, 2009.

[3] R. Parekh, 2006, *Principles of Multi-Media*, New Delhi: Tata McGraw-Hill, 2006.

[4] A.J. Bower, "Nicam 728 – Digital two channel sound for terrestrial television," *BBC Research Report*, 1990.

[5] ITU, (online) http://www.itu.int/rec/R-REC-BS.774/en (18/11/2009).

[6] A. J. Bower, "Digital Radio - The Eureka 147 DAB System," *Electronic Engineering*, April 1998, pp. 55-56.

[7] J. Whitaker, *The Electronics Handbook*, Florida:CRC Press, 2005.

[8] Professional Broadcast, TV and STB Development Products & Test Solutions, (online) www.nyetec.com (19/06/2010)

[9] P. Shelswell, "COFDM: The modulation system for digital radio," *BBC Research Department Report*, 1996.

[10] R. Gustafsson, "Combating Inter-symbol Interference and Co-channel Interference in Wireless Communication Systems", *Blekinge Institute of Technology Licentiate Series* No. 2003:04.

[11] W. Fischer, *Digital Video and Audio Broadcasting: A Practical Engineering Guide*, Springer-Verlag Berlin Heidelberg, 2008

[12] K. Pahlavan and A. Levesque, *Wireless Information Networks*, New Jersey: John Wiley & Sons Limited, pg. 326, 2005.

[13] Signals and Systems Group, "The Gigawave Guide to Digital Microwave Technology: COFDM Compression" (online) http://www.sas.el.utwente. nl/ open/research (28/12/2009).

[14] J. Stott, "Explaining some of the magic of COFDM", *Proceedings of the 20th International Television Symposium*, Montreux, Switzerland, 13-17 June1997.