Introduction

DAB was adopted in many countries as early as 15 years ago, most notably in the United Kingdom where it has enjoyed great success. Most consumers liked the ease of use and the extra value added enhancements offered by DAB. Other than that, digital-only services made the decision to buy a DAB digital radio even easier.

In Australia, we decided to adopt DAB+, which is a similar technical framework for digital radio, that is the Eureka-147 transmission standard with one major difference to DAB in terms of the audio encoding algorithm.

DAB uses the MPEG1-LayerII algorithm which was cutting edge when DAB was first introduced. Over the years, many other psychoacoustic algorithms have cropped up over the years and arguably the best one is HE-AAC. DAB+ uses HE-AAC (versions 1 and 2). This allows for very efficient use of the spectrum, something that is very precious and is becoming increasingly rare. With DAB+ very high quality audio is achievable at impossibly low bitrates.

The following points form a technical summary of DAB+ along with comparisons with DAB, where relevant.

General DAB+ Information

Audio encoding

DAB+ differs largely from traditional DAB because of the audio encoding algorithm used.

DAB uses MPEG1-LayerII whereas DAB+ uses a modified HE-AAC (High Efficiency - Advanced Audio Coding) developed by Coding Technologies. HE-AAC v1 and v2 are toolboxes for AAC that allow for very high efficiency use of low bitrate data channels to encode high bandwidth audio. Among the key features of HE-AACv2 are:

Spectral Band Replication (SBR)

SBR can be selectively enabled in the DAB+ transmission to enhance audio at low bitrates. With SBR, the encoder allows for very high quality encoding of the lower end of a given audio spectrum (the actual cutoff frequency is determined by the DAB+ sub-channel bit-rate and its corresponding DAB+ payload after removing header overheads).

On the assumption that the higher frequencies of a given audio spectrum is usually a harmonic of the lower frequencies, the encoder only transmits the high quality low-frequency data with only the envelope of the high frequencies as side information (the envelop can be likened to a curve representing the volume of the higher frequencies in a given audio spectrum) to the decoder within the receiver. The decoder then faithfully reproduces the lower frequencies and synthesizes the higher frequencies based on how it would sound like if it were a harmonic. This high-frequency data is then rendered based on the volume levels transmitted as side information.

Parametric Stereo (PS)

Another tool in the HE-AACv2 toolbox is parametric stereo. This is used to further enhance the perceived quality of audio at very low bitrates.

Parametric stereo works by transmitting a mono audio signal with side information. This side information is then used by the decoder in the receiver to determine where individual digital audio samples should be panned (in discrete values to the left or right channel).

The resulting audio is a very good representation of the original audio fed into the encoder at often dramatically different bitrates compared to what would be required using an MPEG1-LayerII encoding algorithm as employed by DAB.

The graphics below indicate comparative audio qualities and show the corresponding bitrates required to achieve them

DAB+	DAB
Audio codec is aacPlus v2	Audio codec is MPEG 1 Layer II
“Better than FM quality” needs 56-96 Kbps	“Better than FM quality” needs 192-256 Kbps
"Similar to FM quality” needs 40-64 Kbps	“Similar to FM quality” needs 160-192 Kbps
“Acceptable, worse than FM quality” needs 24-48 Kbps	“Okay, but worse than FM” needs 128-160 Kbps

Note: The comparison above is very subjective and is based on stereo services

The nature of DAB and DAB+ allows for reconfiguration of bitrates and services on the fly with minimal impact on the listener. For example, a particular station could have its main service on one channel and throughout the day sporadically introduce a new secondary channel to play the days news highlights. This is done transparent to the receiver. The listener does not have to retune their radio as these reconfigurations are signaled in advance so that the receiver knows how to cope with them. At worst, there may be a very minor audio glitch of around a tenth of a second during a reconfiguration (if a channels bit-rate is decreased or increased). There theoretically should not be a glitch if other services are added or removed.

Data Services

One of the key advantages of DAB+ digital radio is the ability to have accompanying value-added data services along with the main audio services.

These can either be sent within the audio frames or in a separate channel altogether as a packetized or streaming data channel. Possibly the most exciting aspect of this is to be able to use the Multimedia Object Transfer (MOT) protocol to send objects with a registered mime type to receivers.

Data services within the audio frame can be configured to occupy a fixed bitrate within the audio payload or a variable one depending on the information being transmitted. As an example of this, a service can be configured to drop its data rate to 2 or 3 kbps when transmitting text or ASCII information but can be automatically ramped up to 32kbps when transmitting larger file objects such as images.

Among the more useful and widely accepted data applications that can be enabled on these services include:

Dynamic Label Segment (DLS)

It would be fair to say that almost all DAB and DAB+ radios will have a DLS display. This is a simple scrolling text display similar to RadioText we can see on RDS enabled FM radios. A maximum of 128 characters can be transmitted per text block with special character sets to clear the receiver screen and other operations such as new-line and carriage return (for receivers with multi line DLS displays). Information such as telephone numbers, website addresses, track-now-playing and announcer names are examples of what radio stations might decide to do with DLS.

MOT Slideshow

MOT Slideshow allows for the transmission of JPEG slides that can be selectively reconstructed by supporting receivers (those with screens and built-in MOT Slideshow decoding capabilities). A useful slide would be around 320 x 240 pixels and using a 32kbps data allowance in the audio payload, a 20kB JPEG can be transmitted within 9 to 15 seconds of being loaded onto the data multiplexer.

Some examples of slides that may be transmitted include product shots, studio webcams, weather maps, album cover information, stock market graphs, racing information and just about any information that can be conveyed as a simple graphic image.

MOT Broadcast Website (BWS)

The BWS application allows for the transmission of a series of HTML files (along with corresponding graphic images) to receivers that have a built in web browser. This creates a pseudo-interactive walled-garden web-browsing experience. If a receiver has a built in back-channel (for example, a GPRS enabled mobile phone with a DAB+ radio), it could access links that would take it to the outside world.

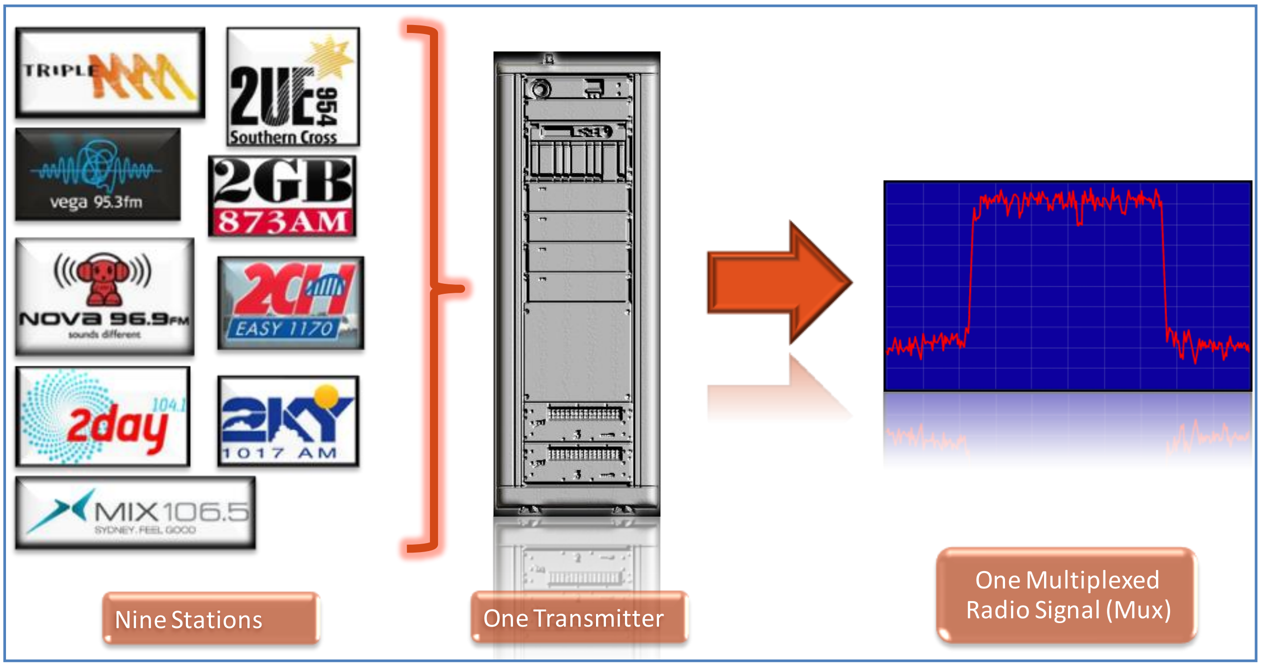
Electronic Programme Guide (EPG)

EPG's can be transmitted to supporting receivers. Apart from providing more information on the current and upcoming programs, digital radios that have recording capabilities can be scheduled to record based on the EPG information.

The DAB+ Signal

Multiplexing

Being in the digital domain, the DAB+ system allows for multiplexing of several services on one system as depicted in the simplified graphic below:



With the adaptation of DAB+ as the standard for Australia, up to 9 licence holders can be accommodated on one multiplexer unit.

Each licence holder could choose to multiplex their own audio and data in their studios using their own Service Multiplexer. The output of this service multiplexer (called the Service Transport Interface or STI) will then be collected at the transmitter site the Ensemble Multiplexer. The output of the Ensemble Multiplexer (called the Ensemble Transport Interface or ETI) is then fed to the transmitter as the programme input. The transmitter then processes this input and generates the radio frequency signal for final transmission.

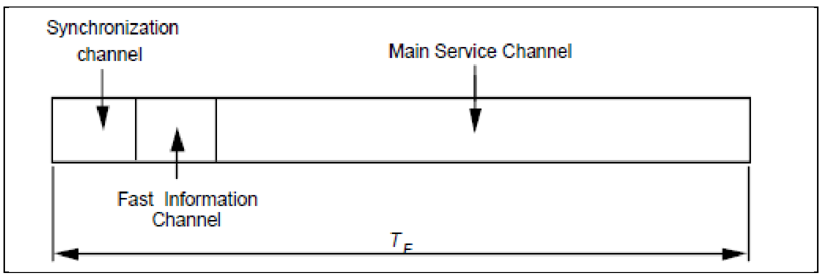
Transmission Method

The two basic methods for transporting data in a DAB+ system are the Fast Information Channel (FIC) and the Main Service Channel (MSC). In the actual transmission, the FIC and the MSC are preceded by a synchronization channel that tells the receiver to expect data in the next few frames. The synchronization channel can also carry transmitter identification information, useful for tuning single frequency networks.

The FIC carries information about what is contained in the MSC's that follow. The FIC contains a collection of Fast Information Groups (FIG's) that may contain the following data:

1. Multiplexer Configuration Information
2. Service Information
3. Fast Information Data Channels

The MSC is the main payload of a DAB+ service. Encoded audio information, packet data as well as auxiliary data services are carried in the MSC. Programme associated data (PAD) is also carried in the MSC either as X-PAD (within the audio superframe) or via a packet data service as X-PAD or N-PAD (non associated data)



Radio Frequency Characteristics

Frequencies used for DAB+

The list below identifies the DAB+ Channel frequencies used in VHF Band-III :

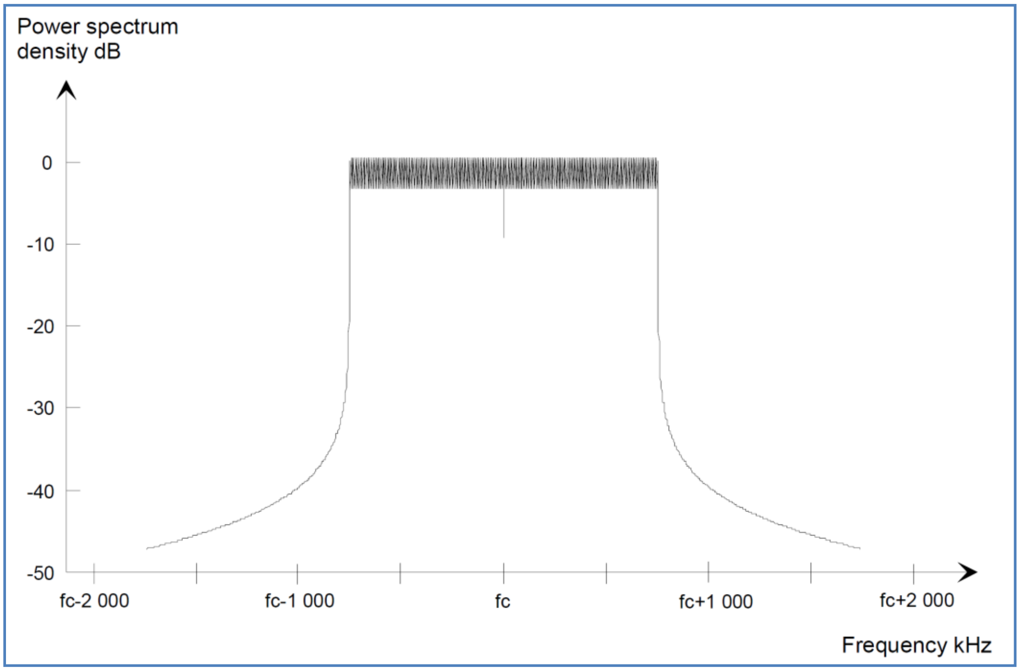
Channel	Center Frequency (f_c)	Channel	Center Frequency (f_c)	Channel	Center Frequency (f_c)
5A	174.928 MHz	**8C**	199.360 MHz	**11D**	222.064 MHz
5B	176.640 MHz	**8D**	201.072 MHz	**11N**	217.088 MHz
5C	178.352 MHz	**9A**	202.928 MHz	**12A**	223.936 MHz
5D	180.064 MHz	**9B**	204.640 MHz	**12B**	225.648 MHz
6A	181.936 MHz	**9C**	206.352 MHz	**12C**	227.360 MHz
6B	183.648 MHz	**9D**	208.064 MHz	**12D**	229.072 MHz
6C	185.360 MHz	**10A**	209.936 MHz	**12N**	224.096 MHz
6D	187.072 MHz	**10B**	211.648 MHz	**13A**	230.784 MHz
7A	188.928 MHz	**10C**	213.360 MHz	**13B**	232.496 MHz
7B	190.640 MHz	**10D**	215.072 MHz	**13C**	234.208 MHz
7C	192.352 MHz	**10N**	210.096 MHz	**13D**	235.776 MHz
7D	194.064 MHz	**11A**	216.928 MHz	**13E**	237.488 MHz
8A	195.936 MHz	**11B**	218.640 MHz	**13F**	239.200 MHz
8B	197.648 MHz	**11C**	220.352 MHz		

Note: The Koreans use a different set of Band-III frequencies.

The trials in Sydney are on DAB+ Channels 9A, 9B and 9C.

Spectrum characteristics

The DAB+ signal has a bandwidth of 1,536 MHz and can operate in one of four transmission modes. Looking at a "clean" DAB+ signal through a spectrum analyzer, this is what is what one would normally expect to see:



Modulation, Levels of Protection and Error Correction

Multi-Carrier Modulation Technique (COFDM)

The modulation technique employed in DAB+ is COFDM (Coded Orthogonal Frequency Division Multiplexing). The difference from conventional analogue modulation methods is that COFDM is a multi-carrier transmission method. Conventional FM is a single-carrier modulation method. One DAB+ signal (in transmission mode I) contains around 1,536 separate carriers with a carrier spacing of around 1KHz.

Because of this very close carrier spacing and the effects that real world propagation can have on these the DAB+ signal is heavily protected both from a radio frequency and byte-wise aspect to improve robustness and reduce reception errors. Among the methods used to protect the DAB signal are:

Energy dispersal and convolutional coding (Radio Frequency)

Energy dispersal is applied to the output signal to ensure regularity in the transmitted signal Convolutional coding is then applied to the signal to generate redundancy. This heightens robustness in less than desirable signal propagation conditions. With DAB+ it is recommended that the Equal Error Protection is applied to protect the entire MSC. This is generally because the DAB+ superframe sits in what used to be 5 DAB (MPEG1-LayerII) frames and equal protection needs to be across all sub-frames.

Cyclic Redundancy Checks (Byte-Wise)

CRC-16 is applied to the audio superframe to enable the receiver to calculate if the bits received are intact.

Reed-Solomon Encoding (Byte-Wise)

The end result is then put through Reed-Solomon encoding (transparent to the configuration or the size of the frames) to further strengthen and protect the bitstream.