DAB Radio: Digital Audio Broadcasting

If you go into an electrical store you will see more and more radios that are labelled as 'DAB' radios. But what are they and why have they suddenly become popular?

The driving force behind DAB is that broadcasters have to pay for the radio spectrum that they use. The radio spectrum is becoming very crowded with people wanting to use slices of it. Any technical development that allows the broadcaster to make more efficient use of their bit of spectrum is very useful.

Briefly, there are four major reasons for broadcasters switching over to DAB:

- New possibilities: it allows radio broadcasts to be accompanied by text, still pictures and service information of all types
- Will deliver to the listener high quality sound of almost ‘CD quality’
- Economic: DAB enables very efficient use to be made of an increasingly crowded radio spectrum
- Will not suffer from fading, interference and distortion associated with analogue broadcasts.

How does DAB work?

Before you read on you might like to know something about binary numbers. How we can change ordinary numbers like 15, 65, 21 into strings of ones and zeroes.

Think of a series of boxes:

128	64	32	16	8	4	2	1

Starting at the right hand end with the number 1, each box represents double the number of the one that went before it as we move along the row.

If there is a '1' placed in the box it means that number is included. If there is a '0' then it means that number is excluded.

For example:

The number 15 is 8+4+2+1. In binary this would be

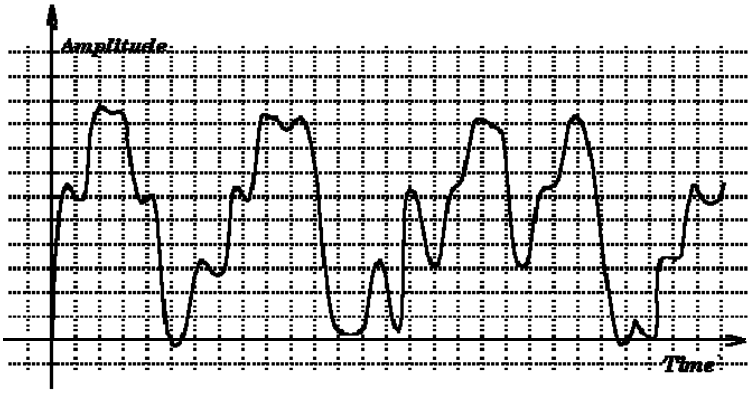
128	64	32	16	8	4	2	1
0	0	0	0	1	1	1	1

Which is 00001111 in binary.

The number 64 would be

128	64	32	16	8	4	2	1
0	1	0	0	0	0	0	0

Which is 01000000 in binary.



All types of sound broadcasting convert the sound into an electrical signal of some kind. The first stage in this process is what goes on in the microphone. The microphone transfers sound energy into a varying electrical voltage. This is an analogue signal (it wanders up and down in response to the sounds the microphone picks up). This signal in a conventional broadcast system is used to vary a 'carrier signal'.

The carrier signal is a wave that is transmitted through the air carrying the varying voltage signal with it. This is known as **modulation**. In conventional broadcasting the carrier wave is modulated by the varying voltage coming originally from the microphone.

Broadly there are two main ways of making the sound affect the carrier. The first is **Amplitude Modulation,** where the varying voltages coming from the microphone changes the amplitude of the radio carrier wave. In the same way that we can alter the brightness of a light bulb making it brighter or dimmer, we can change the 'brightness' of the carrier wave by mixing it with the electrical signal from the microphone.

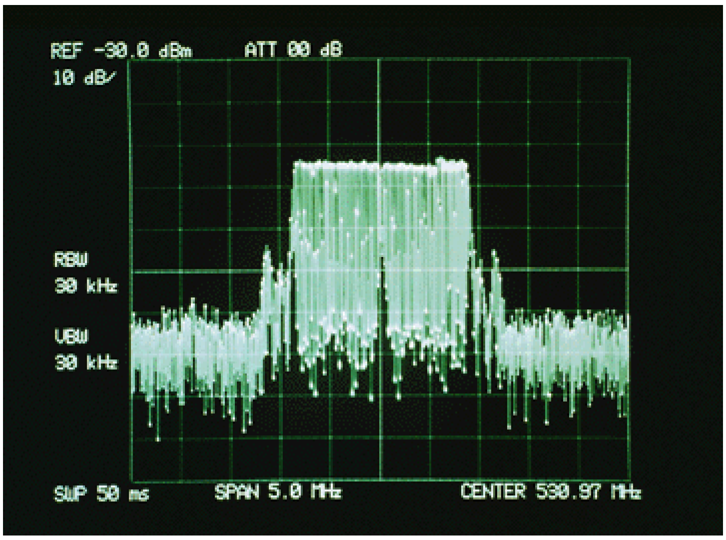
The receiver that the listener is using is tuned into the carrier and detects the changes in its 'brightness'. These changes are then converted back into sound electronically.

The second major method is **Frequency Modulation** where the microphone signal changes the frequency of the carrier about some fixed value. This can be thought of as changing the 'pitch' of the carrier wave. These 'pitch' changes are detected by the receiver and converted back into sound.

Both of these modulation techniques are capable of giving good results under ideal conditions. However, when conditions are somewhat less than ideal they are both subject to various kinds of reduction in quality

Digital Radio:

When we think about AM and FM broadcasts we tend to think of carrier waves arriving at the antenna of the receiver and then either its 'pitch' or 'brightness' being processed to recover the original sound. With a DAB broadcast we imagine a continuous stream of data in the form of pulses arriving at the radio receiver antenna. The spectrum of such a signal looks like this:-



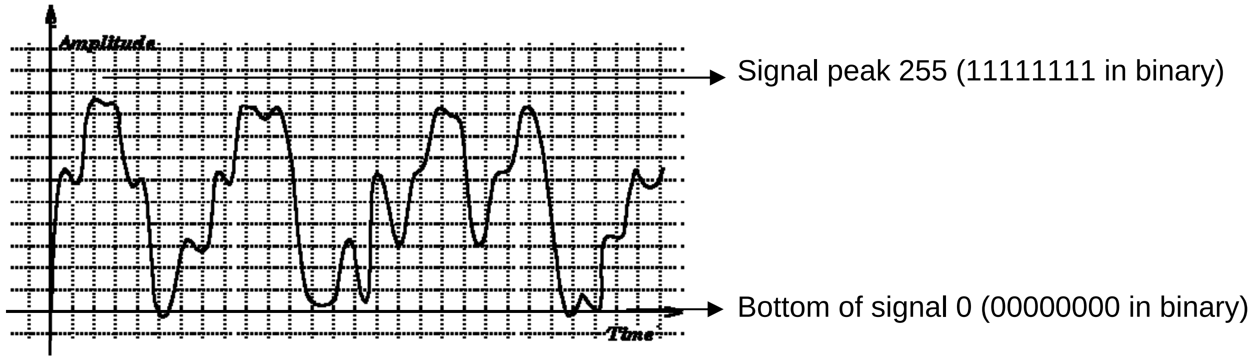
What is being received at the antenna of the receiver is a carrier wave that has been modulated by a continuous stream of zeroes and ones. These binary 'bits' contain all the information required to recreate the original sound within the receiver**.** Instead of altering the pitch or brightness of the carrier wave we can think of it being flashed on and off very rapidly.

The sharp peaks and troughs in the signal represent the zeroes and ones that are the digital sound and other information in the radio signal.

Key Fact:
By transmitting the information as a stream of binary bits it allows mathematical processing in the receiver to remove the effects of noise, fading, and other types of distortion caused by the signal reflecting off buildings and so on.

Making a Multiplex signal for transmitting sound:
The signal that comes from the radio transmitter and goes into the receiver is called a multiplex. A multiplex is a signal that contains different items of information all bundled up together.

The original sound that enters the microphone has to be converted into a stream of 0's and 1's. The electronic device that does this is called an analogue to digital converter or ADC. The varying voltage from the microphone is turned into a series of zeroes and ones by the ADC.
An overview of how the ADC works:



The diagram above shows the original voice signal. What comes out of the ADC is a stream of binary numbers or 'bits'. These are all zeroes and ones. No other numbers are possible.

However, there is a problem that has to be overcome. If we want to represent all of the sound coming from the microphone completely accurately then the amount of digital information we have to move around becomes so large and takes so much time that it is simply too much. Although it could be done, it would require such a large slice of expensive radio spectrum that it would not be worthwhile.

In order to achieve a relatively accurate sample of the original sound, certainly one that sounds of good quality, the sampling rate has to run at twice the highest frequency that the broadcaster wishes to transmit. If this is in the region of 20 kHz, and so the sampling rate is about 40 thousand samples per second.

This creates problems because if each sample was 8 bits in length, the broadcaster would have to find a way of processing and transmitting 320000 bits each second. Technically this is feasible but it would represent an undue burden on the system that was encoding and transmitting the information. It simply would be a waste of resources.

The solution to the problem is to only transmit those parts of the sound that the ear and brain are actually aware of. It is a fact that some of the sound the ear receives is ignored by the brain and so broadcasting it would have no effect on what we actually hear.

MUSICAM is an analytical piece of software that extracts all the sound from the original, takes away the sounds that the brain ignores leaving just that which causes a response in the listeners' ear.

This reduces the amount of data to be processed and transmitted: a large saving.

Transmitting the data:

Perhaps the greatest feature of digital sound transmission is that it can be mathematically processed to provide a channel of sound that is of high quality, resistant to the effects of fading, noise, and echoes, and capable of being received with virtually zero errors except under the most extreme of conditions. Digital signals are either excellent or totally absent, there is no room for poor quality.

How is this done?

There are two main techniques that are used by broadcasters; OFDM transmission and digital error correction.

Robust Modulation – OFDM

Orthogonal Frequency Domain Modulation (OFDM) is a more modern technique than AM or FM and is essentially where data is spread over of hundreds of ‘sub-carriers’.

- The stream of data coming from the transmitter instead of being placed onto a single carrier wave is spread across many carriers. In DAB there are 1536 of these ‘sub-carriers’ transmitted together, spaced just 1kHz apart from each other. The data is spread all across these sub-carriers so no single one bears all the information and each sub-carrier is modulated at a lower rate.
- This has a very important benefit to reception. Because an individual subcarrier is modulated at a low data rate (1 kilobit per second), the data symbol lengths are nearly a millisecond long giving plenty of time to accurately detect the data. Multipath reception, where unwanted reflected signals are received after a several microseconds delay compared to the wanted signal can therefore be easily countered.
- A further refinement is to transmit the data in blocks (called frames), but to spread the data bits of the frame over different carriers at different times – a process called time and frequency interleaving. This mitigates loss of data due to dropouts or impulse interference. By adding a short delay called the ‘Guard Interval’ between each data frame this adds additional resistance to multipath signals. It’s a bit like putting your fingers in your ears, and then suddenly unstopping them to listen, then immediately plugging your ears again so any echoes or reflections cannot be heard. Any data you hear is valid.
- This technique of spreading the data out over many carrier waves, so that echoes are ignored, also has another very important benefit. The right combination of guard interval and modest subcarrier datarates enable several nearby radio transmitters to use the same frequency as each other – effectively an artificial multipath situation. Because their data arrive at our receiver at slightly different times it’s the same situation as receiving an echo. The data from one transmitter is locked onto and decoded, and the receiver ignores the rest. Because every transmitter is using the same frequency it means that more radio spectrum is made available for other purposes. This feature permits the creation of Single Frequency Networks (SFNs) across a region or the whole the country, allowing mobile reception without having to retune.
- The amount of RF power (summed over several discreet transmitters) needed for a given coverage area can also be lower than a single very high power transmitter. A DAB multiplex can transmit just over a megabit per second, enough for several stereo audio streams in the same bandwidth that a single FM network would need, and with improved resistance to multipath for mobile reception. This makes DAB a very efficient way using of the available radio spectrum.

Digital Error Correction Coding

DAB uses Convolutional Encoding – ‘Coded OFDM’ (or COFDM for short). This makes the received data almost immune to error. Very briefly the data to be transmitted is coded with the addition of extra information to give a high degree of redundancy. Very simply it works this way:

A way of understanding this is to imagine we want to send the letter ‘A’ across a transmission system that is susceptible to noise and signal loss. If we do nothing the letter ‘A’ will simply be lost in the other noise that is in the radio spectrum or prone to noise that would cause it to be misinterpreted as say ‘W’. It’s a bit like trying to talk to someone at a party where there is a lot of noise.

In convolutional encoding we have the answer to the problem: Suppose the letter ‘A’ is coded as ‘ABC’. The letters ‘B’ and ‘C’ have been added to the data to make the message longer.

What is transmitted is ‘ABC’. Suppose the receiver simply detects ‘B’ and ‘C’ in the incoming signal. If it knows the rules (in this case the alphabet), It can add the A itself, thus restoring the lost data.

Even if it received just ‘C’ knowing that C is always preceded by A and B it can restore the data back to its original form. Clearly by adding more bits to the signal it makes the original data much more resistant to corruption. However, there is a trade off. The more data added to make the signal more resistant to corruption the longer it takes to process it. Thus the encoder has to make a decision on actually just how much extra data to add.

The rules that are used to code the data before transmission are ‘understood’ by the receiver and are decoded with great rapidity by a Viterbi Decoder. The Viterbi decoding system as well as knowing the rules is able to seek missing data by taking mathematical shortcuts that produces the ‘most likely’ results that are re -assembled back into the original sound.

All this is complicated but it produces a clear sound free from background noise particularly in radios that are found in cars. The car radio is now free from fading and 'fluttering' that can often spoil our listening.