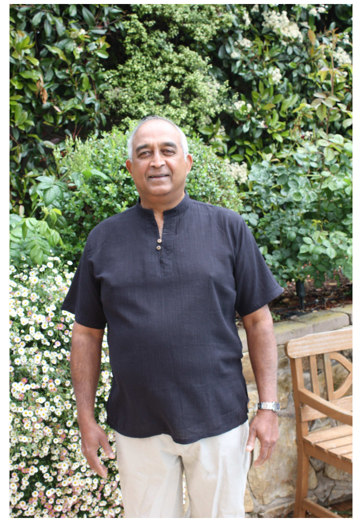
Digital Radio: How does it Work?



Shanthilal Nanayakkara
Principal Engineer, Radio Planning
Digital Transition Division Australian
Communications and Media Authority

Introduction

The ever-increasing computer processing power has opened up significant opportunities in radio communications. In particular, the ability to process audio signals in the digital domain in real time, has evolved new technological broadcast mindsets, which would improve the quality and clarity of broadcasting content while using the same analogue channel.

With consumers craving for increased clarity and diversity in radio content, many countries are venturing into this new mode of broadcasting to offer improved broadcast quality and increased content than would be possible in the analogue era.

Digital Radio has finally arrived, thanks to the ever-increasing real time processing power of the electronic chip.

What is digital Radio?

Digital radio is a digitized stream of radio signals used in transmission, which can carry increased program content, ie. additional information, using the same analogue bandwidth.

Concepts in Digital Radio

What do we hear

The human ear can generally hear sounds in the range 20 Hz to 20 kHz. But with age, the range decreases, particularly at the upper limit. Lower frequencies cannot easily be heard and this is why a sub-woofer is needed to amplify above the threshold of hearing.

The human hearing intensity range of sounds is substantial. The lower limit of audibility is extremely low, but the upper limit is very high and sound pressure levels above 80-100 could damage the eardrum.

The minimum threshold at which a sound can be heard is frequency dependent and, typically, the ear shows peak sensitivity (i.e. the lowest hearing threshold of 1 to 5kHz. The ability of the ear to discern frequencies is about 2 Hz in the middle frequency range. This means changes in pitch larger than the ear can perceive 2 Hz.

Humans can only perceive analogue information and speech needs to be continuous. Therefore, the radio receiver by nature needs to be provided with an analogue signal. Thus, the use of a digital stream should effectively be between the source and the receiver.

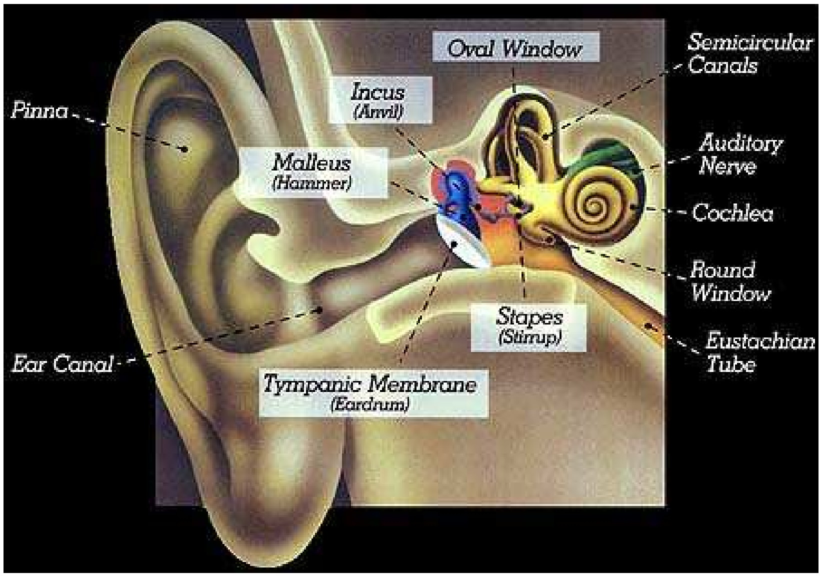


Figure 1:Human Ear

Masking Effects of theEar

In some situations, sounds that had previously been clear can be masked by another sound. For example, conversation at a bus stop, which was audible at first becomes impossible to hear, if a loud bus is driving past. This phenomenon is called ‘masking’. When two sounds occur at the same time the weaker sound is masked, if it cannot be heard in the presence of a louder sound.

A sound close in frequency to the louder sound is more easily masked than if it is far apart in frequency. This is known as frequency masking.

Similarly, a weak sound emitted soon after the end of a louder sound is masked by the louder sound. In fact, even a weak sound just before a louder sound can be masked by the louder sound.

Irrelevancy and Redundancy of Sounds

Unlike analogue radio, digital radio relies on the two concepts of irrelevancy and redundancy of the human ear. In the analogue domain, the electrical signals of either the speech or music were modulated in its entirety, whereas with digital, ONLY the information that can be heard by the human ear is sent to the receiver. As the total audio information is low compared to the analogue era, digital offers an opportunity for broadcasters to send supplementary information that is relevant such as text and pictures thus optimally utilizing the available information capacity of the channel.

The audio information that cannot be detected by the human ear (irrelevant) and sounds that cannot be heard due to frequency and temporal masking of the ear (redundancy) are not digitised. This process of stripping irrelevant and redundant information is carried out at the transmitting end with the aid of a reference Psycho-acoustic model.

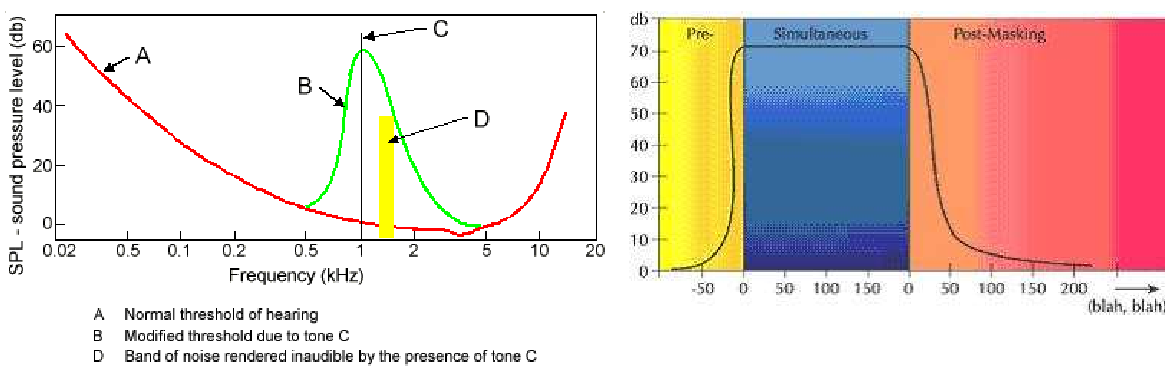


Figure 2. Threshold of hearing and Frequency and Temporal Masking of the Human Ear

The digitisation of sound is then accomplished by analysing audio signal spectrum over a very short period (24mS). This analysis is conducted in two parts. In its simplest form, only the lower part of the audio frequency spectrum between 20 HZ to 7 kHz is digitized from an original signal of 20 Hz to 20 kHz. The lower part of the spectrum is analysed by using a Psycho-acoustic Advance Audio Coder (AAC), which removes irrelevant and redundant information. As for the rest of the spectrum, only limited information is sent to the receiver, which is sufficient for the digital receiver to replicate that part of the spectrum in the receiver. The two parts of the received digitized spectrum are then combined in the receiver in the frequency domain, before it is decoded to analogue.

As the total data rate is significantly low due to spectrum band replication (SBR), the capacity of the channel would also be lower, allowing lower bandwidths for digitised transmission. The cut-off frequency is changeable to separate the AAC and SBR portions of the spectrum. On receipt of the digital signal, the receiver decodes the digitised version of AAC coding and the synthesized lower half spectrum. .

Capacity

The capacity of a radio frequency channel, be it a radio broadcasting or wireless broadband channel, is limited by laws of physics. In particular, Shannon's capacity theorem stipulates that the speed at which the data can be conveyed is depended on the bandwidth of the channel and the associated signal-to-noise ratio (or effectively the level of signal) of the transmission. The important parameter for the consumer here is the speed of data delivery. There are two ways of improving the data speed; higher bandwidth or higher signal to noise ratio or both. A good practical example is how consumers notice the reduction in speed of a wireless broadband channel when their wireless broadband Laptop is moved away from the primary coverage area of the service.

Due to the irrelevancy and redundancy removal from the sounds, digitisation of speech or music allows the effective capacity of the available radio channel to be broadened so that more information can be sent than previously possible. Because of the additional audio information, the clarity of the sound at the receiver would dramatically improve and the digital stream can also convey additional information other than audio. For example, the text and moving pictures associated with the sounds or music being played can now be sent as additional information, the latest novelty being moving pictures. At the recent International Broadcasting Conference in Amsterdam, Digital Radio Mondial (DRM) demonstrated video with radio.

Spectrum productive and Fade Free

Essentially, digital radio applications use coded-orthogonal frequency division multiplex (COFDM) modulation when transmitting digital signals. This technique of transmission not only provides spectrum productivity by allowing the same frequency to be used multiple times by way of single frequency networks, but also provides fade-free transmissions. The fading of the signal due to multipath is an inherent drawback in FM radio broadcasting.

Receivers

The coding and decoding of the digital signals are performed by signal processing at very high speeds, which cannot be noticed in real time. This high speed processing has only become possible with the advancement of signal processing power of the electronic chip. As digital radio processes signals at faster speeds using software modeling to mimic Psycho acoustic decoding, digital receivers essentially are software based with fast processing chips. With the advancement of processing technology, the cost of digital receivers is expected to fall dramatically. The catalyst for digital radio take-up could be the clarity and additional information that it can convey via text, data, pictures and slow moving video.