

Learning Objectives:

At the end of this topic you will be able to;

- Compare analogue and digital communications in terms of noise, attenuation, and distortion.

- State the function of Analogue to Digital Converters (ADCs) and Digital to Analogue Converters (DACs)

Noise, Attenuation and Distortion

In topics 4.3.2 & 4.3.3 we explained how analogue information can be transmitted directly by the amplitude or frequency modulation of a continuous carrier wave. In this topic we will investigate why the telecommunications industry is increasingly committed to the use of digital rather than analogue transmissions.

Before we begin looking at noise, attenuation and distortion, it is important to mention that many textbooks refer to a unit of measurement that is often used in this area of telecommunications work. This is almost certainly a unit you have heard of before, **the decibel**. It is a means of comparing two quantities on a logarithmic scale. The use of this unit is not required in this syllabus but some additional notes have been provided at the end of this section should you want to know more about this.

Attenuation

All carriers in all transmission paths suffer from attenuation; this happens regardless of the type of modulation being used or the nature of the information being carried.

Attenuation is the progressive power loss as a signal travels along a transmission path (e.g. a wire-pair, a coaxial cable, an optic fibre or the atmosphere). It should be clear that the attenuation increases as the length of the transmission path increases.

In a wire-pair or coaxial cable carrying electrical signals, the attenuation is partly caused by heat losses due to currents in the cable resistance and partly by radiation transmitted from the cables, which always occurs when the charges within them undergo rapid acceleration.

In an optic fibre carrying light signals, the attenuation is partly caused by absorption of the light energy owing to impurities in the fibre.

In radio-wave broadcasts, attenuation of the electromagnetic waves is caused by losses due to induced currents in the Earth and absorption in the atmosphere and ionosphere.

Noise

All modern telecommunication systems involve signals (i.e. voltages and currents) in electronic circuits. In every circuit unwanted electrical energy will be present; this will add itself to the information signal. The unwanted energy may be derived from the following sources:

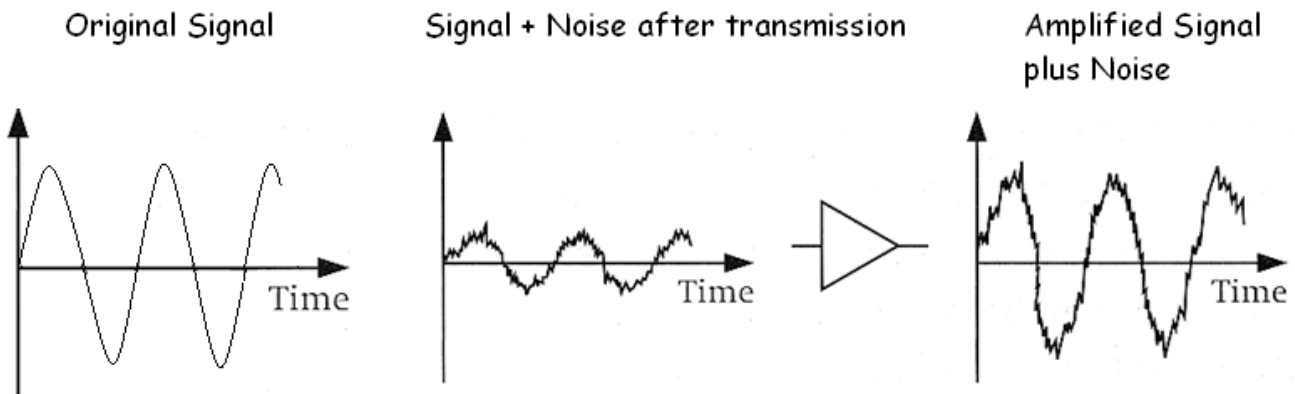
- White noise is present in all electrical circuits and is associated with the process of electrical conduction through the randomly vibrating atoms (their motion increases with temperature). White noise appears as random fluctuations in voltage or current. It tends to have a constant mean power level over a wide range of frequencies.
- Cross-talk occurs when a transmission picks up some of the power radiated from nearby transmissions (it is sometimes referred to as cross-linking).
- Interference occurs when a transmission picks up atmospheric or man-made radiation, e.g. from lightning or electrical machines or unsuppressed engines.

In telecommunications, the **total unwanted energy** or power added to a signal is generally referred to as **noise**.

The amount of noise on a signal is usually quoted as a ratio, known as the signal-to-noise ratio. The higher the ratio, the better is the quality of the information.

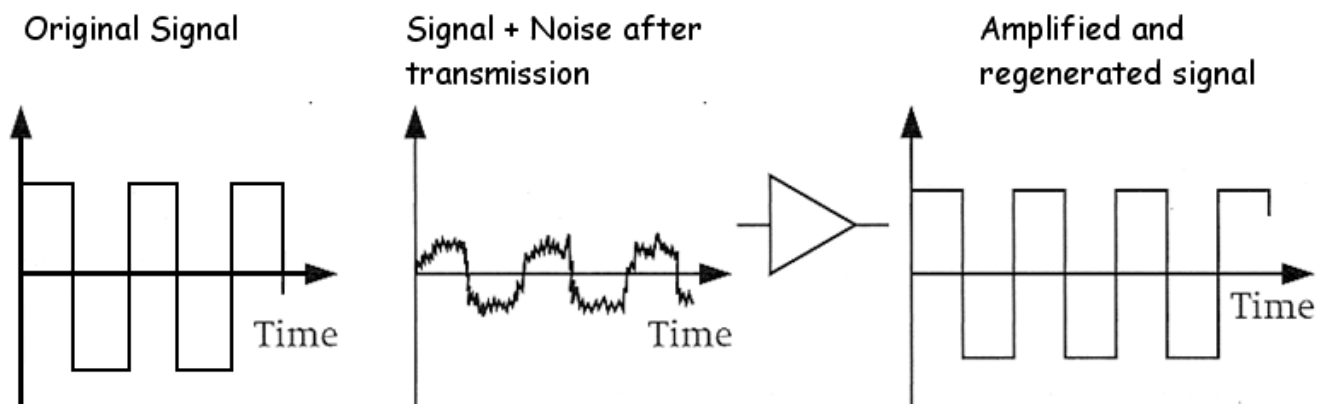
Amplification and noise:

If the signal in a cable is analogue (such as an AM carrier with TV information or telephone information), then the signals before and after amplification will resemble those shown below.



It can be seen that the noise has been amplified as well as the signal and this effect worsens with repeated amplifications. Some filtering can be used to minimise the cumulative effects of noise, but even so, this is a fundamental problem with long-distance analogue transmissions: they are inherently noisy.

If, however, the signal in the cable is digital (such as modern digitised telephone information) then the signals before and after amplification can be made to resemble those shown below.



The reason is that digital pulses always have a standard rectangular shape. Here it can be seen that the noise has been completely removed from the input signal which has thus been **regenerated**.

The amplifier circuit that enables this to happen is called a **Schmitt trigger**, which we will look at in detail in our next topic 4.5.2 - Regeneration. You may recall from your work in ET2 that this is a two-state switching circuit that has the interesting property that the voltage required to make it switch into a high state is not the same as the voltage required to make it switch into a low state.

This ability to regenerate an exact copy of the original signal illustrates one of the most important reasons for transmitting information in digital form: no matter how far it is transmitted, or how many amplifying stages it goes through:

Distortion:

Distortion is caused by the components in a communication system. Unlike noise which is random, distortion occurs because of the way in which circuit components respond to the input signal.

Dynamic Range

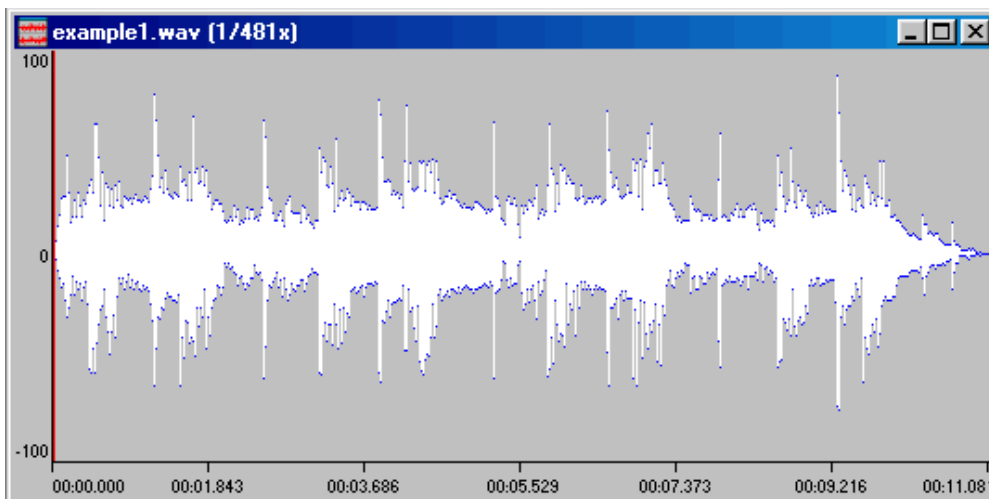
The dynamic range of any communication system is the maximum signal power that the various parts of the system can tolerate without distortion occurring. For example dynamic distortion could occur at

- i. the microphone if held too close to the mouth of a singer who is singing very loudly.
- ii. a signal level at the input to an amplifier being too high,
- iii. a amplifier providing a signal output of 100W power and feeding this signal into a 20W speaker system and hence working outside the design limit of the speaker system.

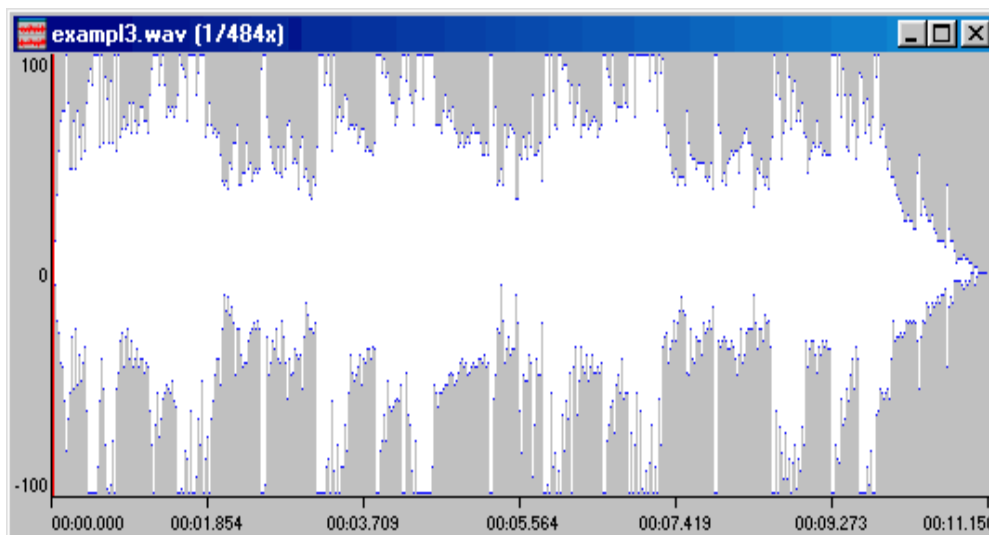
If an audio signal level is too high for a particular component to cope with, then parts of the signal will be lost. This results in a rasping distorted sound.

To illustrate this point, the pictures below represent a few seconds of music which has been recorded by a digital audio program. The maximum possible dynamic range (the range from quietest to loudest parts) of the signal is shown as 0 to ± 100 units

The following graphs show an audio signal recorded at two different gains. If you are linked to the internet you can listen to the sound by holding down the <ctrl> key and clicking on each graph in turn.



In this example, the amplitude (strength / height) of the signal falls comfortably within the ± 100 unit range. This is a well-recorded signal.



In this second example, the signal is amplified by 250%. In this case, the components can no longer accommodate the dynamic range, and the strongest portions of the signal are cut off. This is where distortion occurs.

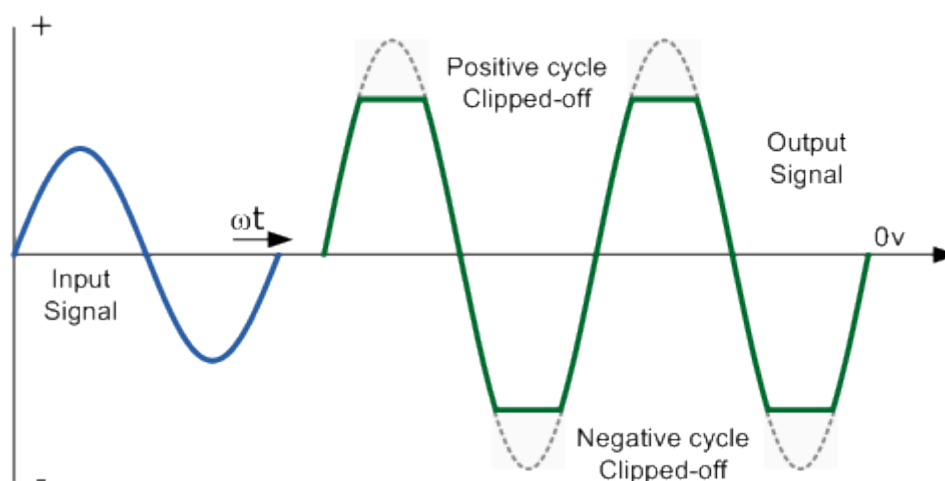
These examples can be used as an analogy for any audio signal. Imagine that the windows above represent a pathway through a component in a sound system, and the waves represent the signal travelling along the pathway. Once the component's maximum dynamic range is breached, you have distortion.

Distortion to some extent is therefore predictable, and some measures can be taken to alleviate its effects.

In particular during this course we are interested in two forms of distortion, clipping and crossover distortion.

i. Clipping Distortion.

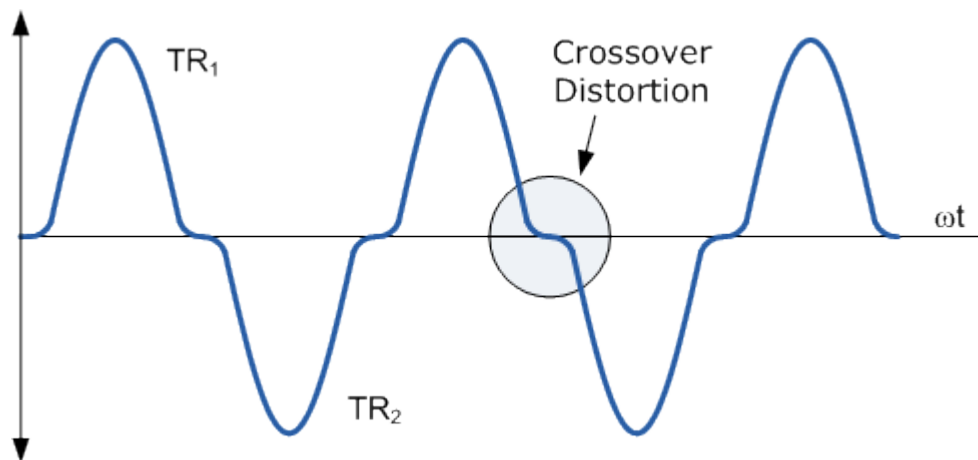
One of the most common types of distortion is clipping distortion which we first discussed in ET1. The following diagram should remind you of what is meant by clipping distortion.



It occurs when the gain of an amplifier is too high for the input signal, which causes the amplifier to produce a saturated output at the extremes of the power supply.

ii. Crossover Distortion.

A second possible source of distortion occurs with basic transistor amplifiers known as push-pull amplifiers - you will meet these in Module ET5. The problem arises when the signal causes the switch over between transistors in the amplifier leading to a flattened part of the output graph around the 0V line.



Whilst distortion is highly undesirable, it can only be eliminated by very careful design of all circuits through which the signal is to pass. Every effort is made to ensure that in broadcast systems the level of distortion is kept to a minimum, but that does not prevent the user from turning the volume up too high and causing distortion by amplifying the signal too much.

Distortion occurs with both digital and analogue signals, however the effect is most noticeable on analogue signals, as the information in an analogue signal is contained in the amplitude of the wave.

The advantages of digital communication

Although the principles of digital communication were developed in 1937 by Alex Reeves, it was not until the 1970s that the telecommunications industry began the move to digital. There were three reasons for this delay.

- Digital circuitry was much more complex than analogue circuitry.
- Early digital circuitry was expensive and unreliable (and difficult to synchronise).
- Transmission of a signal in digital form consumes a greater bandwidth than would be required to transmit in the original analogue.

The change was mainly initiated by the development of digital integrated circuits and, in particular, the microprocessor by Intel in 1971. When the electronics industry began to mass-produce low-cost, but sophisticated, chips for use in digital signal transfer, the switch to digital became inevitable.

The advantages of digital signal transfer are as follows.

- In a similar way to FM the signal information is located in the cross over between logic 1 and logic 0, and is not affected by amplitude changes.
- Digital signals can be perfectly regenerated.
- In long-distance transmission, noise does not accumulate on the signal with repeated amplifications.
- Extra codes can be added to a digital signal to check for errors in transmission. (See Topic 4.5.5 - Asynchronous Transmission.)

Summary of implications of Attenuation, Noise and Distortion.

- All signals in all transmission paths suffer from attenuation.
- Attenuation causes the power in the signal to become less and less.
- Noise is any unwanted energy in a signal and is present in all electronic systems.
- The long-distance transmission of a signal involves repeated amplifications, to avoid losing the signal in the noise.
- All transmission systems have a minimum signal-to-noise ratio.
- Digital signal transfer is a superior process to analogue signal transfer.
- Noise can be removed from digital signals.
- Digital circuits are no longer expensive, lend themselves to computer control and allow a very large number of different signals in a transmission channel.

Analogue - Digital - Analogue Conversion.

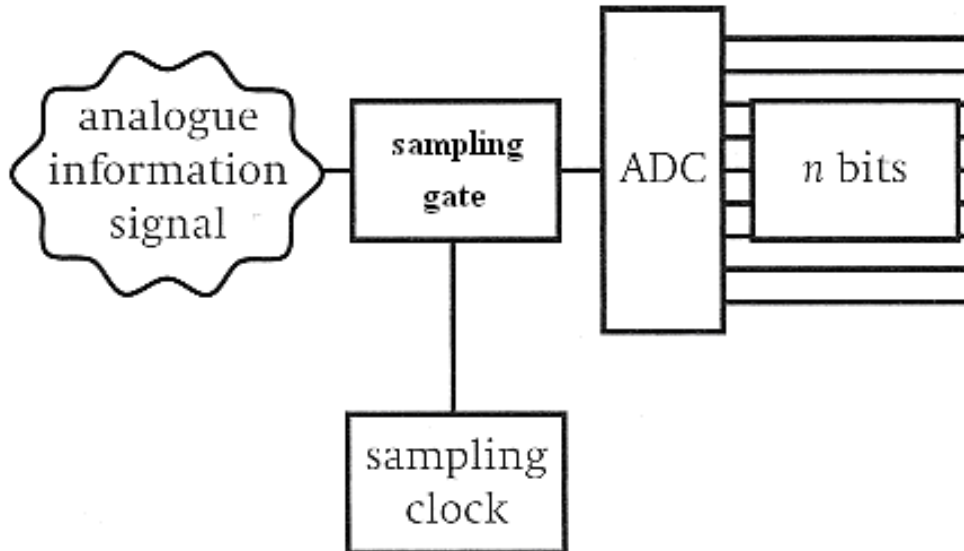
Throughout this topic we have sung the praises of digital communication systems and their superiority to analogue systems. We must remember that as far as speech, music and television are concerned, the original signals are in fact analogue, and that to make a loudspeaker reproduce the original sound then it must be provided with an analogue signal. This poses a simple question, if we start and end with analogue, where does the digital fit in? Clearly there must be some form of transformation from analogue to digital and back to analogue, but where does this take place?

Because of all the benefits that digital technology brings, signals are converted into digital at the earliest possible moment, and then converted back into analogue at the very last moment before passing on to the power amplifier for delivery to a loudspeaker.

Two new and vital subsystems are therefore required for this to take place, and you will deal with the exact construction and design issues of these in Module ET5. For now we will just consider their function, and treat them very much as black boxes, as we do not have time here to go into the construction of these now.

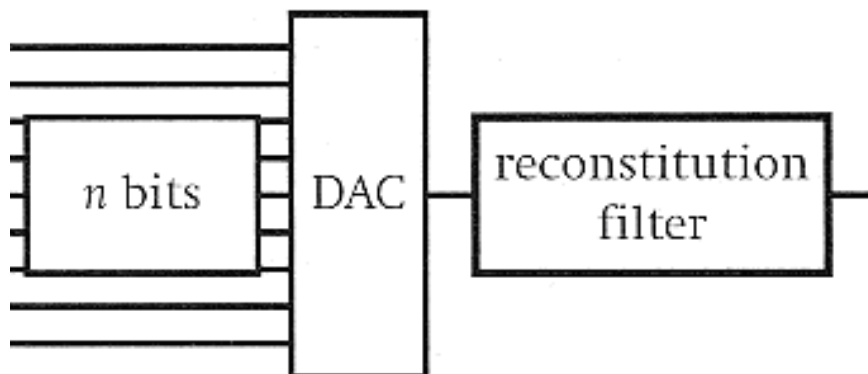
The first subsystem we need is an Analogue to Digital Converter or ADC. As its name implies the purpose of this subsystem is to change the analogue signal into a digital one. This is achieved by taking a sample of the analogue waveform, and then converting this into an n-bit digital code, which could be as low as 2-bits (but this would be a very poor system indeed) or as many as 32-bits (which would provide excellent quality) as we will see when we look at their function in more detail in Topic 4.5.4 Pulse Code Modulation.

The following block diagram sums up the function of the Analogue to digital converter.



The system effectively operates when the sampling clock gives out a high signal; the instantaneous value of the analogue input voltage is taken by the sampling gate. The ADC then converts this into the appropriate n-bit digital signal ready for transmission.




It should now be very clear that at the receiving end there is a need for the reverse process to happen, a second device performing a Digital to Analogue (DAC) conversion is required. The following block diagram shows how this works.



The incoming digital signal is fed into the DAC, which contains a special amplifier called a summing amplifier which translates the digital code at the input into an analogue voltage. The filter helps to smooth the changes

between values every time the DAC outputs a new voltage. As was mentioned earlier the exact construction of these devices will be covered in Module ET5, and we will discuss the operation of these in more detail in Topic 4.3.4 Pulse Code Modulation, but this is sufficient information for now.

Self Evaluation Review

Learning Objectives	<i>My personal review of these objectives:</i>		
			
Compare analogue and digital communications in terms of noise, attenuation, and distortion.			
State the function of Analogue to Digital converters (ADCs) and Digital to Analogue converters (DACs)			

Targets: 1.

 2.

For the Enthusiast

The decibel.

This unit is extremely useful when comparing physical quantities that have a huge range in magnitude. Your ear, for example, can detect a sound intensity (i.e. a power per unit area) that can vary from about 10^{-12} Wm^2 (your threshold of hearing) to about 10 Wm^2 (when you feel pain). So, instead of quantifying a sound intensity by its absolute magnitude (by stating, for example, that a concert generated 0.2 Wm^2 at the back of the hall) we often compare the sound intensity with our threshold of hearing to get a measure of how much louder one sound is than the other. So, how could you compare 0.2 Wm^2 (the concert) with 10^{-12} Wm^2 (the quietest sound that can be heard)? If you subtracted the two intensities, you would obtain $0.199\,999\,999\,999 \text{ Wm}^2$, which is clearly too tedious to contemplate. If you divided the two intensities, you would obtain 2×10^{11} which is better but is still an extremely large number to contemplate.

The decibel (dB) makes working with such large comparisons easy. The **decibel** comparison of two quantities P_1 and P_2 is defined as 10 times the logarithm of their ratio: number of decibels (dB) = $10 \log \left(\frac{P_1}{P_2} \right)$

We can now compare the sound intensity P_1 produced by the concert with the threshold of hearing P_2 :

$$\text{number of dB} = 10 \log \left(\frac{P_1}{P_2} \right) = 10 \log \left(\frac{0.2}{10^{-12}} \right) = 10 \log (2 \times 10^{11}) = 113 \text{ dB}$$

and we have arrived at a number that is easy to comprehend.

Worked Examples

- As you listen to a normal conversation, the sound intensity at your ear is measured to be $4.5 \times 10^{-7} \text{ Wm}^2$. Calculate the decibel comparison with the threshold intensity of hearing (10^{-12} Wm^2).

So for this question $P_1=4.5 \times 10^{-7} \text{ Wm}^2$, and $P_2=10^{-12} \text{ Wm}^2$.

$$\text{Number of dB} = 10 \log \left(\frac{P_1}{P_2} \right) = 10 \log \left(\frac{4.5 \times 10^{-7}}{10^{-12}} \right) = 10 \log(450000) = 57 \text{ dB}$$

- Calculate the Power of sound intensity at your ear if the comparison to the threshold of hearing is 16dB.

So for this question $P_1 = ? \text{ Wm}^2$, and $P_2=10^{-12} \text{ Wm}^2$, the number of dB=16

$$\text{Number of dB} = 10 \log \left(\frac{P_1}{P_2} \right)$$

$$16 = 10 \log \left(\frac{P_1}{10^{-12}} \right)$$

$$\frac{16}{10} = \log \left(\frac{P_1}{10^{-12}} \right)$$

$$1.6 = \log \left(\frac{P_1}{10^{-12}} \right)$$

$$10^{1.6} = \frac{P_1}{10^{-12}}$$

$$39.81 = \frac{P_1}{10^{-12}}$$

$$P_1 = 39.81 \times 10^{-12} = 3.981 \times 10^{-11} \approx 4 \times 10^{-11} \text{ Wm}^2$$

Although the decibel was originally devised for sound comparisons, it is now used extensively in telecommunications to compare two power values (where power = energy/time).

The important point to remember is that the decibel is not an absolute unit. It is 10 times the logarithm of the power *ratio*. Thus it is meaningless to make a statement such as 'the signal power is 70 dB', unless it is understood that it is 70 dB above or below a particular reference power level.

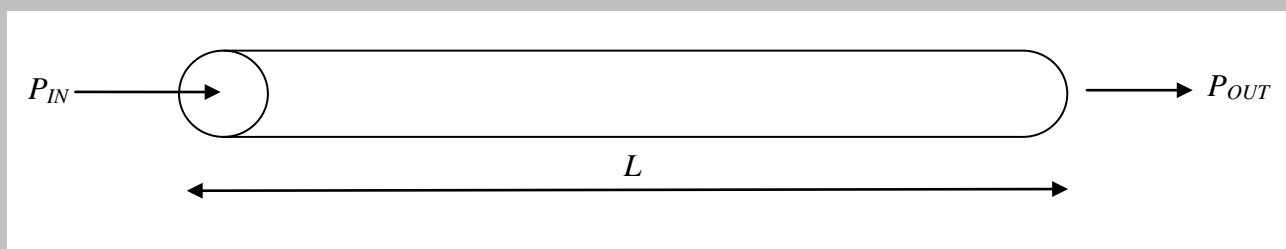
In telecommunications, this reference level is commonly taken to be 1 mW. For example, when the power of a sound is 10^4 greater than threshold, this is 40 dB.

The decibel in telecommunications

There are three important areas of application for the decibel in telecommunications. The decibel makes calculations of signal power and noise power less tedious than they would otherwise be.

- Attenuation

The diagram below shows a cable (it could be electrical or optic-fibre) of length L , into which is applied a signal power P_{IN} and out of which emerges a smaller signal power P_{OUT} .



The total attenuation of a signal in this cable can be expressed in decibels as follows:

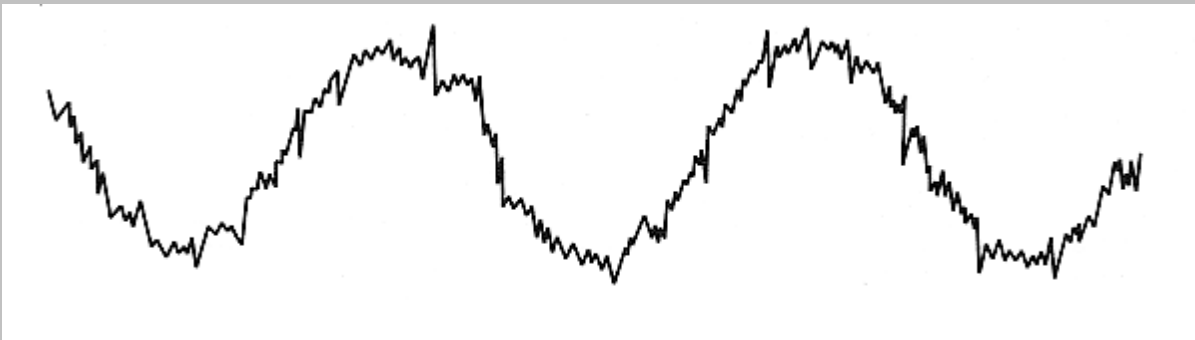
$$\text{total attenuation of signal} = 10 \log \frac{P_{OUT}}{P_{IN}}$$

Note that this will be negative, since P_{OUT} will be smaller than P_{IN} . The characteristic signal attenuation of the cable is usually expressed in decibels per kilometre as follows:

$$\text{signal attenuation per kilometre} = \frac{10 \log \frac{P_{OUT}}{P_{IN}}}{L}$$

Again, the value will be negative.

- Signal-to-noise ratio

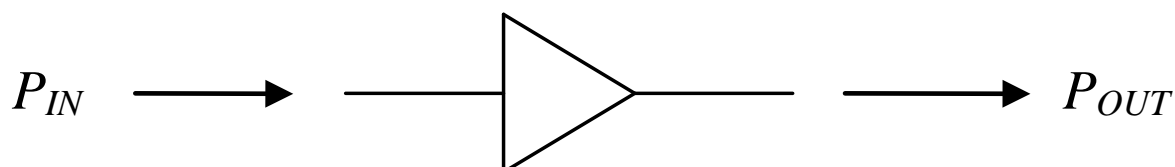


The diagram above shows a signal of power P_{sig} on which there is superimposed noise of power P_{noise} . The signal-to-noise ratio is normally expressed in decibels as follows:

$$\text{signal-to-noise ratio} = 10 \log \left(\frac{P_{sig}}{P_{noise}} \right)$$

This should be positive!

- Amplifier power gain

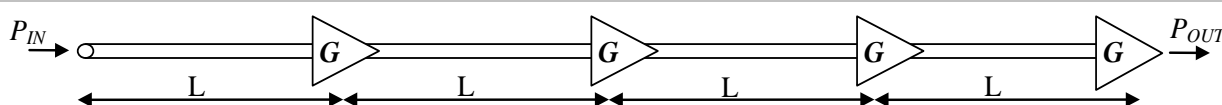


The diagram above shows an amplifier which accepts a small input signal of power P_{IN} and which outputs a larger signal of power P_{OUT} . The power gain of this amplifier is normally expressed in decibels as follows:

$$\text{power gain} = 10 \log \left(\frac{P_{OUT}}{P_{IN}} \right)$$

Again, the value is expected to be positive.

To apply these definitions, consider the transmission path of a typical signal, as shown below.



Here, a signal of power P_{IN} is applied to a cable of length L and attenuation $-A$ dB/km. After travelling a distance L , the signal emerging from the cable is amplified by an amplifier of power gain G dB. Then it is transmitted a further distance L before being amplified again by a similar amplifier. This is repeated over a transmission distance of $4L$ as shown. Now for a few questions - don't worry the answers are provided!

1. What is the total attenuation of the signal caused by the transmission cable?

The answer is : $-(A \times 4L)$ dB; note that it is negative.

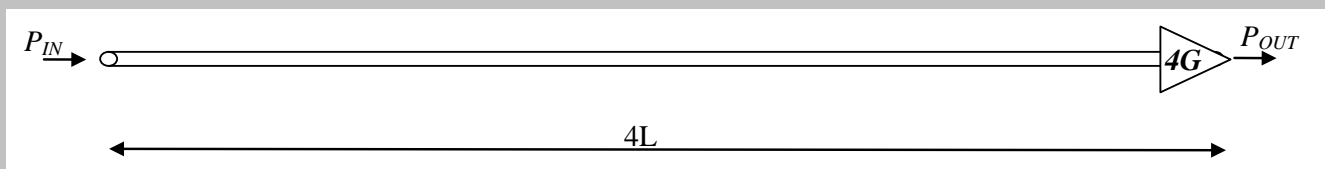
2. What would be the power gain of the signal if it passed through the four amplifiers without any attenuation along the line?

The answer is : $4G$ dB; it will be positive.

3. What is the net change in signal power after this transmission, taking attenuation into account?

It can be shown that, in terms of decibel level, the competing effects of amplifier gain and attenuation are simply subtracted, so that the net change in signal power level is equal to the gain decibels minus the attenuation decibels, $(4G - 4AL)$ dB. Note that if the gain just compensates for the loss (i.e. $4G = 4AL$), then the total change in signal power will be 0 dB. This does not mean a zero signal but that the final power out is the same as the original power in.

4. Why is the transmission cable broken up into four sections, each ending with an amplifier? Why can we not transmit the signal uninterrupted for the whole distance $4L$ and then have an amplifier of gain $4G$ at the end of it (such a system is shown below)?



The reason is as follows. As the signal travels along the cable it suffers attenuation. Furthermore, in the transmission path there is a noise power that remains more or less constant. Consequently, if the signal is left to travel without interruption, it would eventually become swamped by and lost in the noise. Except for unusual and sophisticated modulation systems, it would then not be possible to recover the signal from the noise. Breaking up the transmission system into sections allows the amplifiers to lift the signal well above the noise level.

In all transmission systems, it is essential to avoid letting the signal-to-noise ratio become too small otherwise the signal will become lost in the noise.

The amplifiers with which long-distance cables are broken up are called repeaters. The 13000km under-sea coaxial cable that links Sydney in Australia and Vancouver in Canada is broken up with 1000 repeaters, i.e. there is an amplifier every 13 kilometres.

Worked examples

1. A signal of 500mW is input to a cable of attenuation 4dB/km. Calculate the power of the output signal P_{OUT} , when it emerges after a travelling through 12km of this cable.

Solution:

The total attenuation will be : $-4 \times 12dB = -48dB$

Therefore

$$\begin{aligned}
 \text{attenuation} &= 10 \log \left(\frac{P_{OUT}}{P_{IN}} \right) \\
 -48 &= 10 \log \left(\frac{P_{OUT}}{500 \times 10^{-3}} \right) \\
 -4.8 &= \log \left(\frac{P_{OUT}}{500 \times 10^{-3}} \right) \\
 10^{-4.8} &= \frac{P_{OUT}}{500 \times 10^{-3}} \\
 1.58 \times 10^{-5} &= \frac{P_{OUT}}{500 \times 10^{-3}} \\
 P_{OUT} &= 1.58 \times 10^{-5} \times 500 \times 10^{-3} \\
 P_{OUT} &= 7.9 \times 10^{-6} = 7.9 \mu W
 \end{aligned}$$

2. The signal entering an amplifier has a power of $3.6 \mu W$ and the signal leaving the amplifier is $0.48 W$. Calculate the power gain in decibels of the amplifier.

Solution:

$$\begin{aligned}
 \text{power gain} &= 10 \log \left(\frac{P_{OUT}}{P_{IN}} \right) \\
 &= 10 \log \left(\frac{0.48}{3.6 \times 10^{-6}} \right) = 10 \log(133333.33) = 51.24 dB
 \end{aligned}$$

3. The signal emerging from a coaxial cable has a power of 63mW. If the signal-to-noise ratio is 22dB calculate the noise power P_{noise} , in the cable.

Solution: The signal-to-noise ratio = 22dB

Therefore

$$\text{signal} - \text{to} - \text{noise} = 10 \log \left(\frac{P_{\text{signal}}}{P_{\text{noise}}} \right)$$

$$22 = 10 \log \left(\frac{63 \times 10^{-3}}{P_{\text{noise}}} \right)$$

$$2.2 = \log \left(\frac{63 \times 10^{-3}}{P_{\text{noise}}} \right)$$

$$10^{2.2} = \frac{63 \times 10^{-3}}{P_{\text{noise}}}$$

$$P_{\text{noise}} = \frac{63 \times 10^{-3}}{10^{2.2}}$$

$$P_{\text{noise}} = \frac{63 \times 10^{-3}}{158.49} = 3.98 \times 10^{-4} = 398 \mu\text{W}$$

Student Exercise 1

1. A 25 mW signal enters a cable system of length 150 km. The cable has an attenuation of 7 dB km⁻¹. Amplifiers of gain 36dB are located at 5 km intervals. Calculate:

a. the total power loss of the signal as a result of travelling through the 150 km cable;

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b. the total signal gain as a result of passing through the 30 amplifiers in the system;

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c. the signal power emerging from the thirtieth amplifier, at the end of the system.

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Solutions to Student Exercise

1. a. the total power loss = $150 \times 7 = 1050$ dB
- b. number of amplifiers = $150 \div 5 = 30$

the total signal gain = $30 \times 36 = 1080$ dB

- c. the signal power emerging from the thirtieth amplifier, at the end of the system.

Overall gain = $1080 - 1050 = 30$ dB

$$Gain = 10 \log \left(\frac{P_{out}}{P_{in}} \right)$$

$$30 = 10 \log \left(\frac{P_{out}}{25 \times 10^{-3}} \right)$$

$$3 = \log \left(\frac{P_{out}}{25 \times 10^{-3}} \right)$$

$$10^3 = \frac{P_{out}}{25 \times 10^{-3}}$$

$$P_{out} = 10^3 \times 25 \times 10^{-3}$$

$$P_{out} = 25W$$