PHYSICAL SENSORS FOR ENVIRONMENTAL SIGNALS

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Master Degree in Artificial Intelligence for Science and Technology (AI4ST)

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OUTLINE OF THE COURSE

- Lecture 1: Introduction to environmental signals and physical sensors
- Lab 1: Introduction to digital signal processing and instruments for measurements
- Lecture 2: Vibrations: sources and detection
- Lab 2: Characterisation of an acoustic system
- Lecture 3: Distance, position and speed measurement
- Lab 3: Measuring distance with ultrasounds and speed with an accelerometer
- Lecture 4: Electromagnetic radiation: sources and detection
- Lab 4: Detecting and generating light

SENSING THE ENVIRONMENT

SENSING THE ENVIRONMENT

Sources

- Temperature
- Pressure
- Distance and position
- Speed
- Vibrations
- Acoustic
- Radiations: particles & light
- Chemical pollutants

SENSORS AND SIGNAL PROCESSING

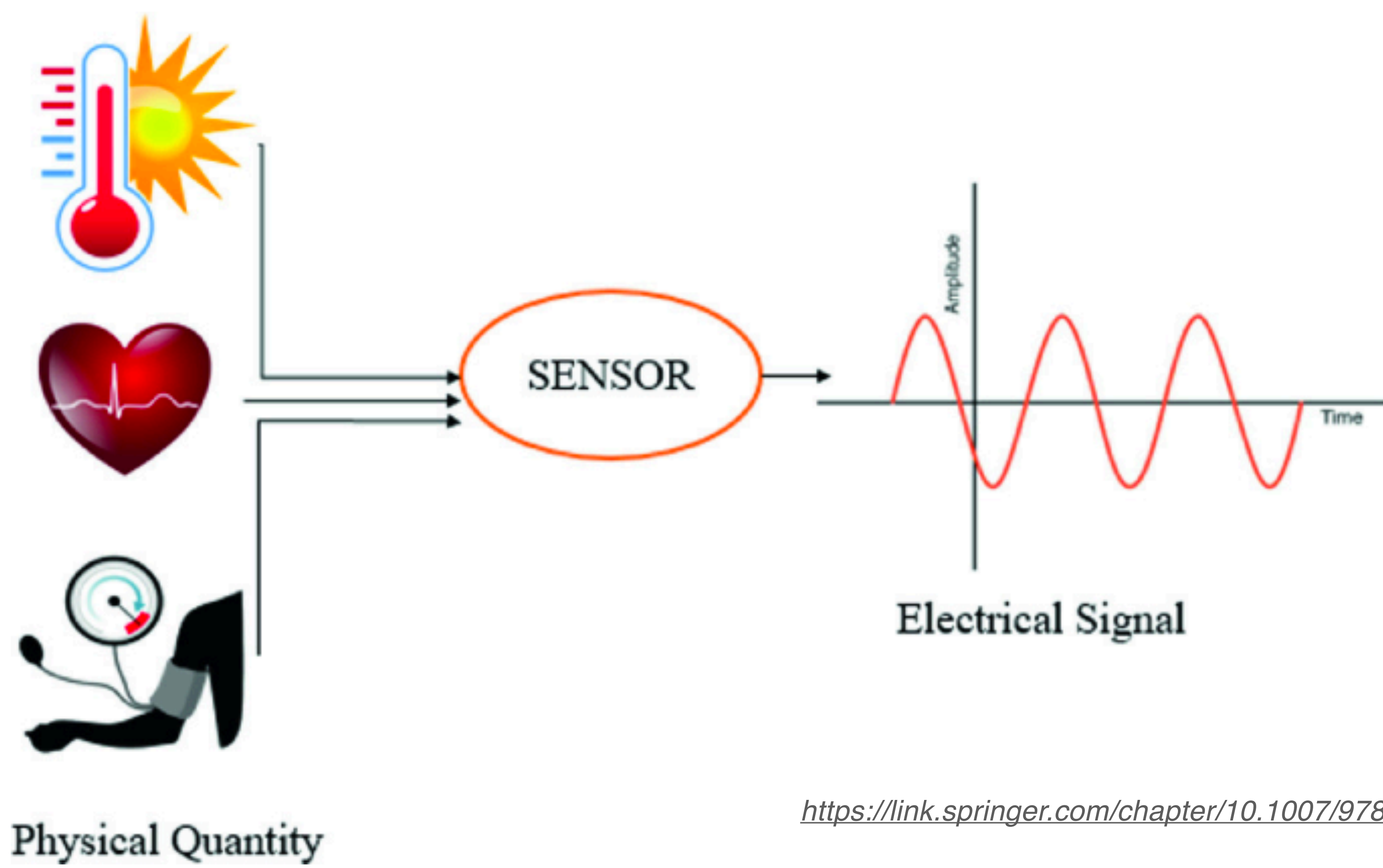
THE GENERAL PRINCIPLE OF A SENSOR

External source $\xrightarrow{\text{Probe}}$ Sensor $\longrightarrow$ Output signal

Physical quantity: temperature, position, speed, pressure,…

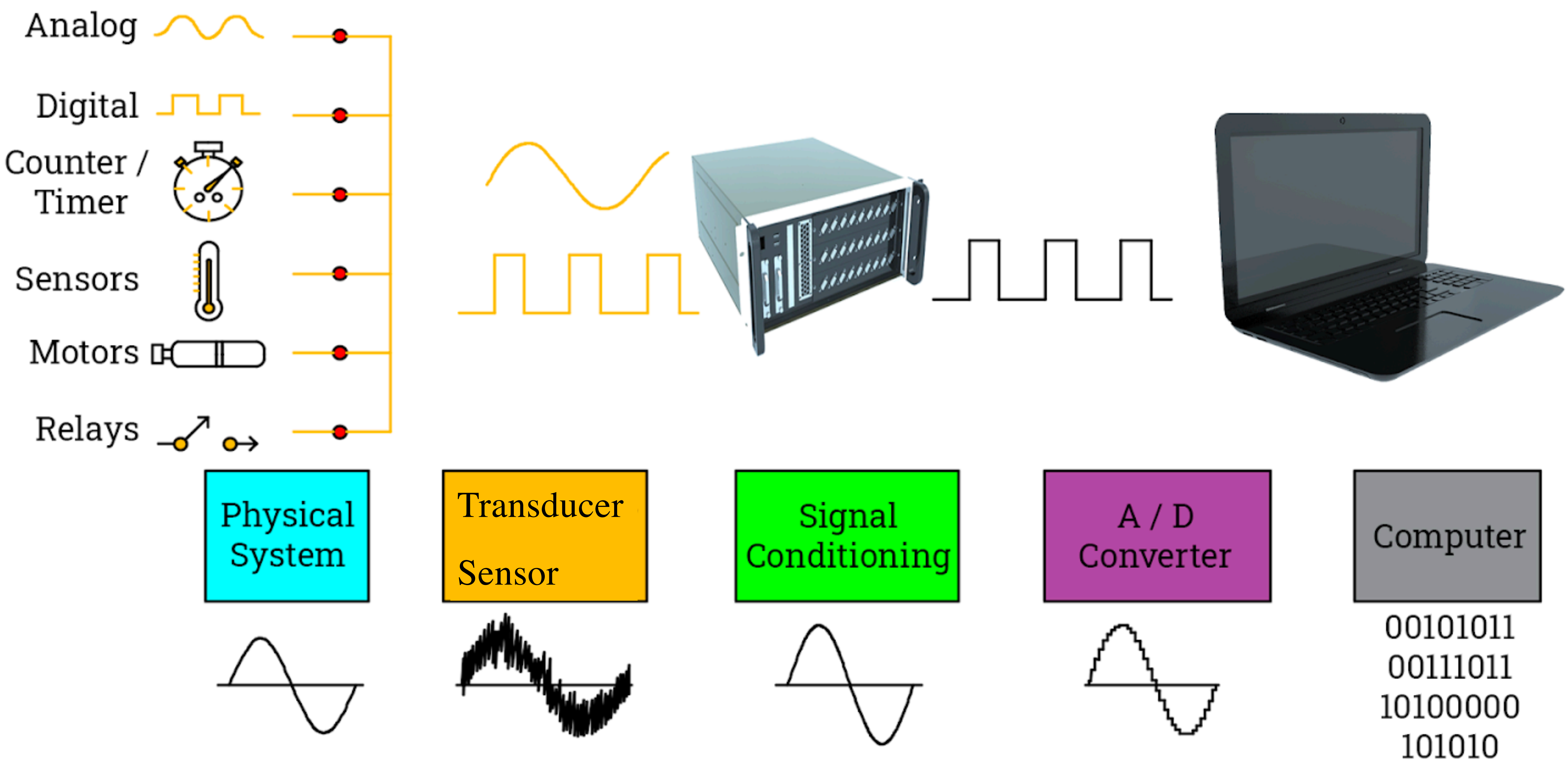
Sensing the physics phenomena and converting that into an electrical form

Electrical (voltage/ current) output



https://link.springer.com/chapter/10.1007/978-981-19-1408-9_17

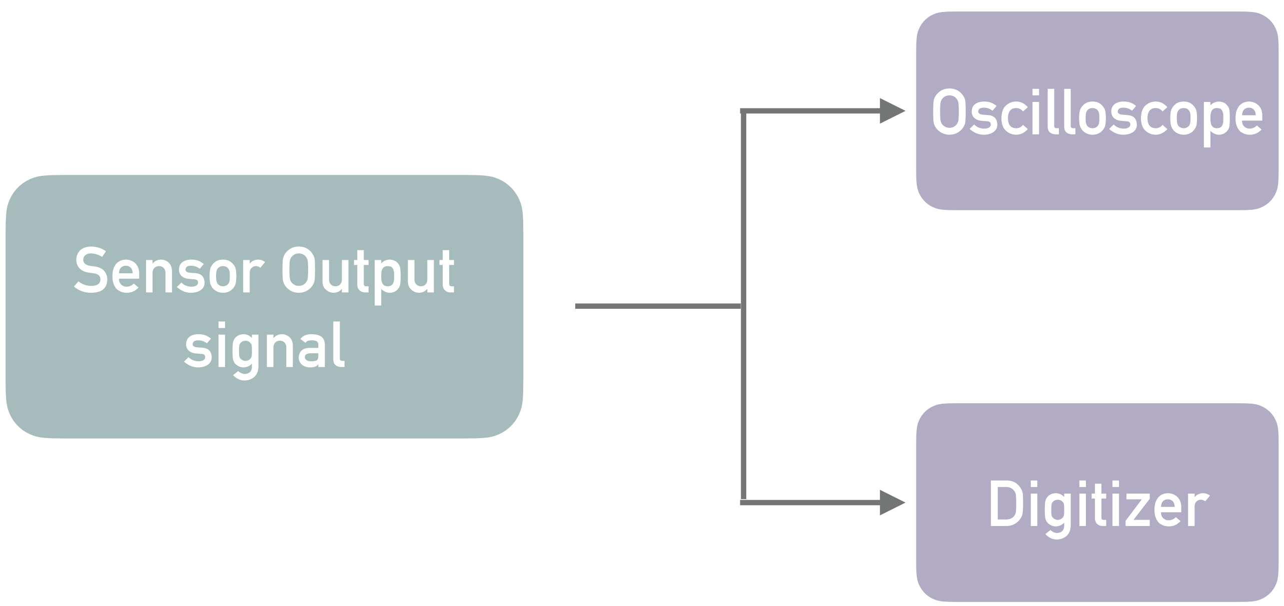
SENSOR DATA ACQUISITION CHAIN

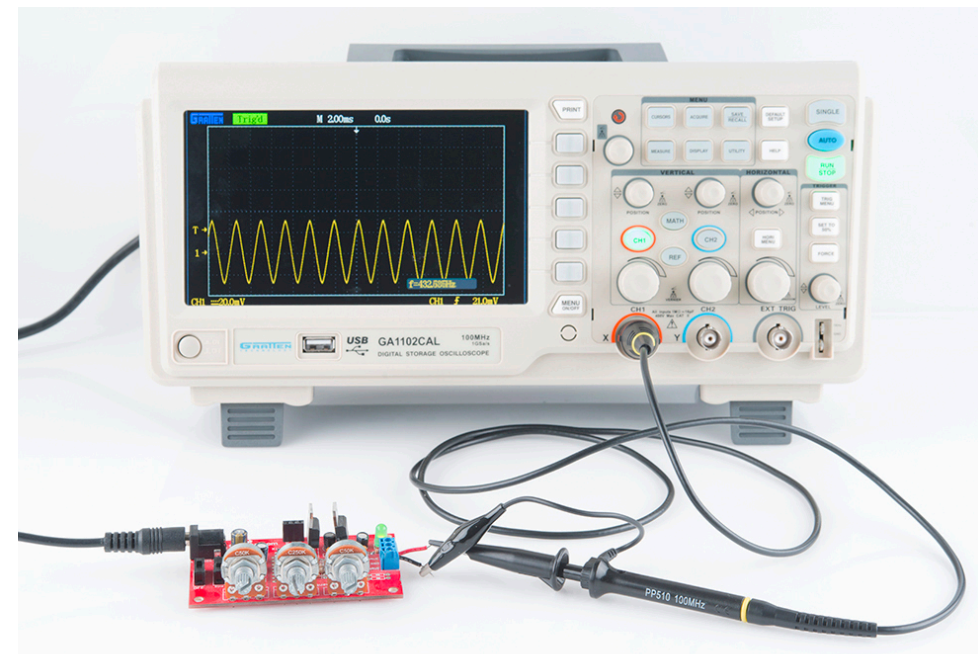


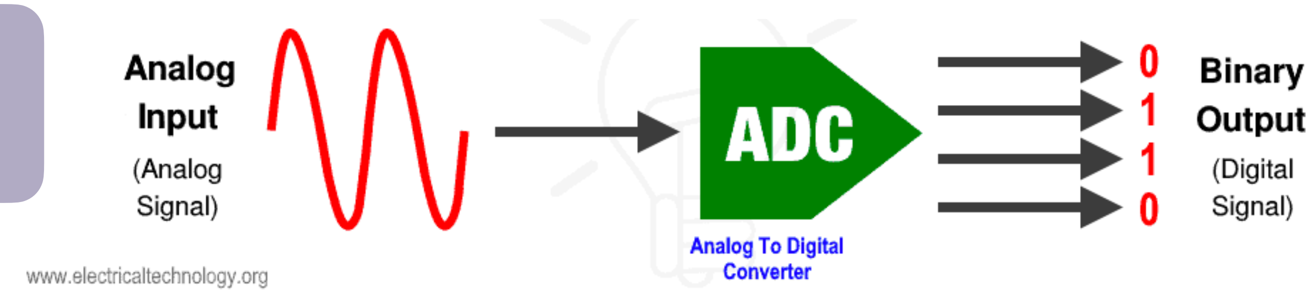
SENSOR DATA ACQUISITION CHAIN

Acquisition of the sensor output:

- Digitisation of the electrical signal
- Storage/display of the waveform
- Digital signal processing



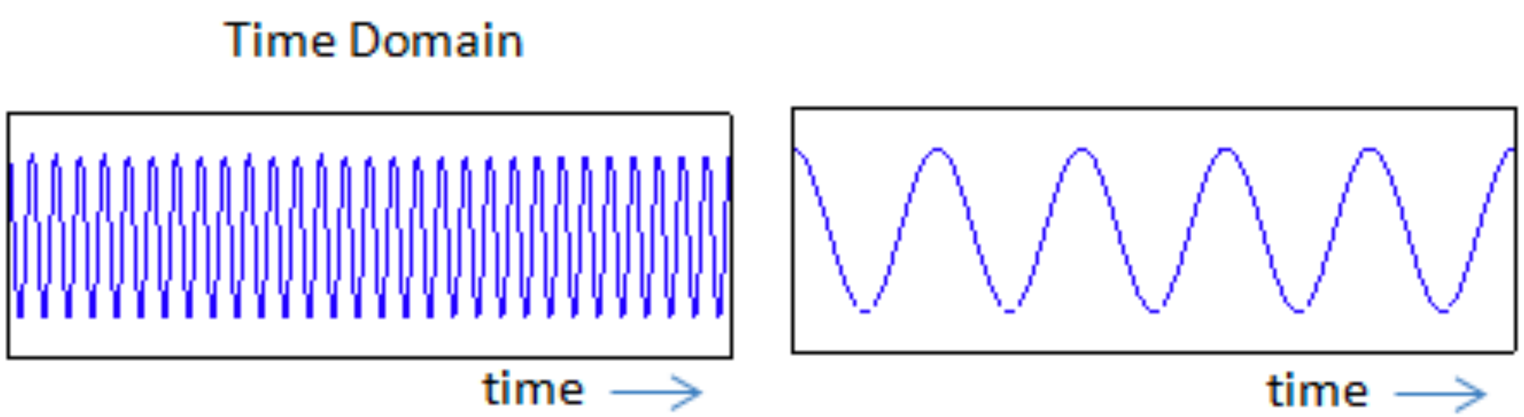




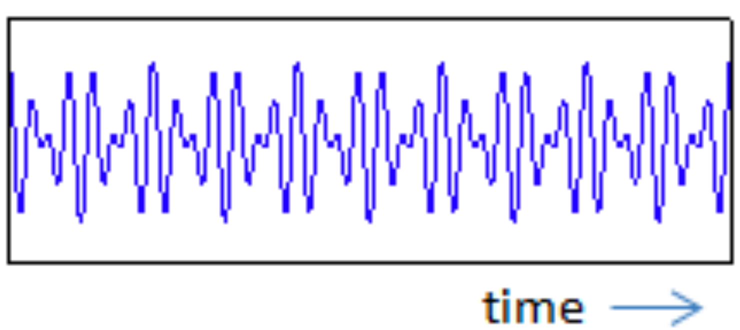
SIGNALS

Signals in the time domain

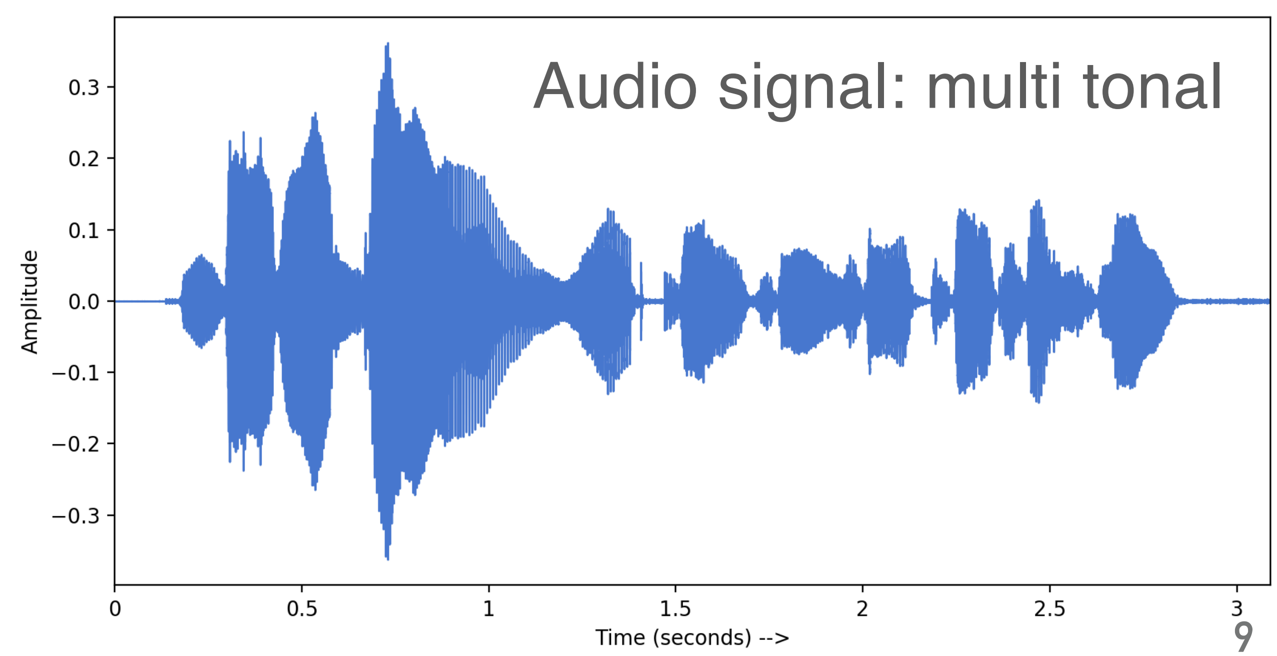
The time domain is simply the graph or plot of a signal f(t) as a function of time or other time variables. It provides information about the signal's properties and the changes it undergoes with respect to time.



Mono tonal signals



Bi-tonal signal



SIGNALS

From time to frequency

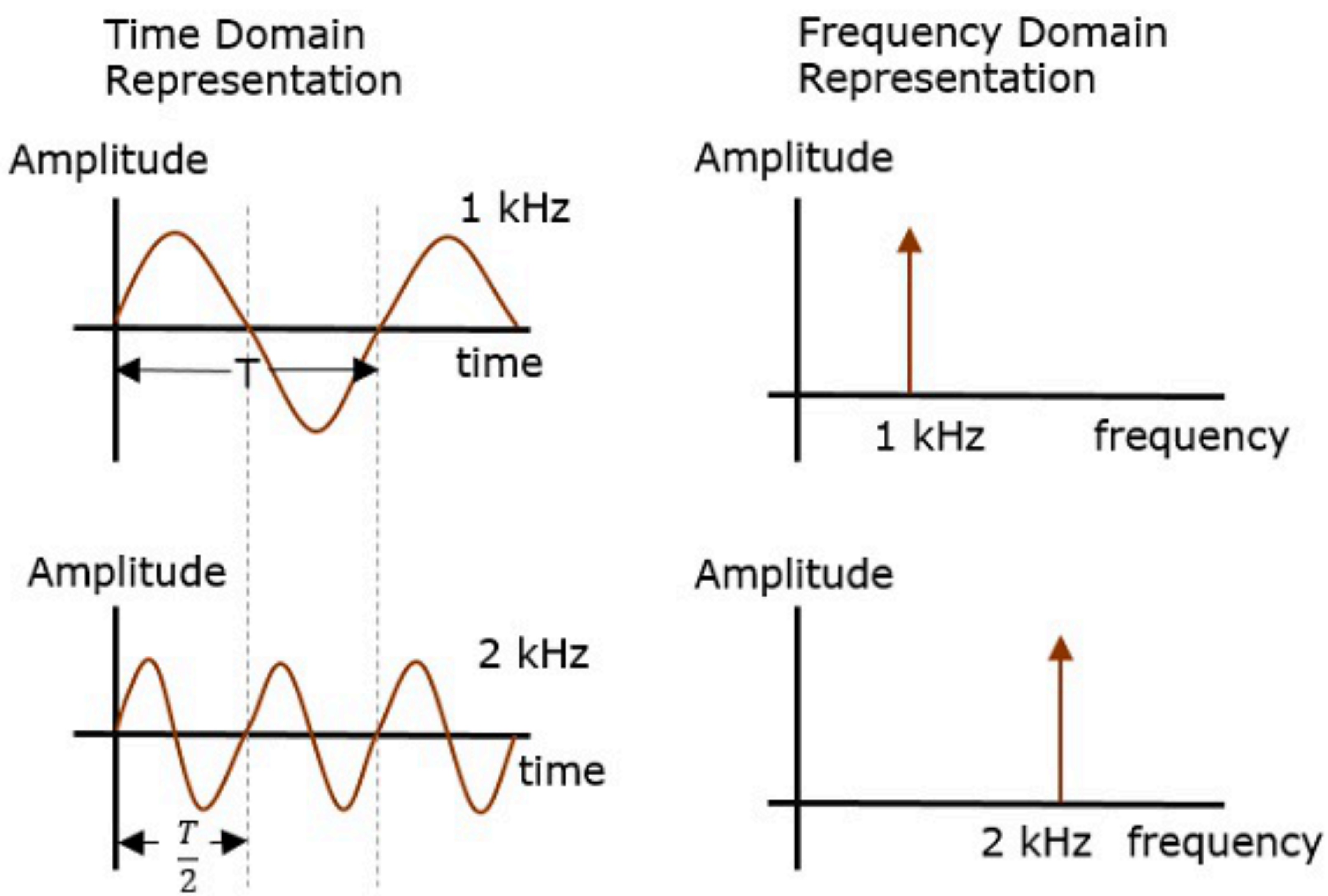
The frequency domain is a graphical representation of a signal's amplitude at different frequencies. It provides insight into the frequency content of a signal and the relative contribution of different frequency components to the overall signal.

T period

f frequency

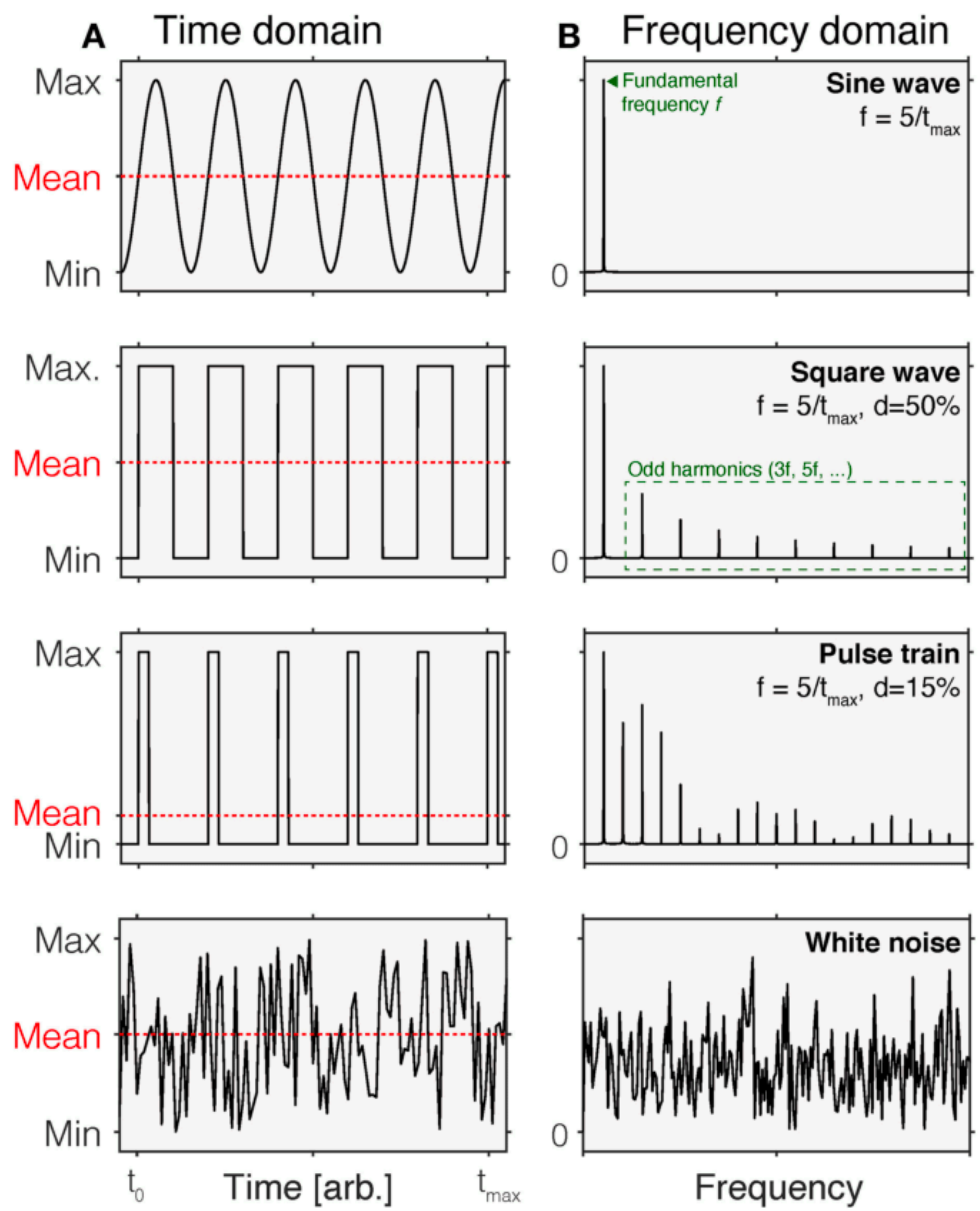
$$f = \frac{1}{T}$$

$$\omega = 2\pi f$$



SIGNALS

From time to frequency: spectral content



From: 10.3389/fneur.2021.654158

SIGNALS

Signals in the frequency domain: the Fourier transform

The Fourier Transform is a mathematical operation that transforms a function of time (or space) into a representation in the frequency domain. It is named after the French mathematician Joseph Fourier, who first introduced the concept. The Fourier Transform is a fundamental tool in signal processing, mathematics, and various scientific and engineering fields.

Mathematically, the continuous Fourier Transform F(ω) of a function f(t) is defined as follows:

$$F(\omega) = \int_{-\infty}^{\infty} f(t) \cdot e^{-i\omega t} \, dt$$

- $F(\omega)$ is the complex function in the frequency domain,
- $f(t)$ is the function in the time domain,
- ω is the angular frequency, and
- i is the imaginary unit.

The inverse Fourier Transform, which transforms a function from the frequency domain back to the time domain, is given by:

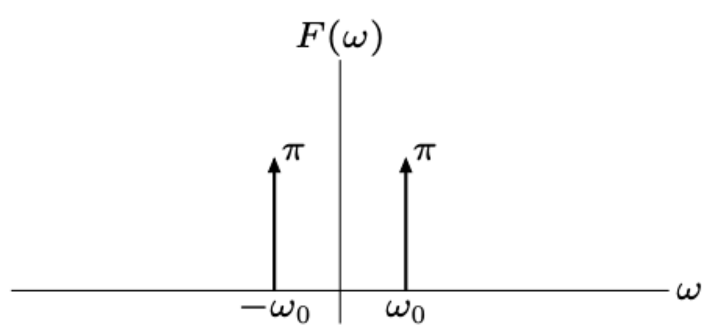
$$f(t) = \frac{1}{2\pi} \int_{-\infty}^{\infty} F(\omega) \cdot e^{i\omega t} \, d\omega$$

The Fourier Transform essentially decomposes a signal into its constituent frequencies. It represents a signal as a sum of sinusoidal functions with different frequencies, amplitudes, and phases. The resulting frequency domain representation provides valuable insights into the frequency content of the original signal.

Signals in the frequency domain: the Fourier transform

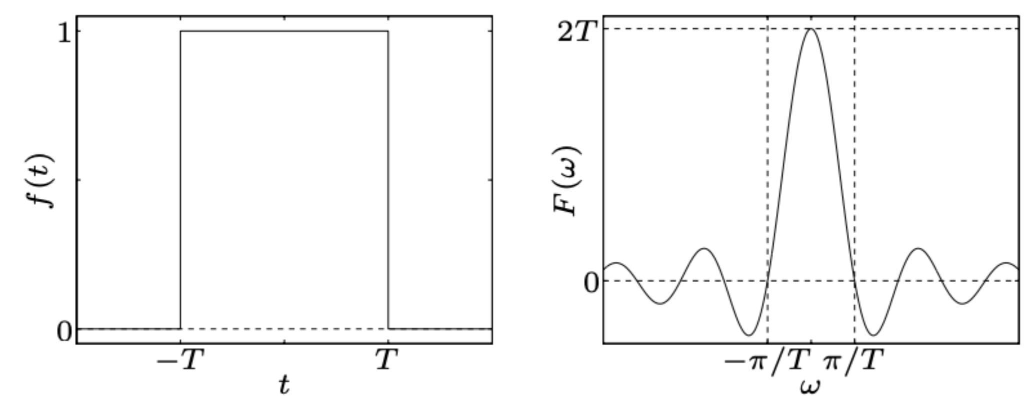
sinusoidal signals: Fourier transform of $f(t) = \cos \omega_0 t$

$$
\begin{aligned}
F(\omega) &= \frac{1}{2}\int_{-\infty}^{\infty} \left(e^{j\omega_0 t} + e^{-j\omega_0 t}\right) e^{-j\omega t}\, dt \\
&= \frac{1}{2}\int_{-\infty}^{\infty} e^{-j(\omega-\omega_0)t} dt + \frac{1}{2}\int_{-\infty}^{\infty} e^{-j(\omega+\omega_0)t} dt \\
&= \pi\delta(\omega-\omega_0) + \pi\delta(\omega+\omega_0)
\end{aligned}
$$



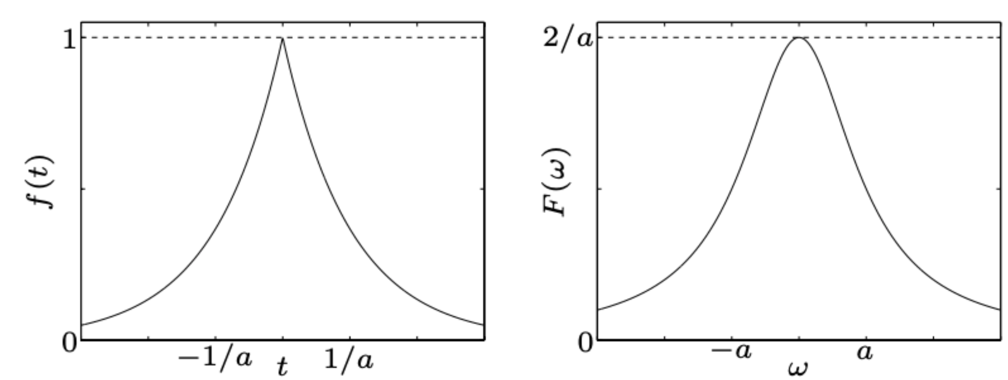
rectangular pulse: $f(t) = \begin{cases} 1 & -T \le t \le T \\ 0 & |t| > T \end{cases}$

$$
F(\omega) = \int_{-T}^{T} e^{-j\omega t}\, dt = \frac{-1}{j\omega}\left(e^{-j\omega T} - e^{j\omega T}\right) = \frac{2\sin \omega T}{\omega}
$$



double-sided exponential: $f(t) = e^{-a|t|}$ (with $a > 0$)

$$
\begin{aligned}
F(\omega) = \int_{-\infty}^{\infty} e^{-a|t|} e^{-j\omega t}\, dt &= \int_{-\infty}^{0} e^{at} e^{-j\omega t}\, dt + \int_{0}^{\infty} e^{-at} e^{-j\omega t}\, dt \\
&= \frac{1}{a - j\omega} + \frac{1}{a + j\omega} \\
&= \frac{2a}{a^2 + \omega^2}
\end{aligned}
$$



https://web.stanford.edu/class/ee102/lectures/fourtran

SIGNALS

Signals in the frequency domain: the Fourier transform

Applications of the Fourier Transform include:

1. Signal Processing: Analysis and filtering of signals in various applications, such as audio processing, image processing, and communications.

2. Communication Systems: Modulation and demodulation of signals in communication systems.

3. Spectral Analysis: Identification of frequency components in a signal, essential in fields like vibration analysis and spectrum analysis.

4. Quantum Mechanics: Representation of wavefunctions in quantum mechanics.

5. Medical Imaging: Techniques like Magnetic Resonance Imaging (MRI) and computed tomography (CT) use Fourier Transform methods.

6. Optics: Analysis of light waves and optical systems.

The Fourier Transform is a powerful tool that has broad applications in understanding and manipulating signals in various domains of science and engineering. It provides a way to analyze complex signals and gain insights into their frequency characteristics.

DIGITAL SIGNAL PROCESSING: A BRIEF INTRO

Digital Signal Processing (DSP) refers to the manipulation, analysis, and interpretation of signals using digital techniques. Signals, in this context, are representations of information that vary over time. These signals can be analog in nature, originating from the physical world, or already in digital form. DSP involves the application of mathematical and computational methods to process these signals for various purposes.

DSP allows for the extraction of meaningful information from signals, the enhancement of signal quality, and the efficient transmission and storage of digital data for a wide range of applications.

https://link.springer.com/chapter/10.1007/978-981-10-8081-4_12

DIGITAL SIGNAL PROCESSING: APPLICATION

Key aspects of Digital Signal Processing include:

1. **Representation of Signals**:

Signals can be analog or digital. Analog signals are continuous and can take any value within a range, while digital signals are discrete and typically represented as a sequence of numbers.

2. **Sampling and Quantisation**:

Analog signals are often converted into digital signals through a process called sampling, where the signal is measured at discrete time intervals. Quantization involves representing the sampled values with a finite number of bits.

3. **Fourier Transform**:

The Fourier Transform is a fundamental tool in DSP that decomposes a signal into its frequency components. This is valuable for analyzing and manipulating signals in the frequency domain. In practice, when dealing with discrete signals (such as those represented in digital form), the Discrete Fourier Transform (DFT) or its fast computation algorithm, the Fast Fourier Transform (FFT), is often used.

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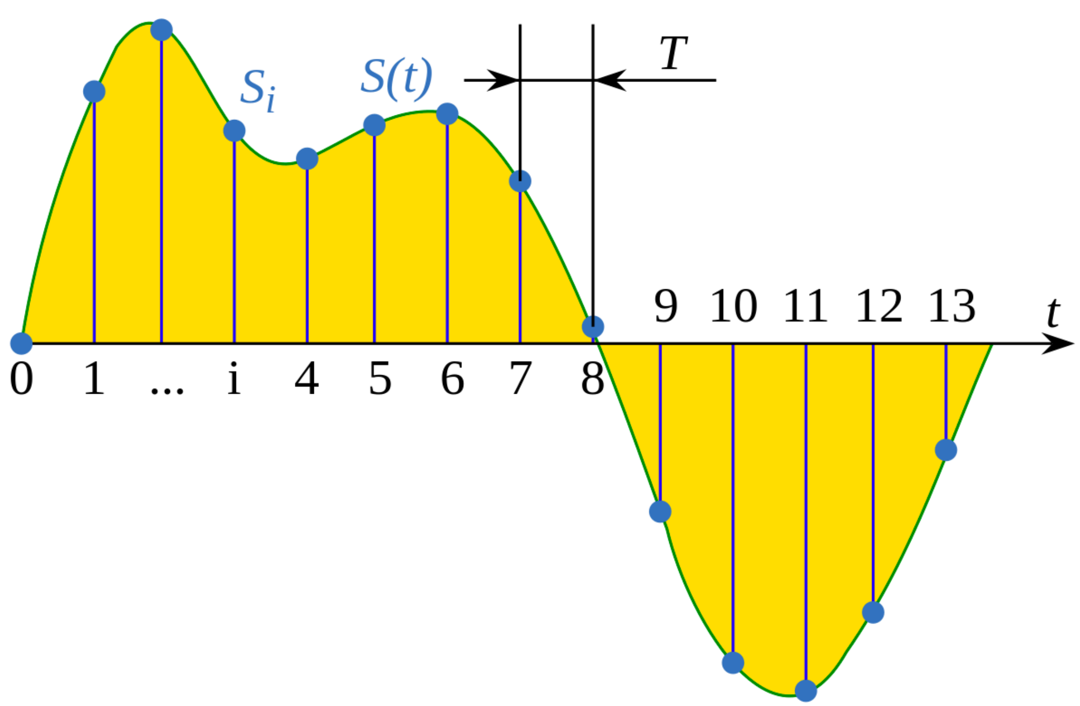
DIGITAL SIGNAL PROCESSING: APPLICATION

A real analog signal: Sampling and quantisation

Signal Sampling:

Definition: Sampling is the process of converting a continuous-time signal into a discrete-time signal by measuring its amplitude at regular intervals.

Sampling Rate (Sampling Frequency, f_{sampl}): The number of samples taken per unit of time is known as the sampling rate. It is usually measured in hertz (Hz) and is crucial in determining the fidelity of the digitized signal. The Nyquist-Shannon sampling theorem states that the sampling rate must be at least twice the highest frequency present in the signal to avoid aliasing.



$$S(t) \longrightarrow \Sigma_{i=0 \ldots N} S_i(t_i)$$

$$t_i = i \cdot T \ , \ T = \frac{1}{f_{sampl}}$$

DIGITAL SIGNAL PROCESSING: APPLICATION

A real analog signal: Sampling and quantisation

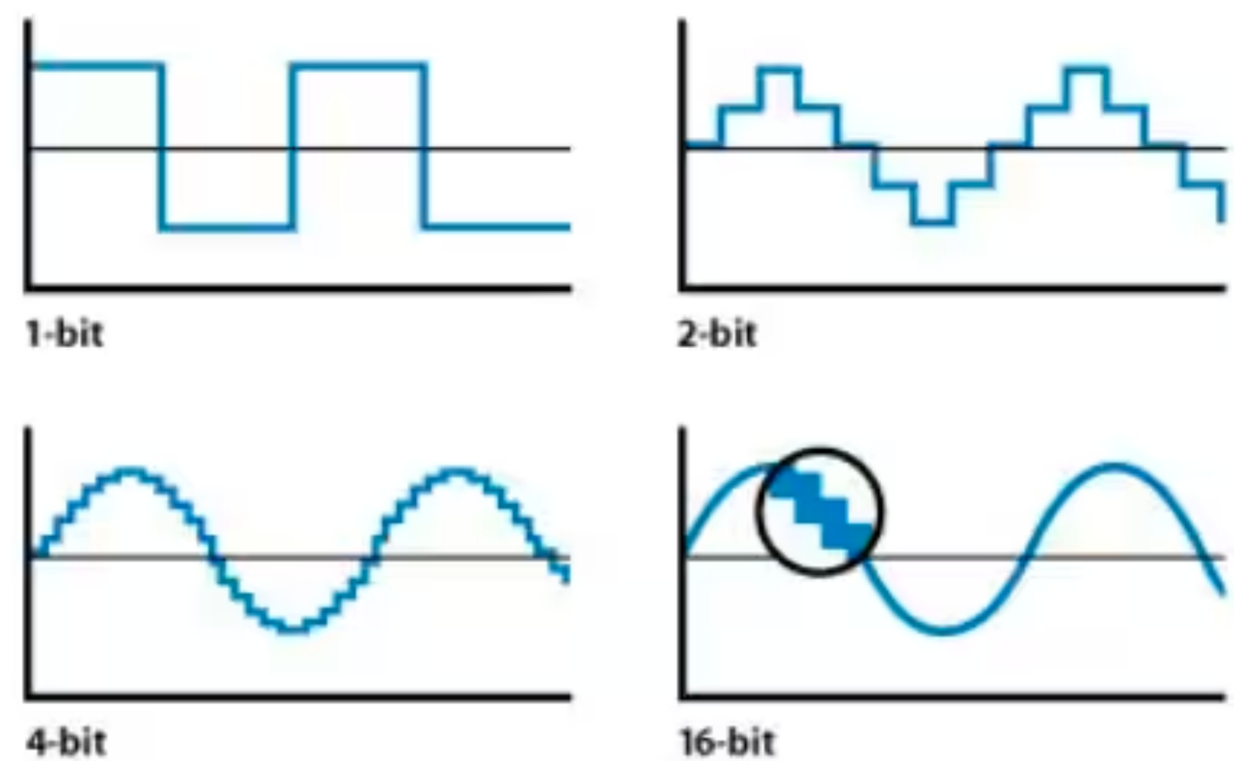
Signal Quantization:

Definition: Quantization is the process of converting the continuous amplitude values obtained through sampling into a finite set of discrete digital values.

Quantization Levels: The range of possible amplitude values is divided into a finite number of discrete levels. The resolution of quantization is determined by the number of bits used to represent each sample. More bits result in a higher resolution but also larger file sizes.

Quantization Error: Due to the finite number of levels, quantization introduces an error known as quantization error, representing the difference between the actual analog value and its quantized representation.

Example: In an 8-bit quantization (ADC - analog to digital converter), the continuous amplitude values obtained from sampling are approximated to the nearest of the 256 available digital levels.



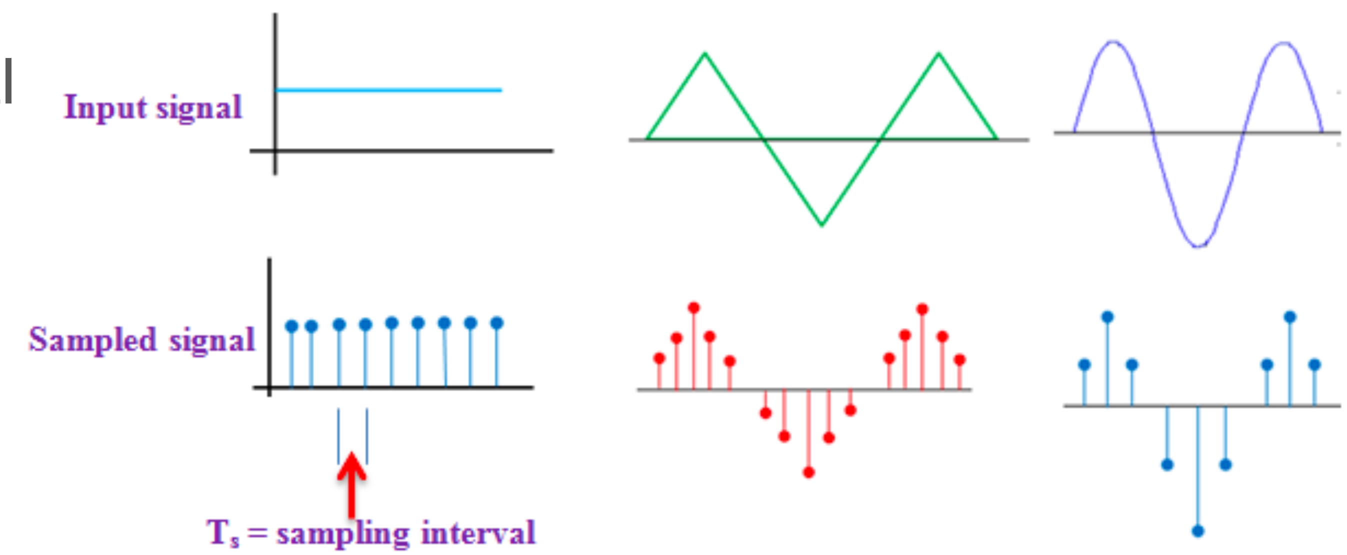
DIGITAL SIGNAL PROCESSING: APPLICATION

From analog signal to a digital signal

Digital Signal:

Combination: The combined process of sampling and quantization results in a digital signal, where the original continuous-time signal is represented as a sequence of discrete amplitude values.

Advantages: Digital signals are easier to manipulate, store, and transmit in electronic systems. They are less susceptible to noise during transmission and can be processed using digital signal processing (DSP) techniques.



In summary, signal sampling involves discretizing the time dimension by measuring a continuous signal at regular intervals, while quantization involves discretizing the amplitude dimension by representing the continuous amplitude values with a finite set of digital levels. Together, these processes convert a continuous analog signal into a digital signal suitable for processing in digital systems.

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DIGITAL SIGNAL PROCESSING: APPLICATION

A real analog signal: Discrete Fourier transform

The Discrete Fourier Transform (DFT) is a mathematical transformation that converts a finite sequence of equally spaced samples of a function into its frequency domain representation. It is a discrete version of the continuous Fourier Transform and is particularly important in the analysis of digital signals. The DFT is commonly computed using algorithms like the Fast Fourier Transform (FFT) for efficient calculations.

Mathematically, given a sequence x[n] of length N the DFT X[k] is computed by the formula:

$$X[k] = \sum_{n=0}^{N-1} x[n] \cdot e^{-i2\pi kn/N}$$

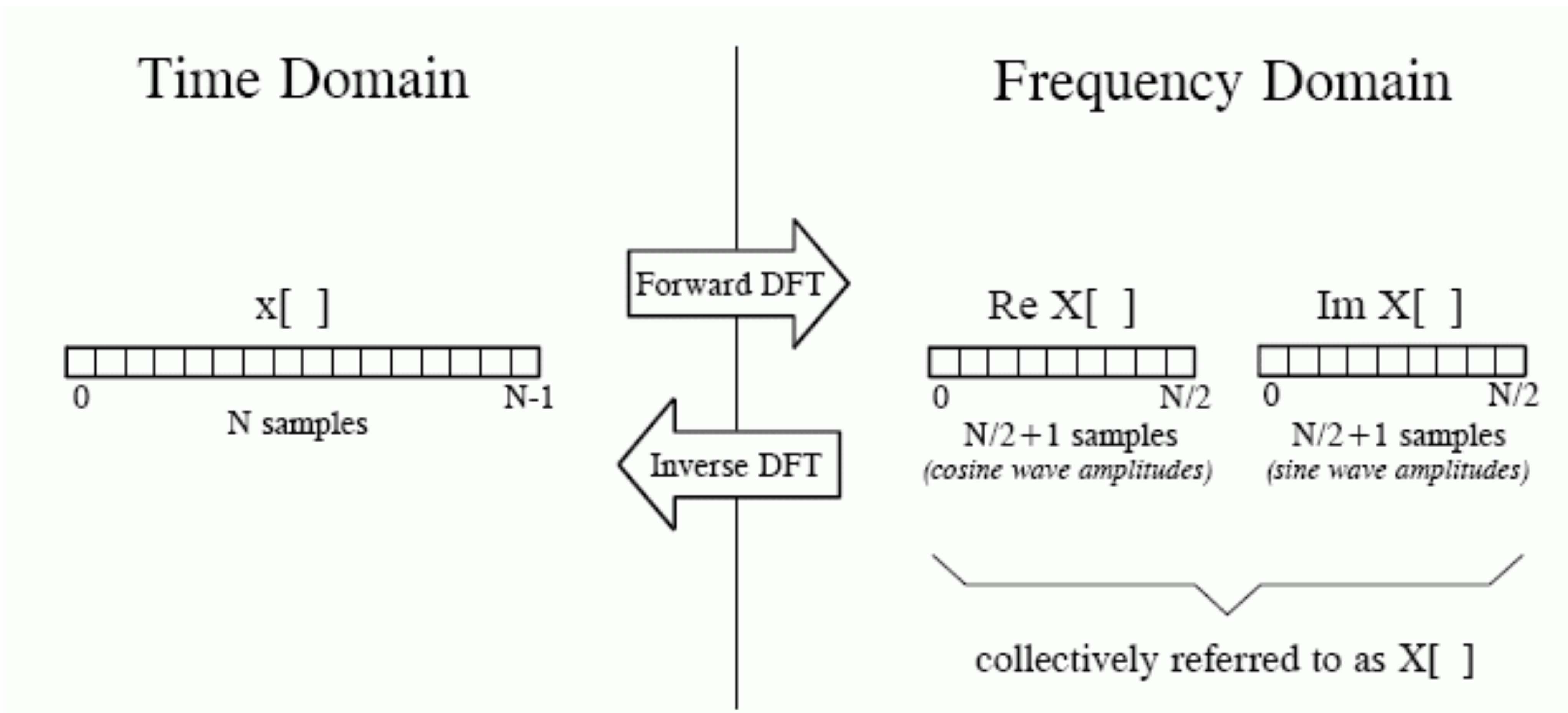
- $X[k]$ is the complex amplitude at frequency index k,
- $x[n]$ is the discrete sequence in the time domain,
- N is the number of samples in the sequence,
- i is the imaginary unit.

The DFT essentially decomposes a sequence of discrete values into its constituent frequencies, providing information about the amplitude and phase of each frequency component.

The Fast Fourier Transform (FFT) is an algorithmic approach to compute the DFT efficiently. It reduces the number of computations needed for the transform, making it practical for real-time applications and large datasets.

DIGITAL SIGNAL PROCESSING: APPLICATION

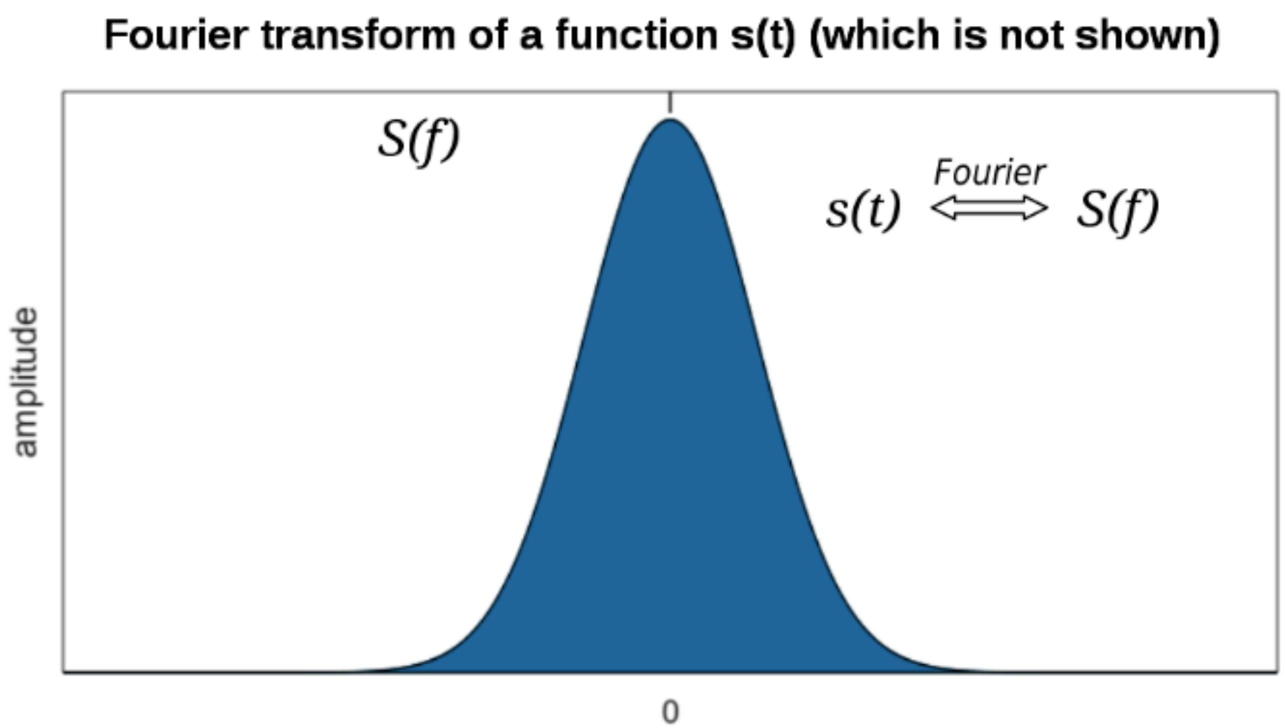
A real analog signal: Discrete Fourier transform



https://www.dspguide.com/ch8/2.htm

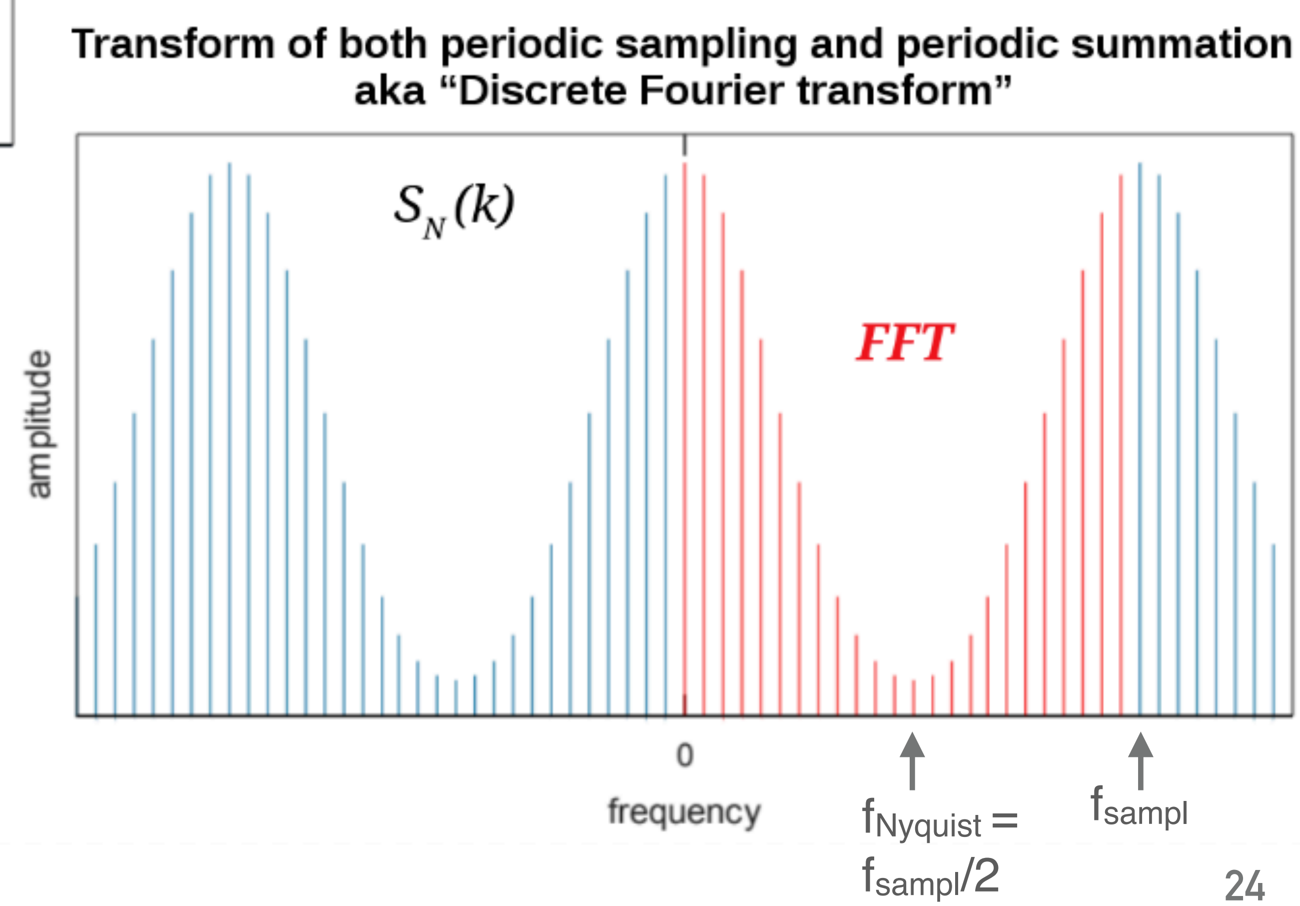
DIGITAL SIGNAL PROCESSING: APPLICATION

A real analog signal: Discrete Fourier transform



FT: from continuous time domain

DFT: from discretized time domain



https://www.dspguide.com/ch8/2.htm

DIGITAL SIGNAL PROCESSING: APPLICATION

Key aspects of Digital Signal Processing include:

4. **Digital Signal Processing Algorithms**:

DSP employs a variety of algorithms to perform operations on digital signals. These algorithms can include filtering, convolution, Fourier analysis, and various mathematical transformations.

5. **Filtering and Filtering Techniques**:

Filtering is a common operation in DSP used to modify or extract information from a signal. Techniques such as low-pass, high-pass, and band-pass filtering are applied to remove or enhance specific frequency components.

6. **Signal Compression**:

DSP techniques are often used for signal compression, reducing the amount of data needed to represent a signal without significant loss of information. This is important for efficient storage and transmission of signals.

7. **Application Areas**:

DSP is utilized in various application areas, including telecommunications, audio and speech processing, image processing, medical signal processing, radar systems, and control systems.

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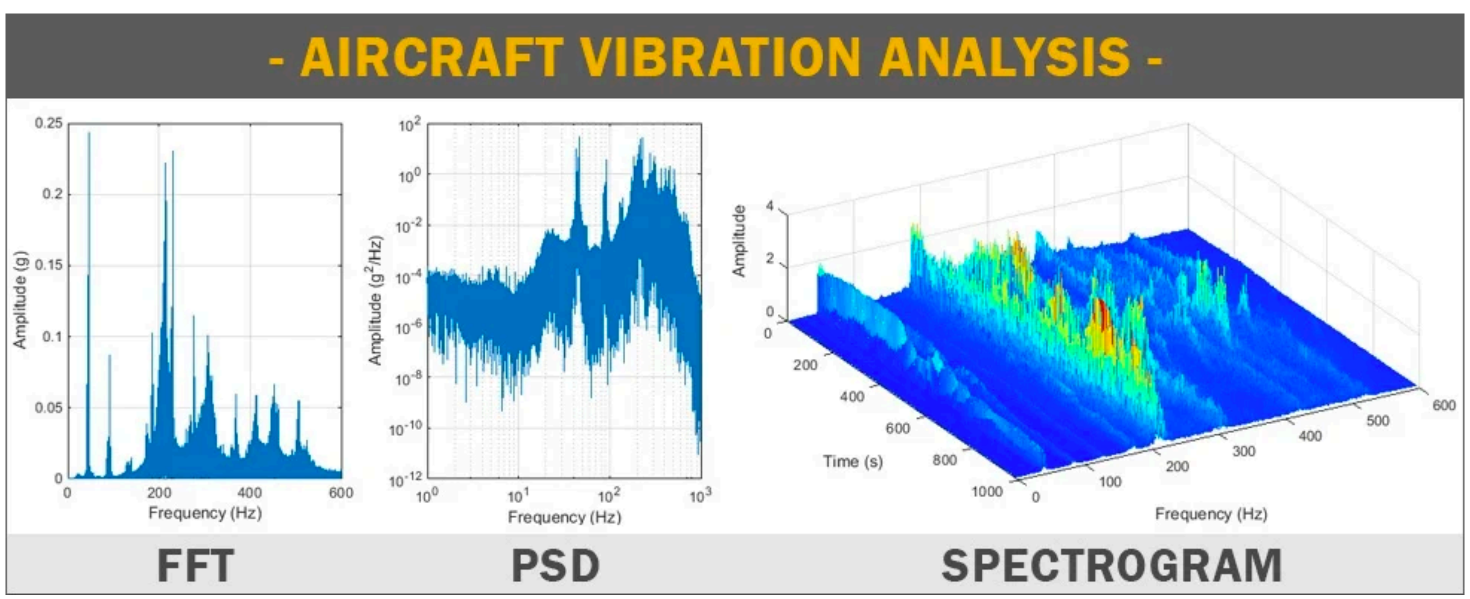
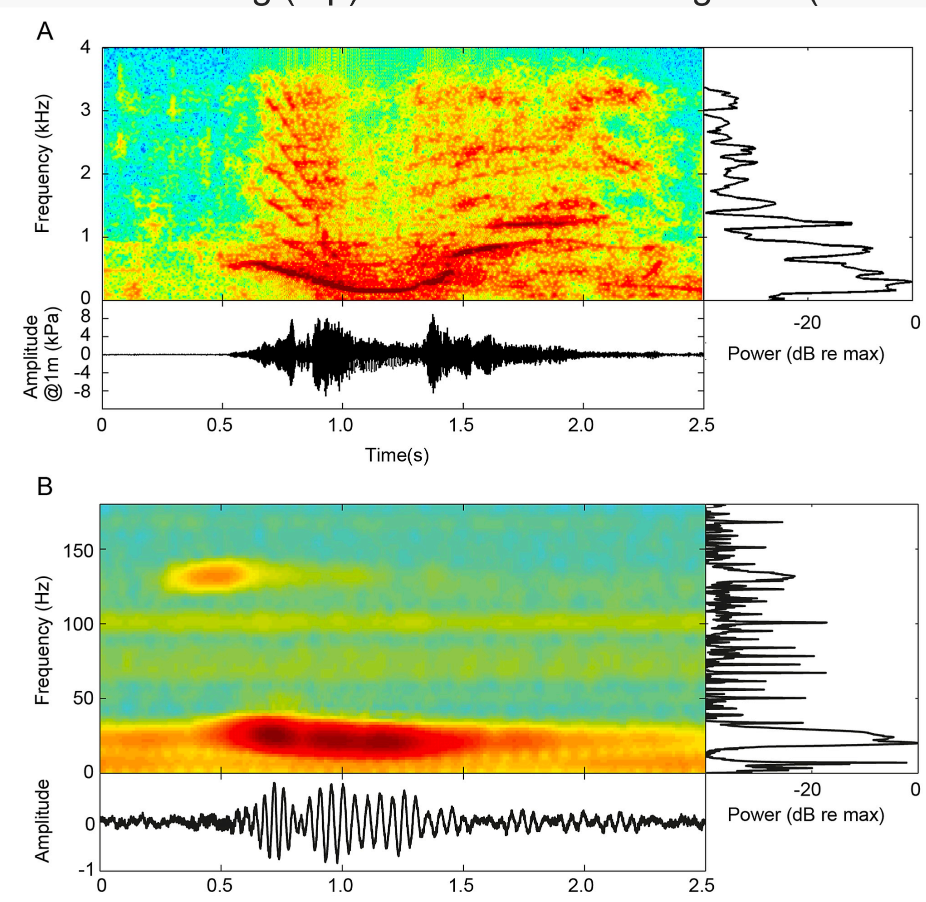
DIGITAL SIGNAL PROCESSING

A real analog signal: Spectral analysis

- **Power Spectral Density** (PSD): measure of how the power of a signal is distributed across different frequencies. It provides information about the intensity of various frequency components within a signal.
- **Spectrogram**: 2D representation that shows how the frequency content of a signal changes over time. It is often used for analyzing time-varying signals.



Bowhead whale song (top) and a fin whale song note (bottom)

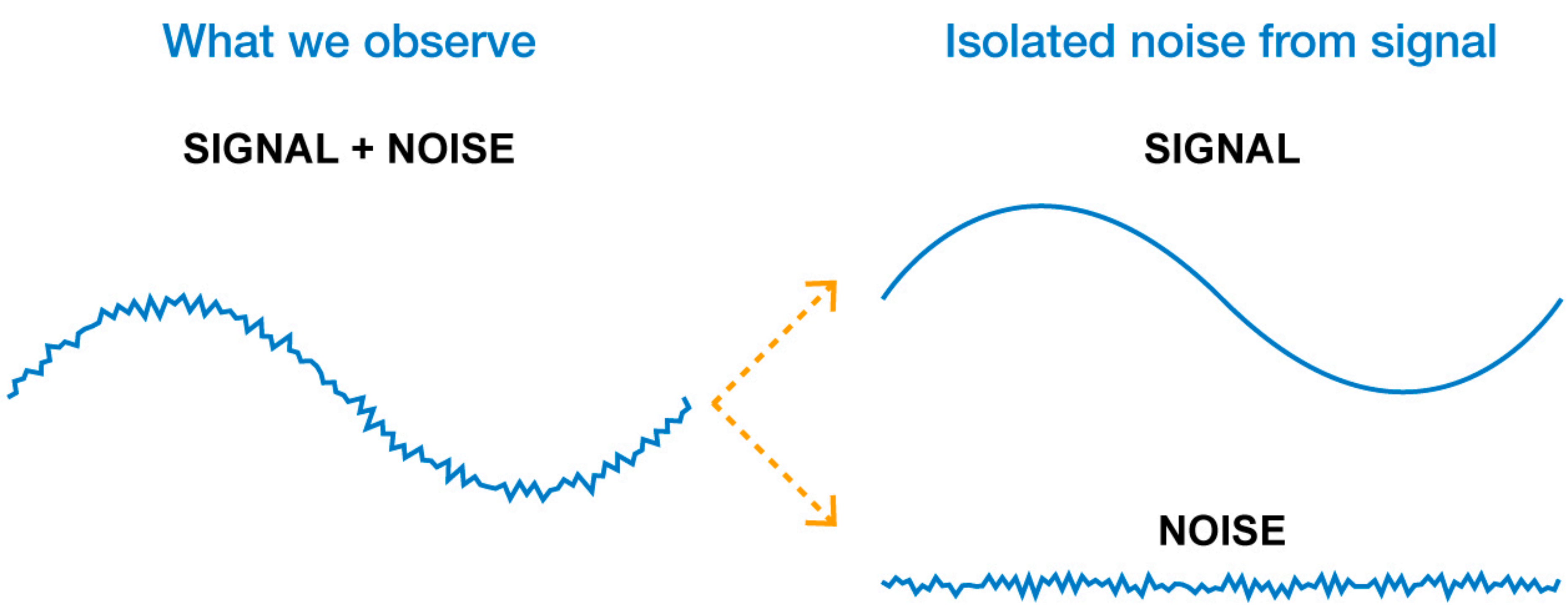


https://doi.org/10.1371/journal.pone.0052072.g004

DIGITAL SIGNAL PROCESSING

A real analog signal: Spectral analysis - signal and noise

Noise in a signal refers to any unwanted or random interference or disturbance that affects the fidelity of the original signal. It can manifest as additional electrical fluctuations, disturbances, or variations in the signal that are not part of the intended information.



DIGITAL SIGNAL PROCESSING

A real analog signal: Spectral analysis - Noise

The two main types of noise that can affect a signal are:

1. **Additive Noise**:

Definition: Additive noise is external interference or random disturbances that are added to the original signal during its transmission or processing.

Characteristics: It manifests as an additional signal that is combined with the original signal, making it more challenging to extract the intended information.

Examples: Electromagnetic interference (EMI), radio-frequency interference (RFI), and thermal noise are common forms of additive noise.

2. **Multiplicative Noise**:

Definition: Multiplicative noise is a type of noise that modulates or scales the amplitude of the original signal.

Characteristics: Instead of being added to the signal, multiplicative noise alters the amplitude of the signal, introducing variability or fluctuations.

Examples: Gain variations in an amplifier, atmospheric turbulence affecting optical signals, and fading in wireless communication channels are examples of multiplicative noise.

In practical scenarios, signals often encounter a combination of additive and multiplicative noise.

DIGITAL SIGNAL PROCESSING

A real analog signal: Spectral analysis - Noise

1. **Continuous or Broadband Noise**:

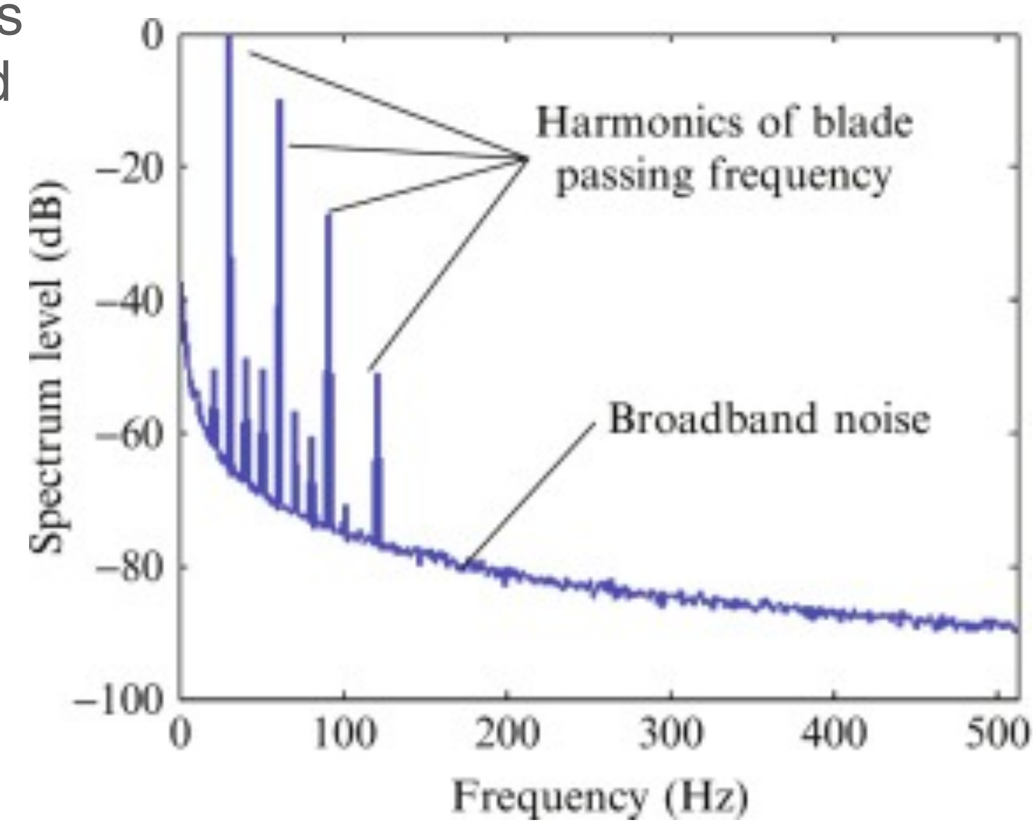
Definition:*Continuous noise is a type of noise that is characterized by equal energy at every frequency. Continuous noise is not limited to specific frequencies and covers a broad range of the spectrum.

Example: White noise generated by a random signal with equal intensity at all frequencies is a common example of continuous noise.

2. **Tonal or Single-Frequency Noise**:

Definition: Tonal noise is a type of noise that is concentrated at a specific frequency or a narrow range of frequencies. Unlike continuous noise, tonal noise is not spread evenly across the spectrum but is prominent at specific frequencies, resulting in one or more distinctive tones/pitch.

Example: A continuous tone from a single frequency interference in an audio signal or a specific frequency component in a communication channel affected by interference.

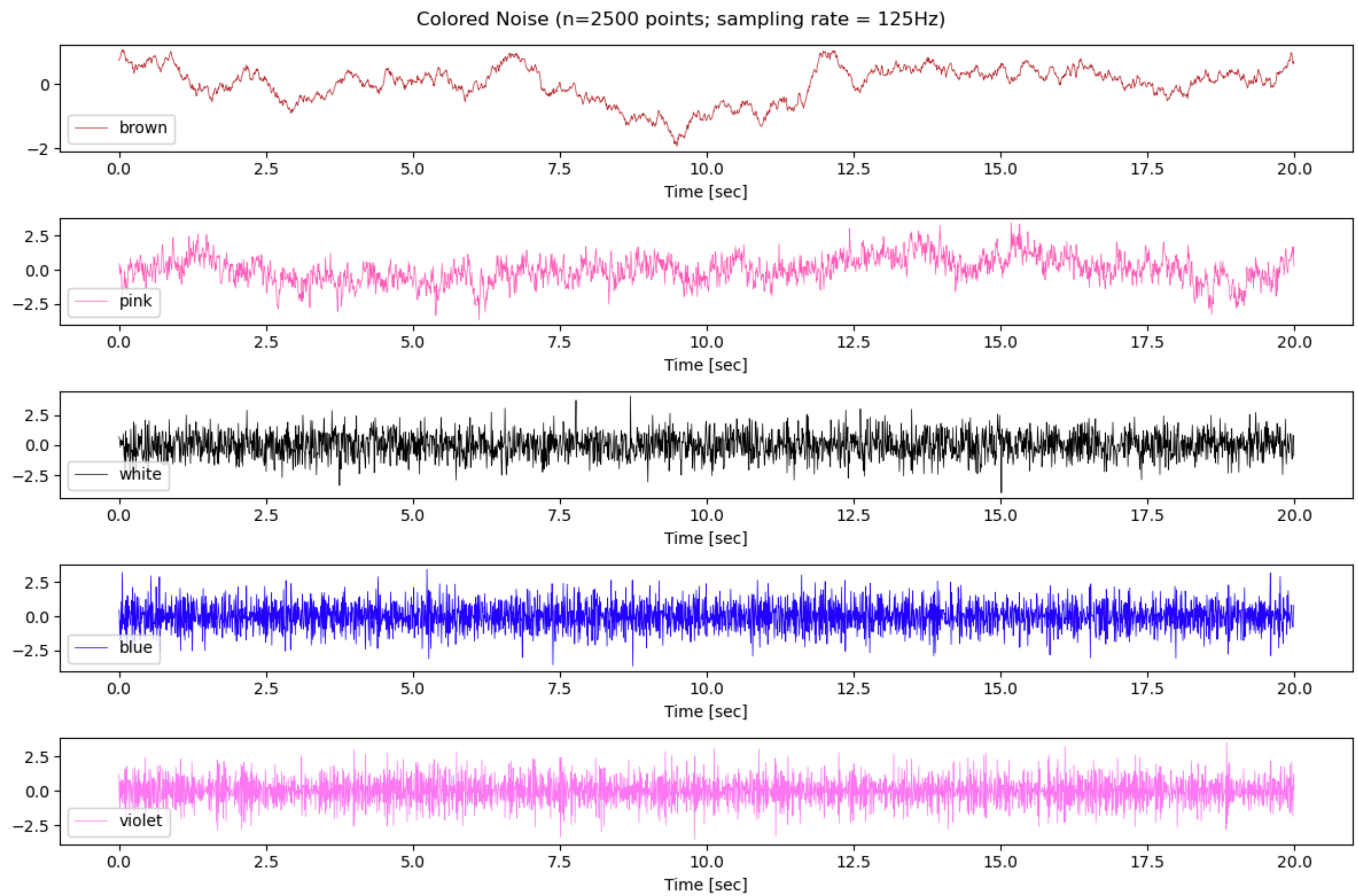


https://www.sciencedirect.com/topics/engineering/tone-noise

DIGITAL SIGNAL PROCESSING

A real analog signal: Spectral analysis - Broadband noise

We refer to "color of noise" to describe the spectral characteristics of different types of random noise.



DIGITAL SIGNAL PROCESSING

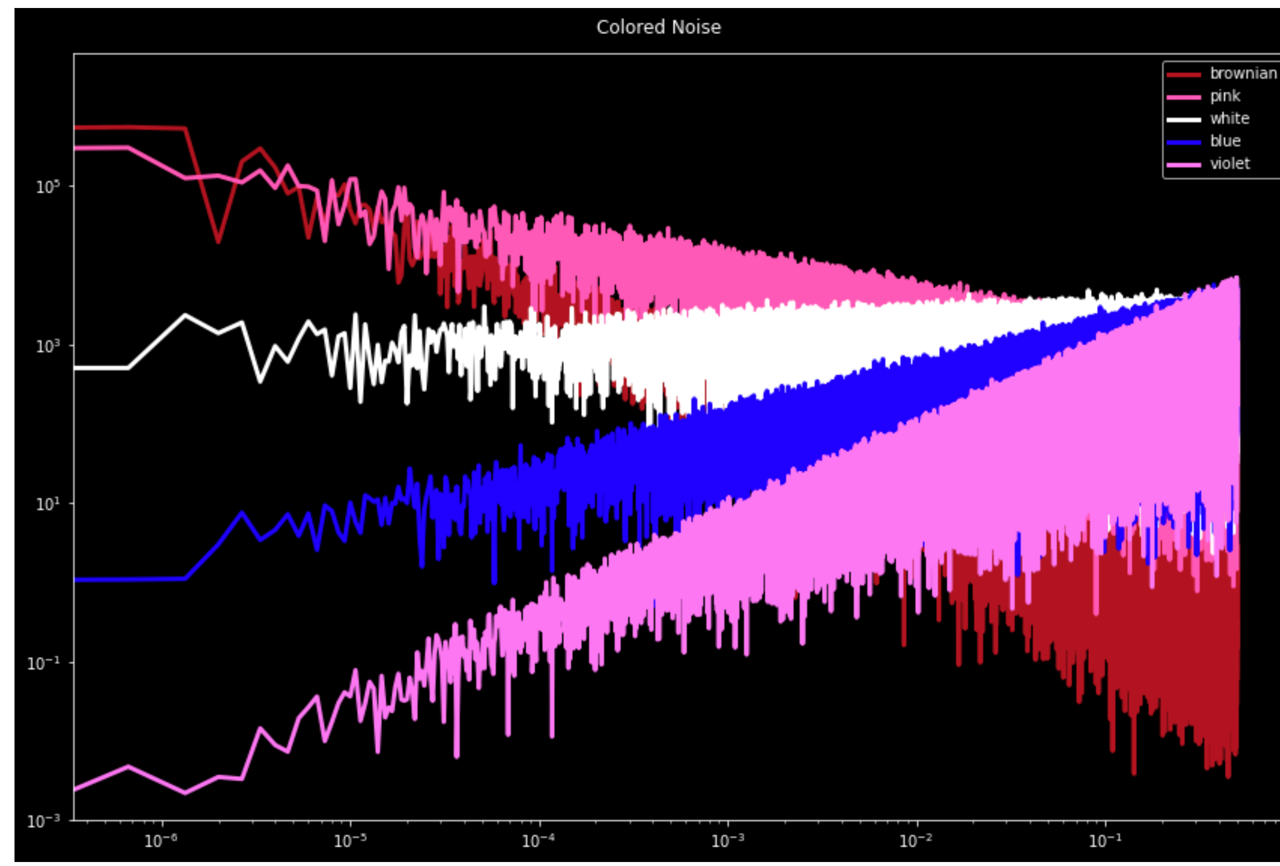
A real analog signal: Spectral analysis - Broadband noise

We refer to "color of noise" to describe the spectral characteristics of different types of random noise.

1. **White Noise:** It has constant power spectral density across all frequencies. In other words, it has equal energy at every frequency.

2. **Pink Noise:**:"1/f noise" or "flicker noise" has a power spectral density inversely proportional to frequency (~ 1/f). As the frequency increases, the power decreases.

3. **Blue Noise:: ”azure noise" has a power spectral density directly proportional to frequency (~ f). It means that as the frequency increases, the power also increases. It is biased towards higher frequencies.erceived as having more energy at the higher end of the spectrum.



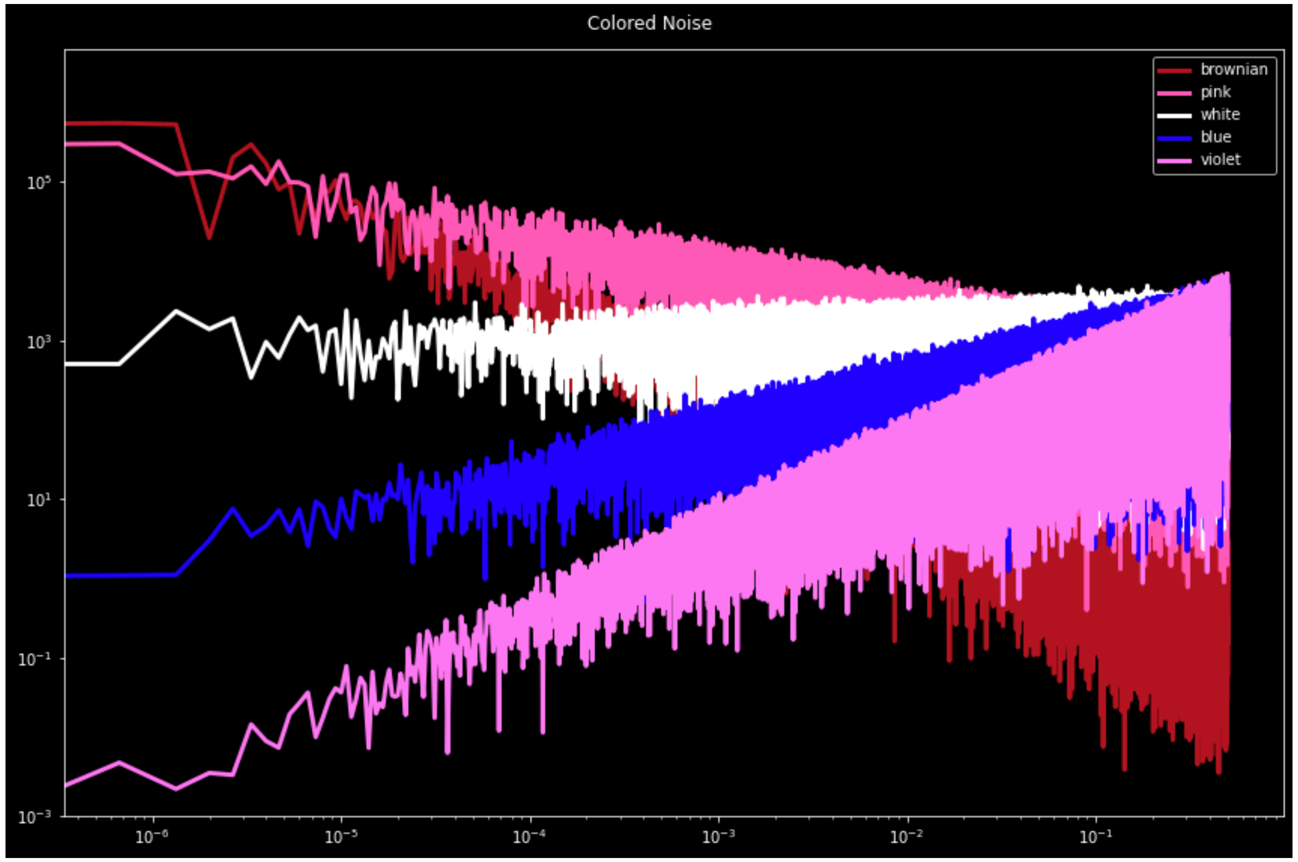
DIGITAL SIGNAL PROCESSING

A real analog signal: Spectral analysis - Broadband noise

We refer to "color of noise" to describe the spectral characteristics of different types of random noise.

4. **Brownian Noise: “Brown noise" or "Red noise" has a power spectral density inversely proportional to the square of the frequency (~ 1/√ f). It has more energy at lower frequencies.

5. **Violet Noise:**: ”purple noise" or "ultraviolet noise," has a power spectral density directly proportional to the square of the frequency (~ f^2). It is biased towards higher frequencies and is perceived as having more energy at the higher end of the spectrum.



DIGITAL SIGNAL PROCESSING: APPLICATION

Key aspects of Digital Signal Processing include:

4. **Digital Signal Processing Algorithms**:

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5. **Filtering and Filtering Techniques**:

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6. **Signal Compression**:

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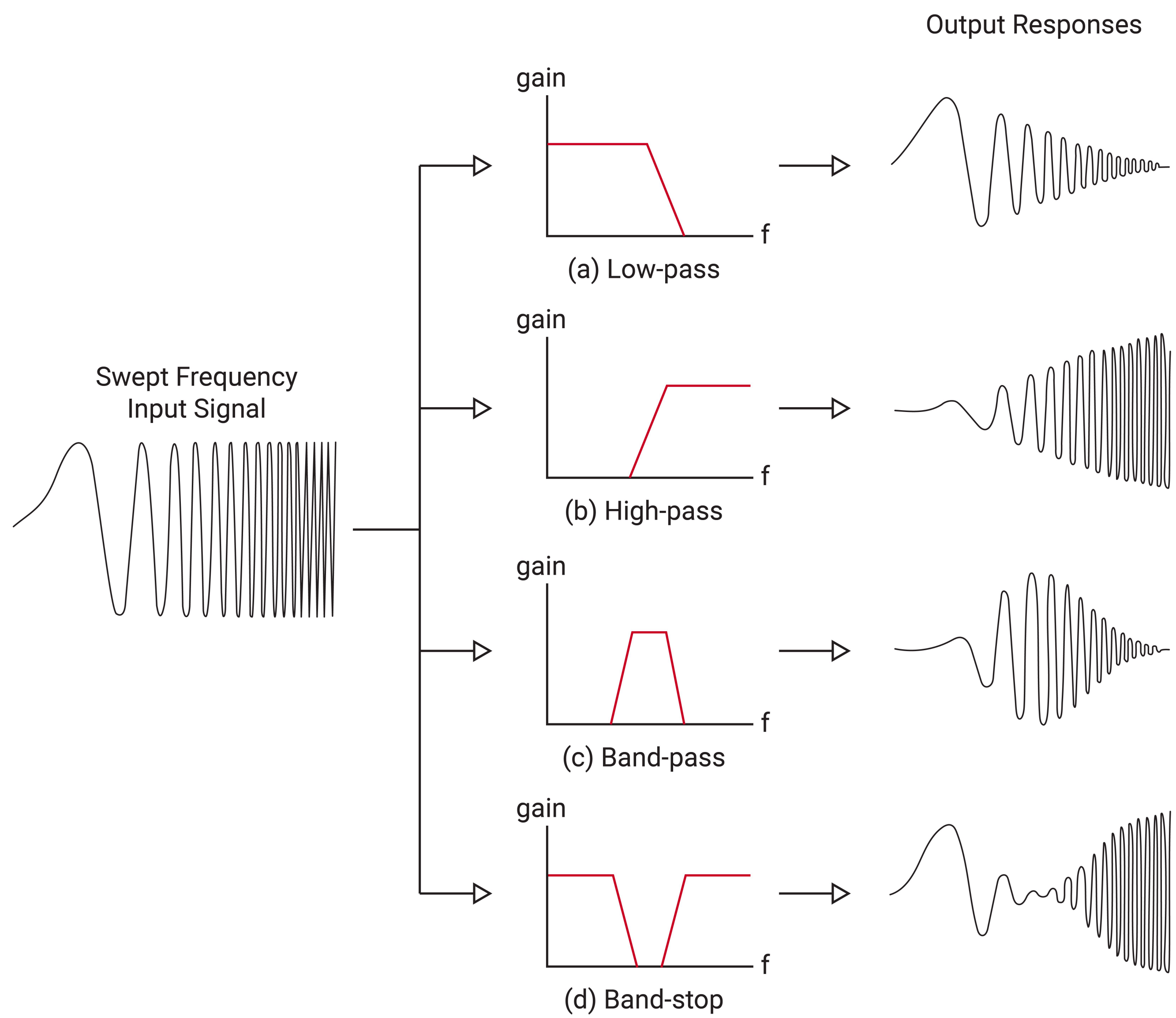
7. **Application Areas**:

DSP is utilized in various application areas, including telecommunications, audio and speech processing, image processing, medical signal processing, radar systems, and control systems.

DIGITAL SIGNAL PROCESSING

A real analog signal: Filters

Signal filtering is the process of modifying or separating components of a signal based on certain criteria. Filters are used to emphasize or suppress specific frequencies in a signal, and they play a crucial role in various applications such as audio processing, image processing, telecommunications, and control systems.



DIGITAL SIGNAL PROCESSING

A real analog signal: Passive filters - examples

Passive filters rely solely on passive components like resistors, capacitors, and inductors.

1. **Low-Pass Filter** (LPF):

Application: Audio Processing

Purpose: Allows low-frequency components to pass through while attenuating higher frequencies.

Example: Filtering out high-frequency noise from an audio signal.

2. **High-Pass Filter** (HPF):

Application: Image Processing

Purpose: Allows high-frequency components to pass through while attenuating lower frequencies.

Example: Enhancing the edges of an image by emphasising high-frequency details.

https://www.edmprod.com/audio-filters/

Fig. 3: original image

Fig. 5: Gaussian high pass filter

https://api.semanticscholar.org/CorpusID:29478051

DIGITAL SIGNAL PROCESSING

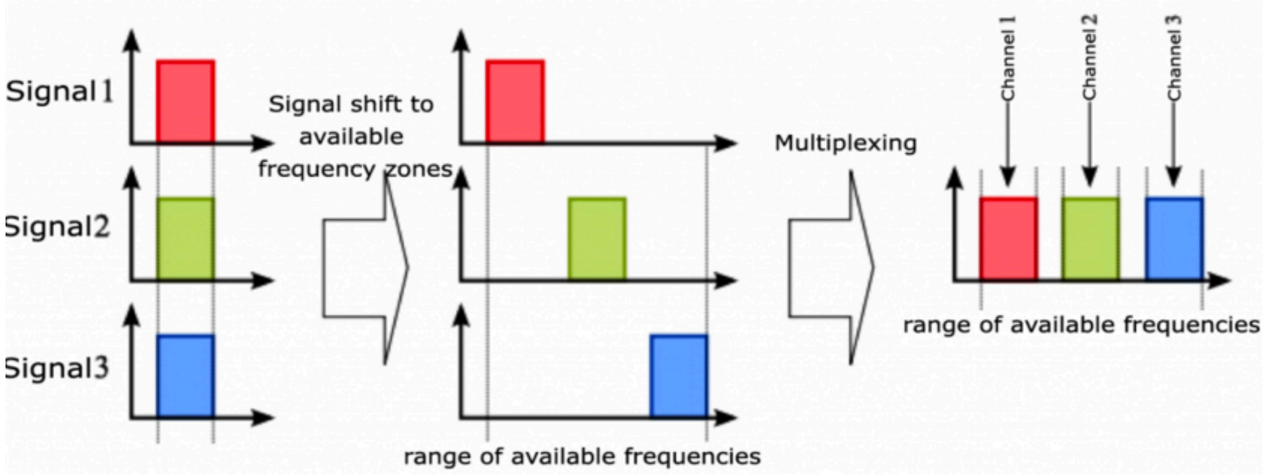
A real analog signal: Passive filters - examples

3. **Bandpass Filter** (BPF):

Application: Communication Systems

Purpose: Selectively allows a certain range of frequencies to pass through.

Example: Filtering specific channels in a frequency-division multiplexing (FDM) communication system to isolate individual signals.



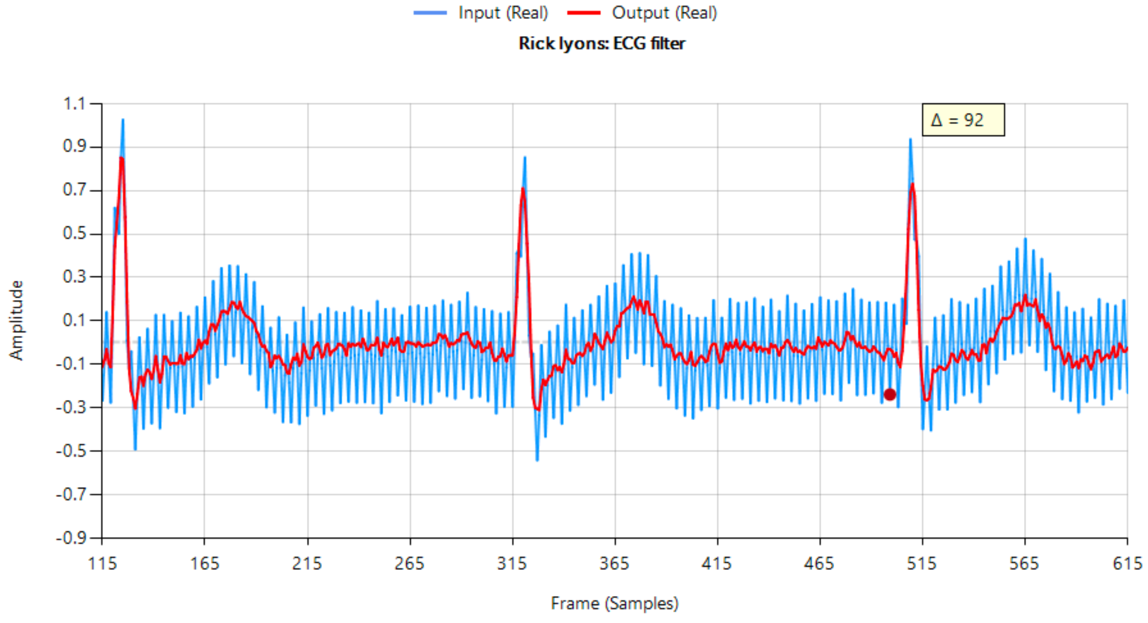
https://electronicspost.com/what-is-multiplexing-frequency-division-multiplexing-fdm-and-time-division-multiplexing-tdm/

4. **Bandstop Filter** (Notch Filter):

Application: Biomedical Signal Processing

Purpose: Eliminates a narrow band of frequencies, often used to remove powerline interference.

Example: Filtering out 50/60 Hz noise from an electrocardiogram (ECG) signal recorded in an environment with electrical interference.



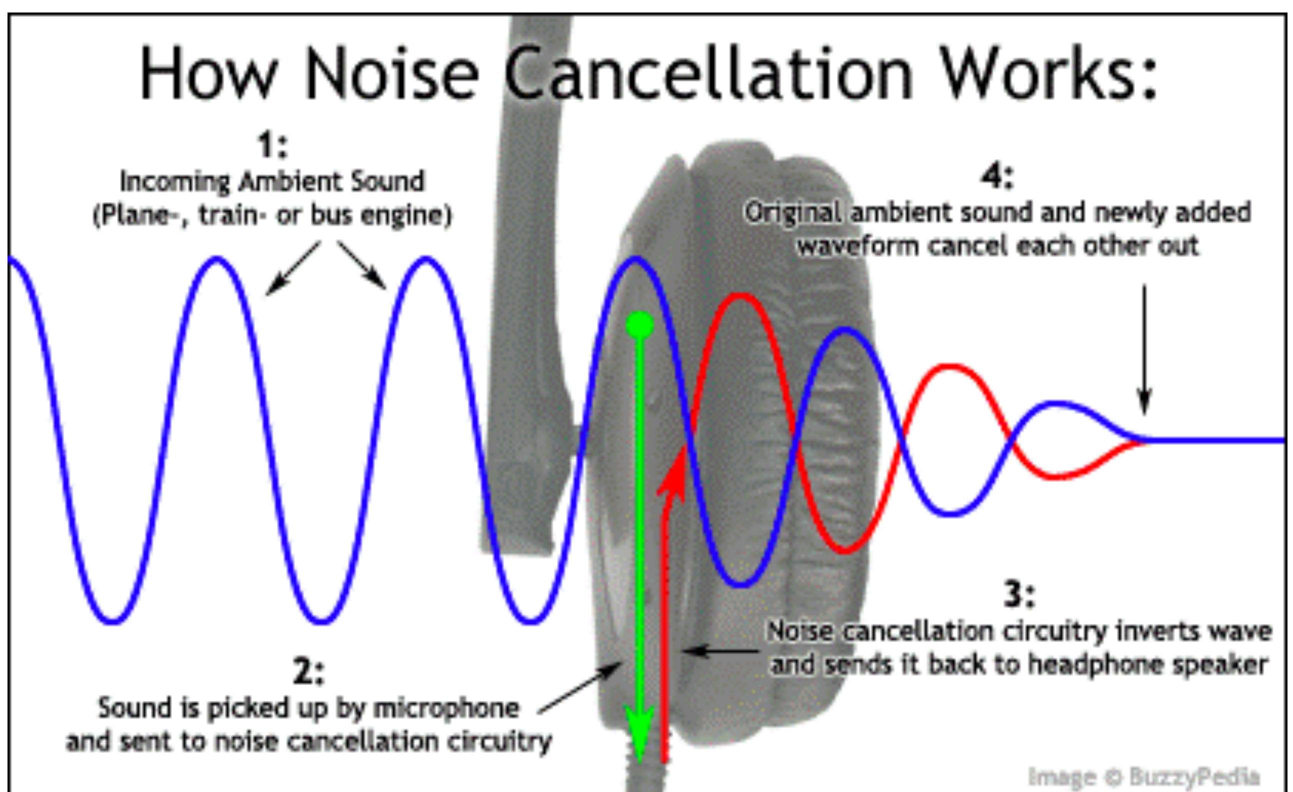
https://www.dsprelated.com/showarticle/1383.php

DIGITAL SIGNAL PROCESSING

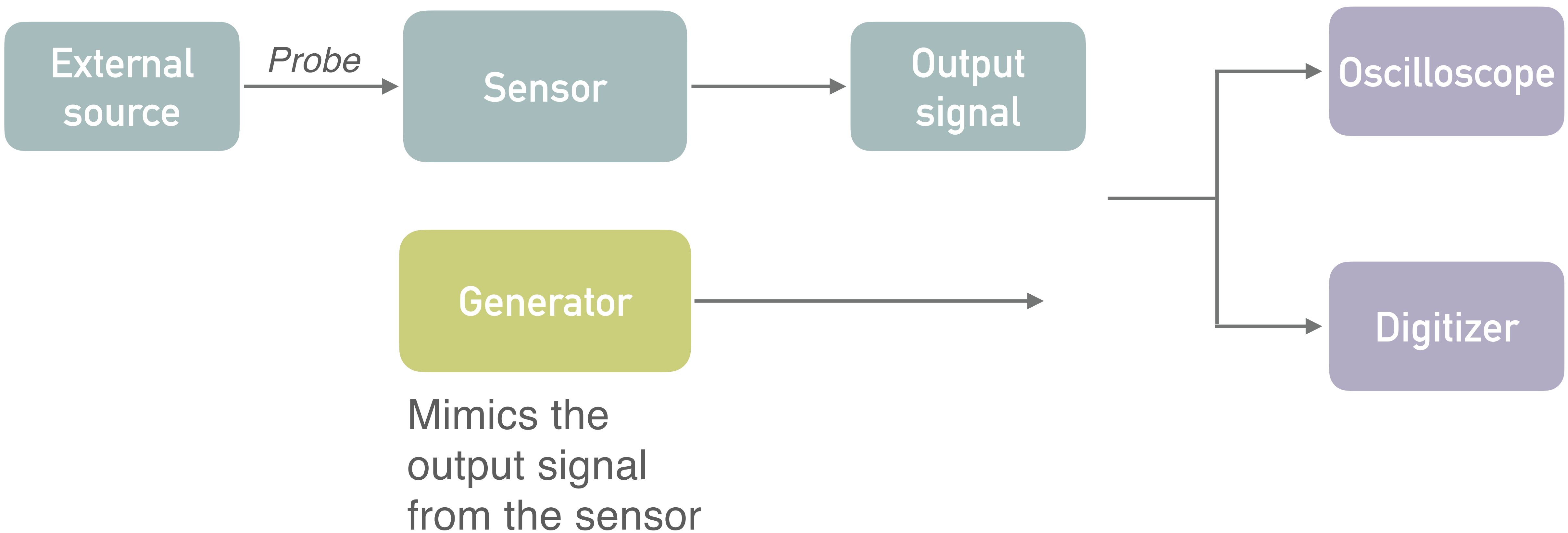
A real analog signal: Active filters - examples

Active filters are a type of electronic filter that utilises active components, such as operational amplifiers and transistors, to achieve desired frequency response characteristics. Active filters can provide amplification and are often used in applications requiring precise frequency shaping.

Active noise cancellation (ANC) techniques: Active noise cancellation involves the use of microphones to pick up ambient noise, and electronic circuits, often implemented with active filters, generate anti-noise signals with inverted phases. These anti-noise signals are then combined with the original noise, effectively canceling out specific frequencies and reducing overall noise levels. Active noise cancellation is widely employed in headphones and other audio systems to provide a quieter and more immersive listening experience by actively combating unwanted background noise.



EXAMPLE: A SENSOR READOUT CHAIN



- Generate a signal: the waveform generator
- Read the signal output (1): the oscilloscope
- Read the signal output (2): the digitiser

Lab.1 today