4. • Analogue to Digital Conversion 1 (ADC)

• Digital Signal processing

Analogue

↓

discrete Sample — ADC — Analogue to digital converter

↓ DSP = Digital Signal processing

processed — DAC — Digital to analogue converter

↓

continous — Filter

↓

Analogue

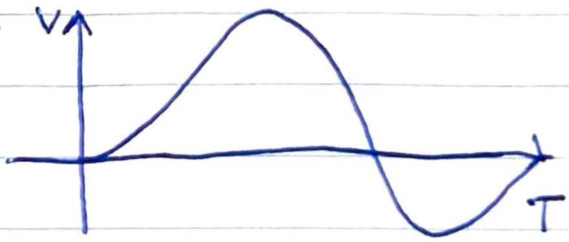
Input = Digital, Output = Analogue

note: • Analog signal = continous

~~Discrete Signal~~

• Digital Signal = discrete

• Continous (Analogue) signals vs Digital Signals



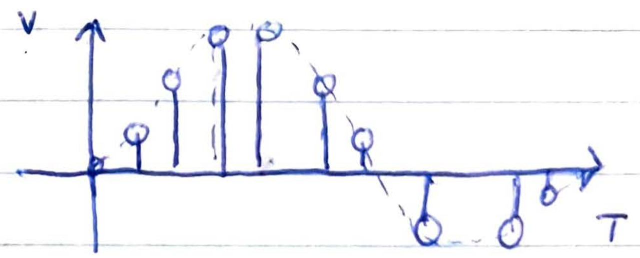
• Analogue/continous signals:

- continous time
- Continous amplitude

* cannot be directly processed by a DSP (Digital Signal processing)

• Digital signals

- discrete time
- discrete-amplitude

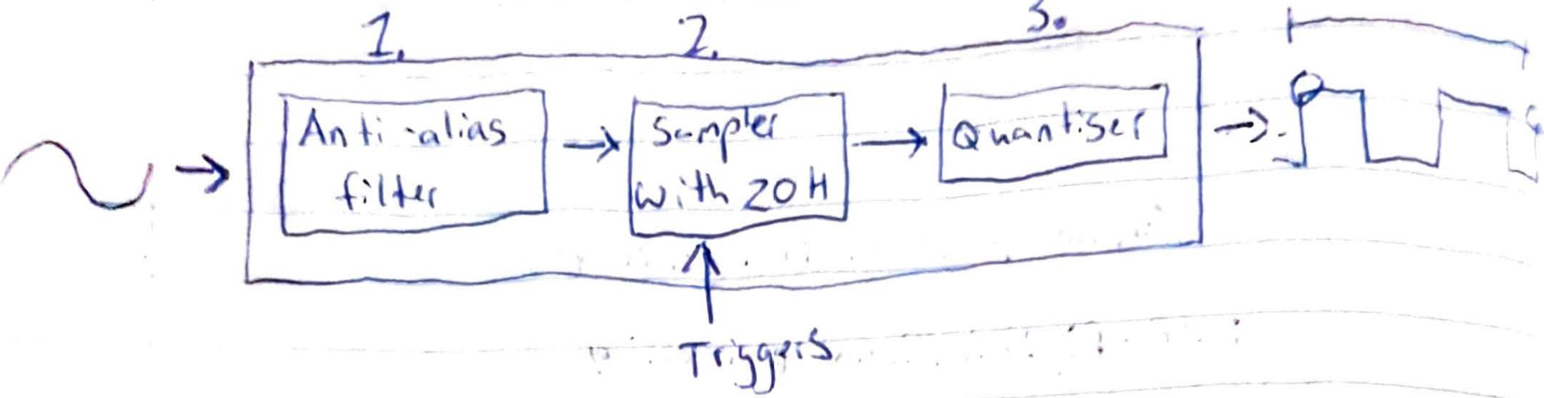


• continous-time → discreted time = Sampling

• Continous-amplitude → discrete-amplitude = quantization

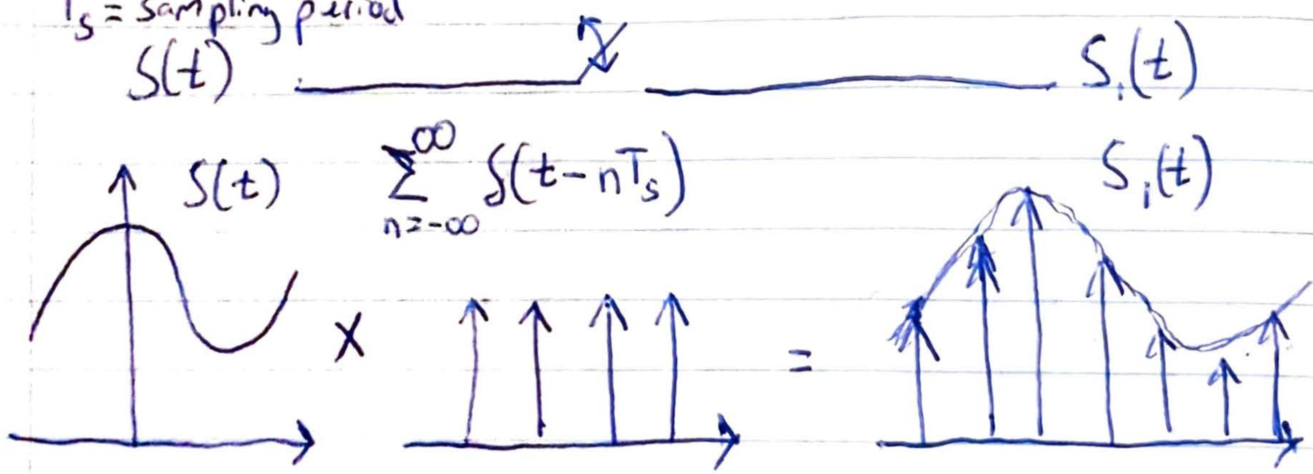
quantization = process of constraining an input from continous to discreted

- Analogue-to-Digital converter (ADC)



- Nyquist-Shannon Sampling Theorem (Way of avoiding Aliasing)
- Continuous-time → discrete-time = sampling
- Ideal sampling = uniform unit impulse train

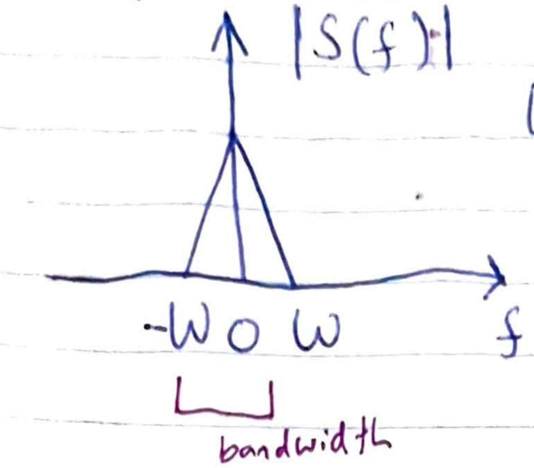
T_s = sampling period



$$S_i(t) = S(t) \times \sum_{n=-\infty}^{\infty} \delta(t-nT_s) = \sum_{n=-\infty}^{\infty} S(nT_s)\cdot\delta(t-nT_s)$$

• Nyquist Rate

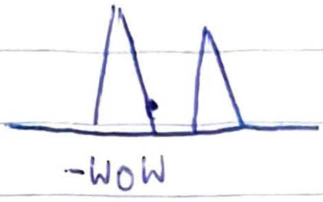
Fourier Transform of $S(t)$



W = bandwidth

$f_s = \frac{1}{T_s}$

$\boxed{\begin{array}{l} \circ \text{ when } f_s \geq 2W \\ \circ \text{ when } f_s \leq 2W \end{array}}$



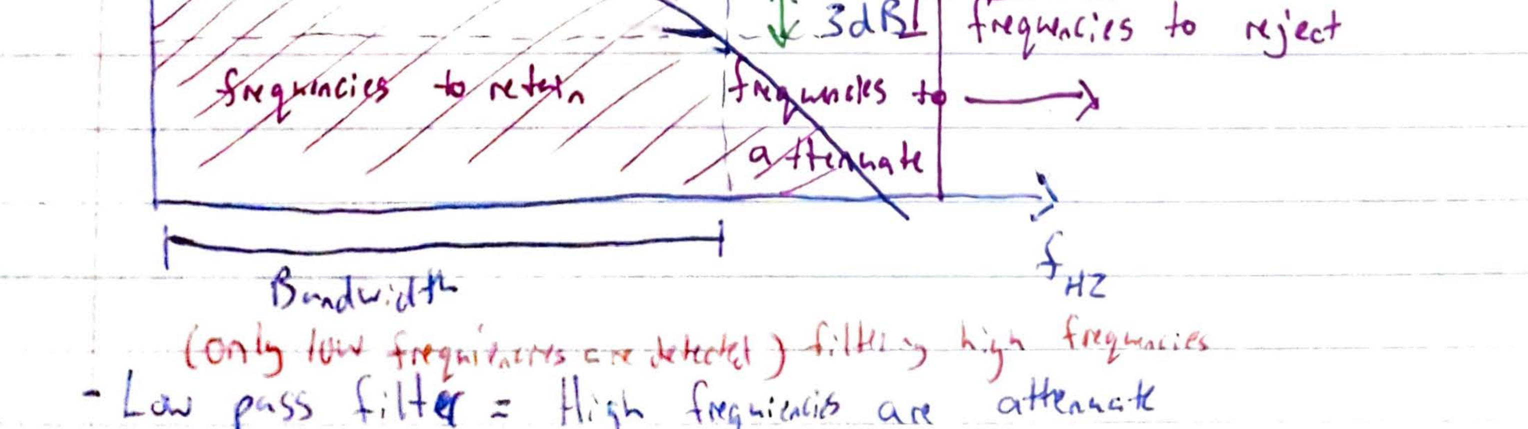
no Aliasing

Aliasing

★ Aliasing = when wave samples overlap eachother

$\boxed{\begin{array}{l} f_s \geq 2W = \text{No Aliasing} \\ f_s \leq 2W = \text{Aliasing} \end{array}}$

• Anti-Alias Filter



(only low frequencies are detected) filtering high frequencies

- Low pass filter = High frequencies are attenuate

(cut's off the high frequencies)

- Digital signal conditioning (via signal averaging)

$$\boxed{z(nT_s) = \frac{1}{M}\sum_{m=0}^{M-1} y((n-m)T_s)}$$

* Reduce noise power by factor of M for independent & identically distributed noise samples