

Analog and Digital Signals

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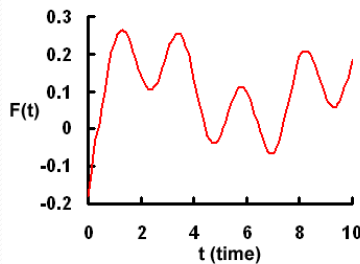
Analog and Digital Signals

1. From Analog to Digital Signal
2. Sampling & Aliasing

Analog and Digital Signals

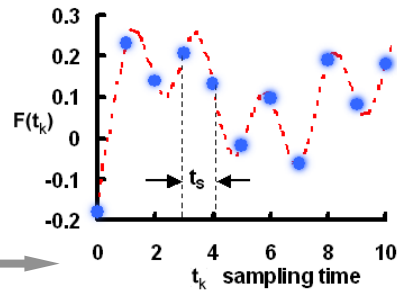
Analog Signals

Continuous function F of a continuous variable t (t can be time, space etc) : $F(t)$



Digital Signals

Discrete function F_k of a discrete (sampling) variable t_k with k an integer: $F_k = F(t_k)$ (F at t_k)



Function F is sampled with sampling frequency f_s (uniformly and periodic)
 $f_s = 1/t_s$ Hz, for example, if $t_s = 0.001$ sec $\Rightarrow f_s = 1000$ Hz

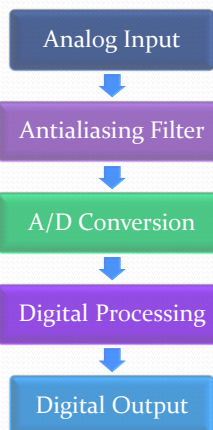
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Digital System Implementation

Important issues:

Analysis bandwidth, Dynamic range



- Pass/stop bands
- Sampling rate, Number of bits, and further parameters
- Digital format

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Sampling

How fast must we sample a continuous signal to preserve its information content?



Examples:

Turning wheels of a car or a train in a movie

- 25 frames per second, i.e., $f_s = 25$ samples/sec ≈ 25 Hz
- Train starts \Rightarrow wheels appear to go clockwise
- Train accelerates \Rightarrow wheels go counter clockwise

Rotating propeller of an airplane captured by a Mobile phone camera.



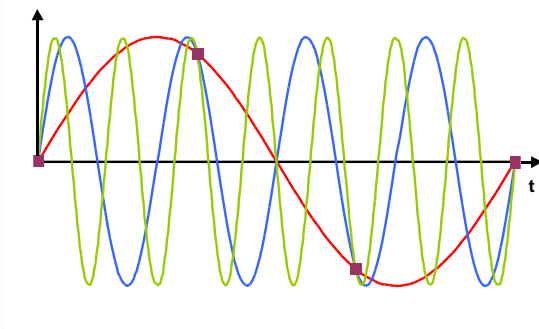
Both examples: Low sampling frequency leading to Frequency misidentification

Note that, we assume **uniform sampling** unless stated otherwise.

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Sampling a Sine Wave



$$\text{— } s(t) = \sin(2\pi f_0 t)$$

$$\blacksquare s(t) @ f_{\text{Sample}}$$

For example:

$$f_0 = 1 \text{ Hz}, f_{\text{Sample}} = 3 \text{ Hz}$$

$$\text{— } s_1(t) = \sin(2\pi 4t)$$

$$\text{— } s_2(t) = \sin(2\pi 7t)$$

$s(t) @ f_{\text{Sample}}$ represents exactly all sine-waves $s_k(t)$ defined by:

$$s_k(t) = \sin(2\pi(f_0 + k f_{\text{Sample}})t), \quad |k| \in \mathbb{N}, \text{ i.e., sin with frequency } f_0 + k f_{\text{Sample}}$$

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The sampling theorem

Theorem

A signal $s(t)$ with maximum frequency f_{MAX} can be recovered if sampled at frequency $f_s > 2 f_{\text{MAX}}$.

* Proposed by: Whittaker(s), Nyquist, Shannon, Kotel'nikov.

Nyquist frequency (rate) $f_N = 2 f_{\text{MAX}}$

Example

$$s(t) = 3 \cdot \underbrace{\cos(25 \cdot 2\pi t)}_{F_1} + 10 \cdot \underbrace{\sin(150 \cdot 2\pi t)}_{F_2} - \underbrace{\cos(50 \cdot 2\pi t)}_{F_3}$$

Condition on f_s ?

$F_1 = 25 \text{ Hz}$, $F_2 = 150 \text{ Hz}$, $F_3 = 50 \text{ Hz}$

$f_{\text{MAX}} = 150 \text{ Hz}$

$f_s > 300 \text{ Hz}$

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Frequency Domain

- Time and Frequency are two complementary signal descriptions.
The signal can be seen as projected onto the time domain or the frequency domain.
- Bandwidth indicates the width of a range in the frequency domain.
 - high bandwidth: a range located high up in the frequency domain
 - passband bandwidth: defined by a lower and upper cutoff frequency

Previous lecture:

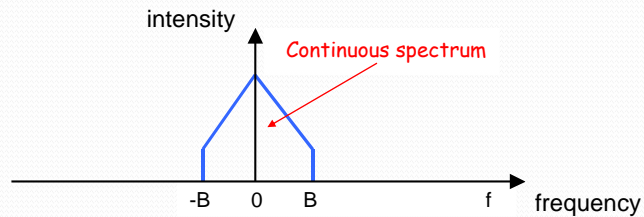
the inner-ear and early neural circuitry acts as a frequency analyser.

The audio spectrum is split into narrow bands thereby enabling detection of low-power sounds out of louder background sounds.

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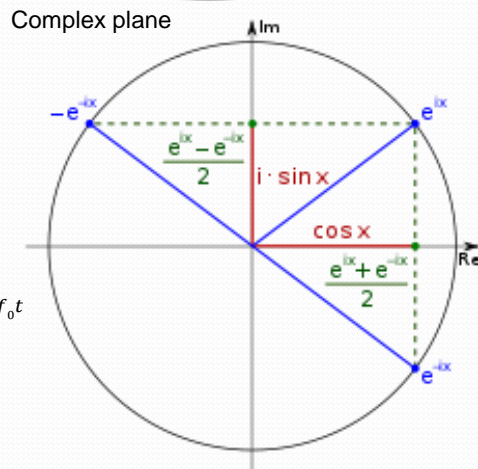
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Spectrum of band-limited signal



Spectrum of a band-limited signal:
frequency components $f \in [-B, B]$

Negative frequencies

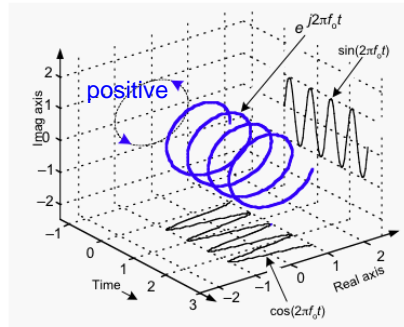


$$\cos(i2\pi f_0 t) + i \cdot \sin(i2\pi f_0 t) = e^{i2\pi f_0 t}$$

$$\sin(2\pi f_0 t) = \frac{e^{i2\pi f_0 t} - e^{-i2\pi f_0 t}}{2}$$

$$\cos(2\pi f_0 t) = \frac{e^{i2\pi f_0 t} + e^{-i2\pi f_0 t}}{2}$$

Negative frequencies

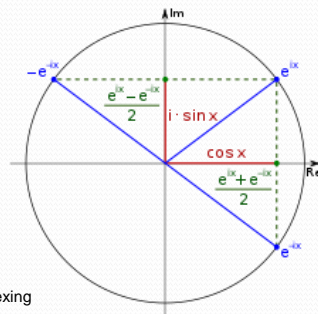


(Source: Richard Lyons)

$$\cos(i2\pi f_0 t) + i \cdot \sin(i2\pi f_0 t) = e^{i2\pi f_0 t}$$

$$\sin(2\pi f_0 t) = \frac{e^{i2\pi f_0 t} - e^{-i2\pi f_0 t}}{2i}$$

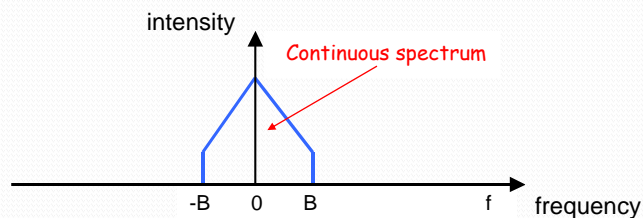
$$\cos(2\pi f_0 t) = \frac{e^{i2\pi f_0 t} + e^{-i2\pi f_0 t}}{2}$$



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Spectrum of band-limited signal

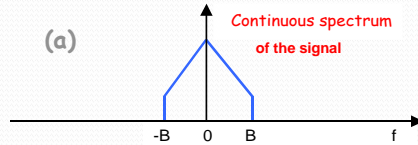


Spectrum of a band-limited signal:
frequency components $f \in [-B, B]$

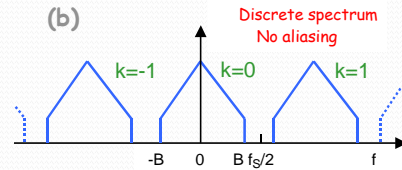
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Sampling Low-Pass Signals



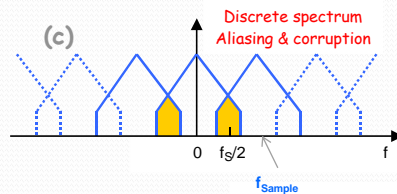
- (a) Given a band-limited signal:
frequencies of the signal in $[-B, B]$
($f_{\text{MAX}} = B$).



- (b) Time sampling with sampling frequency $f_s \Rightarrow$ frequency repetition.
 $f_s > 2B \Rightarrow$ no aliasing.

Note: $s(t)$ at f_{Sample} represents all sine-waves $s_k(t)$ defined by:

$$s_k(t) = \sin(2\pi(f_0 + k f_{\text{Sample}})t), \quad |k| \in \mathbb{N}$$

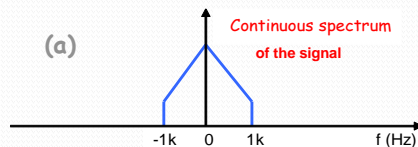


- (c) $f_s \leq 2B \Rightarrow$ **aliasing!**

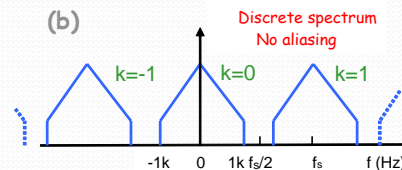
Aliasing: signal ambiguity in frequency domain

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Sampling Low-Pass Signals



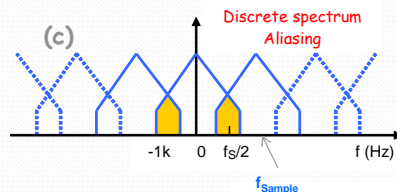
- (a) Given a band-limited signal:
frequencies of the signal in $[-1\text{kHz}, 1\text{kHz}]$ ($B = f_{\text{MAX}} = 1\text{kHz}$).



- (b) Time sampling with sampling frequency f_s
Considering the frequency repetition, as $f_s > 2B$ no aliasing occurs.

Note: $s(t)$ at f_{Sample} represents all sine-waves $s_k(t)$ defined by:

$$s_k(t) = \sin(2\pi(f_0 + k f_{\text{Sample}})t), \quad |k| \in \mathbb{N}$$



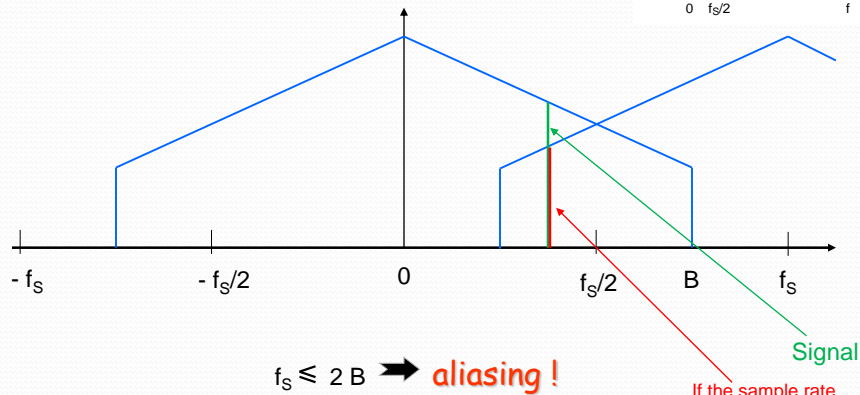
- (c) $f_s \leq 2\text{kHz} \Rightarrow$ **aliasing!**

Aliasing: signal ambiguity in frequency domain

e.g., $f_s = 600\text{ Hz} \Rightarrow$ the bin around 200Hz also gets the 800Hz components.

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Sampling Low-Pass Signals



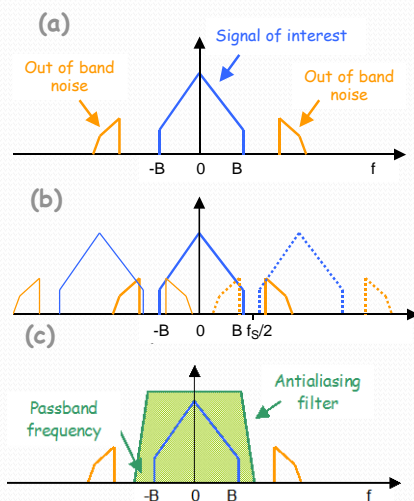
Aliasing: signal ambiguity in frequency domain

If the sample rate is too low for the bandwidth of the signal.

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Antialiasing Filter



(a), (b) Out-of-band noise can alias into band of interest. Filter it before!

Out of band noise(t) will be sampled: noise(t) @ f_s thereby mimicking a non-existing frequency within the band.

(c) Antialiasing filter

Passband: depends on bandwidth of interest.

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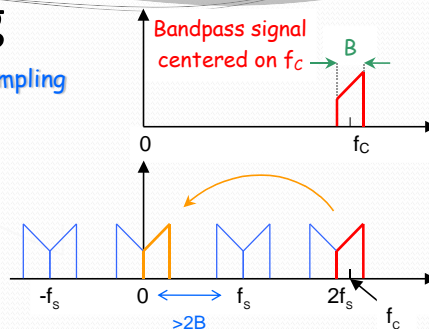
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Under-sampling

Using spectral replications to reduce sampling frequency f_s requirements.

$$\frac{2 \cdot f_c + B}{m+1} \leq f_s \leq \frac{2 \cdot f_c - B}{m}$$

$m \in \mathbb{N}$, selected so that $f_s > 2B$



Note: $s(t)$ at f_{sample} represents all sine-waves $s_k(t)$ defined by: $s_k(t) = \sin(2\pi(f_0 + k f_{\text{sample}})t)$, $|k| \in \mathbb{N}$

Example

$f_c = 20 \text{ MHz}$, $B = 5 \text{ MHz}$

Without under-sampling $f_s > 40 \text{ MHz}$.

With under-sampling:

$f_s = 22.5 \text{ MHz}$ ($m=1$)

$f_s = 17.5 \text{ MHz}$ ($m=2$)

$f_s = 11.66 \text{ MHz}$ ($m=3$) (last > 10MHz!)

Advantages

- Slower ADCs / electronics needed.
- Simpler antialiasing filters.

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Over-sampling

Oversampling : sampling at frequencies $f_s \gg 2 f_{\text{MAX}}$.

Over-sampling & averaging may improve ADC resolution

$$f_{\text{OS}} = 4^w \cdot f_s$$

f_{OS} = over-sampling frequency,
 w = additional bits required.

➡ Each additional bit implies/requires over-sampling by a factor of 4.

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(Some) ADC parameters

1. Number of bits N (~resolution)
2. Data throughput (~speed)
3. Signal-to-noise ratio (SNR)
4. Signal-to-noise-&-distortion rate
5. Effective Number of Bits (ENOB)
6. ...

$$\text{SINAD} = \frac{P_{\text{signal}} + P_{\text{noise}} + P_{\text{distortion}}}{P_{\text{noise}} + P_{\text{distortion}}}$$

Static distortion

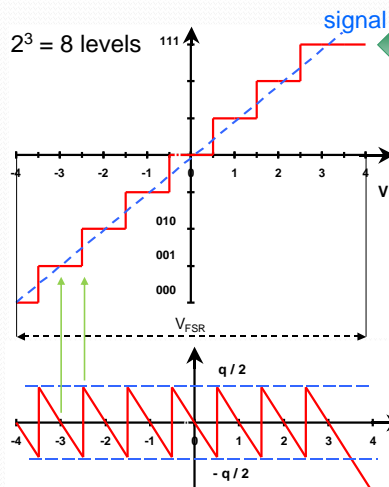
NB: Definitions may be slightly manufacturer-dependent!

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ADC - Number of bits N

Continuous input signal digitized into 2^N levels.



Uniform, bipolar transfer function
(number of bits $N=3 \Rightarrow 8$ levels)

Quantisation step $q = \frac{V_{\text{max}}}{2^N}$

Ex: $V_{\text{max}} = 1\text{V}$, $N = 12 \Rightarrow q = 244.1 \mu\text{V}$

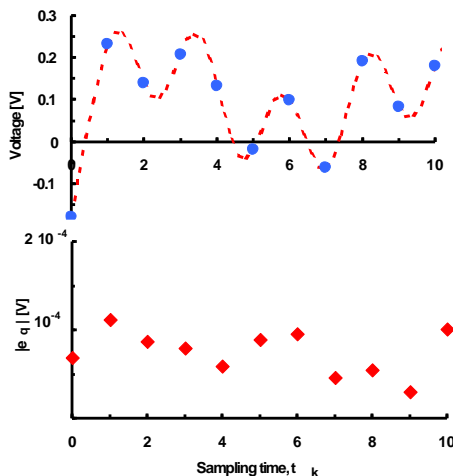
- Voltage (= q)
- Scale factor (= $1 / 2^N$)
- Percentage (= $100 / 2^N$)

Quantisation error

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ADC - Quantisation error



Quantisation step $q = \frac{V_{\max}}{2^N}$

- Quantisation Error e_q in $[-0.5q, +0.5q]$.
- e_q limits ability to resolve small signal.
- Higher resolution (more bits) means lower e_q .

QE for
N = 12
 $V_{FS} = 1$

SNR of ideal ADC

$$\overline{\text{SNR}}_{\text{ideal}} = 20 \cdot \log_{10} \left(\frac{\text{RMS}(\text{input})}{\text{RMS}(e_q)} \right) \quad (1)$$

Also called SQNR
(signal-to-quantisation-noise ratio)

(RMS = root mean square)

$$\text{RMS}(\text{input}) = \sqrt{\frac{1}{T} \cdot \int_0^T \left(\frac{V_{\text{FSR}}}{2} \cdot \sin(\omega t) \right)^2 dt} = \frac{V_{\text{FSR}}}{2\sqrt{2}}$$

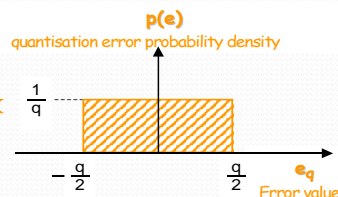
$$\text{RMS}(e_q) = \sqrt{\int_{-q/2}^{q/2} e_q^2 \cdot p(e_q) de_q} = \frac{q}{\sqrt{12}} = \frac{V_{\text{FSR}}}{2^N \cdot \sqrt{12}}$$

(sampling frequency $f_s = 2 f_{\text{MAX}}$)

Assumptions

- Ideal ADC:
- > only quantisation error e_q
 - ($p(e)$ = quantisation error probability density is assumed to be constant, uniform, etc.)
 - > e_q uncorrelated with signal.
 - > ADC performance constant in time.

Input(t) = $\frac{1}{2} V_{\text{FSR}} \sin(\omega t)$.



SNR of ideal ADC

Substituting in (1) =>

$$\overline{\text{SNR}}_{\text{ideal}} = 6.02 \cdot N + 1.76 [\text{dB}] \quad (2)$$

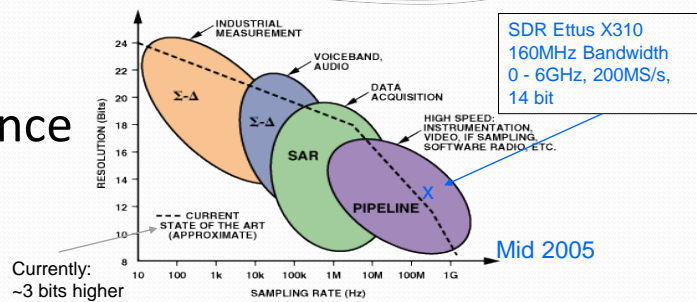
One additional bit ➡ SNR increased by 6 dB

Real SNR lower because:

- Real signals have noise.
- Forcing input to full scale unwise.
- Real ADCs have additional noise (aperture jitter, non-linearities etc).

Actually (2) needs correction factor depending on **ratio between sampling freq & Nyquist freq**. Processing gain due to oversampling.

ADC Performance



From: <http://www.analog.com/library/analogDialogue/archives/39-06/architecture.html>

ADC Topology	Resolution Range (bits)	Maximum Sampling Rate	Physical Size
Flash, direct, or parallel	≤ 8	5 GS/s	Increases exponentially with resolution
Two-step, sub-ranging, or pipelined	8-15	100 MS/s	Increases linearly with resolution
Successive approximation	8/18	2 MS/s	Increases linearly with resolution
Dual slope	12-16	100 KS/s	Does not change with resolution
Sigma delta	16-24	15 KS/s	Does not change with resolution

Table 2: ADC operating characteristics for the five ADC types discussed in this article. (Image source: Digi-Key Electronics) (2018)

Complex Numbers

The *complex numbers* are given by:

$$\mathbb{C} = \{c \mid c = a + bi, \text{ where, } a, b \in \mathbb{R}\}$$

- here i is the imaginary unit that satisfies: $i^2 = -1$
- a is called the real part of c
- b is called the imaginary part of c

If $z = x + yi$, then the *complex conjugate* z^* is defined as $z^* = x - yi$

Complex Numbers

(see also Wikipedia, and/or your Calculus Book)

The *complex numbers* are given by:

$$\mathbb{C} = \{c \mid c = a + bi, \text{ where, } a, b \in \mathbb{R}\}$$

- here i is the imaginary unit that satisfies: $i^2 = -1$

Addition:

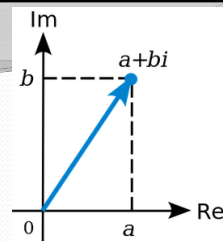
$$(a + bi) + (c + di) = (a + c) + (b + d)i$$

$$(a + bi) - (c + di) = (a - c) + (b - d)i$$

Multiplication:

$$(a + bi)(c + di) = (ac - bd) + (bc + ad)i$$

$$\frac{a + bi}{c + di} = \frac{(a + bi)(c - di)}{(c + di)(c - di)} = \left(\frac{ac + bd}{c^2 + d^2} \right) + \left(\frac{bc - ad}{c^2 + d^2} \right)i$$



Complex Numbers

The *complex numbers* are given by:

$$\mathbb{C} = \{c \mid c = a + bi, \text{ where } a, b \in \mathbb{R}\}$$

The *absolute value (modulus; magnitude)* of $z = x + yi$ is:

$$r = |z| = \sqrt{x^2 + y^2}$$

Note that:

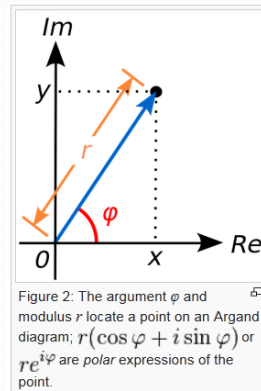
$$|z|^2 = zz^* = x^2 + y^2$$

The *argument (phase)* of $z = x + yi$ is:

$$\varphi = \arg(z) = \{\arctan(y/x), \text{ if } \dots =$$

"the angle of the vector (x,y) with
the positive real axis"

Note: $z = r(\cos\varphi + i\sin\varphi) = re^{i\varphi}$



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Complex Numbers

Let:

$$z_1 = r_1(\cos\varphi_1 + i\sin\varphi_1) = r_1e^{i\varphi_1}$$

$$z_2 = r_2(\cos\varphi_2 + i\sin\varphi_2) = r_2e^{i\varphi_2}$$

Note:

$$\cos(a) \cos(b) - \sin(a) \sin(b) = \cos(a + b)$$

$$\cos(a) \sin(b) + \sin(a) \cos(b) = \sin(a + b)$$

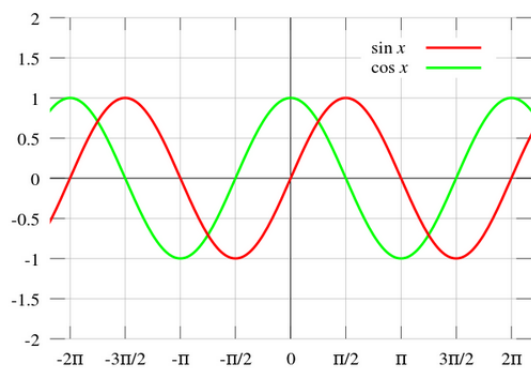
Hence:

$$z_1 z_2 = r_1 r_2 (\cos(\varphi_1 + \varphi_2) + i\sin(\varphi_1 + \varphi_2)) = r_1 r_2 e^{i(\varphi_1 + \varphi_2)}$$

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Sine Cosine Graphs



$$\sin(\varphi + \pi/2) = \cos(\varphi)$$

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From: S. Lang, Linear Algebra, 2nd Ed.
Addison-Wesley Publ. Comp., Reading,
1970.

Fields

$K \subseteq \mathbb{C}$ is a field if it satisfies:

a) If $x, y \in K$, then $x+y \in K$ and $xy \in K$

b) If $x \in K$, then $-x \in K$ and if $x \neq 0$ also,
then $x^{-1} \in K$

c) $0 \in K$ and $1 \in K$ (additive and multiplicative
null-elements, resp.)

closed under
addition and
multiplication

has inverses

has null-elements

Examples of Fields:

- \mathbb{Q} , \mathbb{R} , and \mathbb{C}
- Note, \mathbb{Z} is not a field

Vector Spaces

Field

V is a vector space over K if the following is true:

- if $u, v \in V$, then $u+v \in V$

- if $u \in V$ and $\lambda \in K$, then $\lambda u \in V$

we should define
these operations
 $u+v$ and
 λu

1) $u, v, w \in V$, then $(u+v)+w = u+(v+w)$

2) $\exists 0 \in V$ such that $0+u = u+0 = u \quad \forall u \in V$

3) given $u \in V$, $\exists -u \in V$ such that $u+(-u) = 0$

4) $\forall u, v \in V$: $u+v = v+u$

5) $\forall c \in K$: $c(u+v) = cu + cv$ for all $u, v \in V$

6) $\forall a, b \in K$: $(a+b)v = av + bv$ for all $v \in V$

7) $\forall a, b \in K$: $(ab)v = a(bv)$ for all $v \in V$

8) $\forall u \in V$ $1 \cdot u = u$ (where $1 \in K$)

Examples of Vector Spaces

- $\mathbb{R}, \mathbb{R}^2, \mathbb{R}^3, \dots, \mathbb{C}, \mathbb{C}^2, \mathbb{C}^3, \dots$

Examples of Vector Spaces (cont'd)

Function Spaces

Let S be a set and K a field and $f: S \rightarrow K$ a K -valued function, i.e., a rule that associates to each element of S a unique element of K .

Let V be the set of all functions of S into K .

→ A) If $f, g \in V$ we define $f+g$ as the function whose value at $x \in S$ is the value $f(x)+g(x)$ (again $\in K$ as K is a field).

→ B) If $c \in K$, we define cf to be the function whose value at $x \in S$ is equal to $cf(x)$ (again $\in K$ as K is a field).

- Now it is easy to verify that V is a Vector Space over K .

[$f_0: S \rightarrow K$ where $f_0(x)=0$ for all $x \in S$ is the 0 element]

Other Examples of Function Spaces which are Vector Spaces:

- V the set of all functions of \mathbb{R} into \mathbb{R}
- V the set of all continuous functions of \mathbb{R} into \mathbb{R}
- V the set of all differentiable functions of \mathbb{R} into \mathbb{R}
- V the subspace generated by the functions $f(t) = e^t$ and $g(t) = e^{2t}$ (for all $t \in \mathbb{R}$.)

↓ ↓

[Just check that A and B hold. As \mathbb{R} is a field the claim that V is vector space follows.].

Linearly Dependence

Let V be a vector space over the field K .

Let $v_1, \dots, v_n \in V$. v_1, \dots, v_n are linearly dependent over K

if $\exists a_1, \dots, a_n \in K$ not all equal 0 such that $a_1 v_1 + \dots + a_n v_n = 0$.

- If there do not exist such numbers, i.e., if $a_1, \dots, a_n \in K$ such that $a_1 v_1 + \dots + a_n v_n = 0$, then $a_i = 0 \ \forall i = 1, \dots, n$, then v_1, \dots, v_n are linearly independent.

Example:- Let $V = \mathbb{R}^n$, then $E_1 = (1, 0, \dots, 0)$ are linearly independent. $E_n = (0, 0, \dots, 1)$

- also e^t, e^{2t} are linearly independent.

Basis

If elements $v_1, \dots, v_n \in V$ generate V , and v_1, \dots, v_n are linearly independent, $\{v_1, \dots, v_n\}$ is called a basis of V .

- $v_1, \dots, v_n \in V$ generate V , that is every element of V can be expressed as a linear combination of v_1, \dots, v_n .
- and indeed if $x_1 v_1 + \dots + x_n v_n = x = y_1 v_1 + \dots + y_n v_n$ with $x_1, \dots, x_n, y_1, \dots, y_n \in K$ and for $\forall x \in V$,

$$\text{then } (x_1 - y_1)v_1 + \dots + (x_n - y_n)v_n = 0$$

$$\text{thus } x_1 = y_1, \dots, x_n = y_n.$$

\Rightarrow in a unique way

Scalar Products

Let V a vector space over a field K . (next)

A scalar product on V is an association which to any pair $v, w \in V$ associates a scalar $\langle v, w \rangle$ (also $v \cdot w$)

satisfying:

$$1) \quad \forall v, w \in V \quad \langle v, w \rangle = \langle w, v \rangle$$

$$2) \quad \text{let } u, v, w \in V, \text{ then } \langle u, v+w \rangle = \langle u, v \rangle + \langle u, w \rangle$$

$$3) \quad \text{let } \lambda \in K, \text{ then } \langle \lambda u, v \rangle = \lambda \langle u, v \rangle$$

$$\text{and } \langle u, \lambda v \rangle = \lambda \langle u, v \rangle$$

A scalar product is non-degenerate, if also:

$$4) \quad \text{If } v \in V \text{ and } \langle v, w \rangle = 0 \text{ for all } w \in V, \text{ then } v = 0.$$

Examples of Scalar Products:

- $V = K^n$ $\langle x, y \rangle: x, y \rightarrow x \cdot y = x_1 y_1 + x_2 y_2 + \dots + x_n y_n$ is a scalar product. [this is the 'standard' dot-product].

- let V be the space of continuous real-valued functions on the interval $[0, 1]$. If $f, g \in V$,

$$\langle f, g \rangle = \int_0^1 f(t)g(t) dt.$$

Then $\langle f, g \rangle$ is a scalar product. Homework

Orthogonality

$v, w \in V$ are orthogonal: $v \perp w$ if $\langle v, w \rangle = 0$.

Norm

= The norm of $v \in V$ can be defined by $\|v\| = \sqrt{\langle v, v \rangle}$

It is clear that: $\|cv\| = |c| \|v\|$

- $v \in V$ is a unit vector if $\|v\| = 1$

($v/\|v\|$ is always a unit vector, if $v \neq 0$)

Some Theorems (easy)

We have the following theorems:

Th. If $v, w \in V$ and $v \perp w$ (i.e. $\langle v, w \rangle = 0$), then (Pythagoras)

$$\|v + w\|^2 = \|v\|^2 + \|w\|^2$$

$$\begin{aligned} \text{Proof: } \|v + w\|^2 &= \langle v + w, v + w \rangle = \langle v, v \rangle + 2\langle v, w \rangle + \langle w, w \rangle \\ &= \|v\|^2 + \|w\|^2 \quad (\text{as } \langle v, w \rangle = 0). \end{aligned}$$

Parallelogram law:

$$\forall v, w \in V \quad \text{we have} \quad \|v + w\|^2 + \|v - w\|^2 = 2\|v\|^2 + 2\|w\|^2.$$

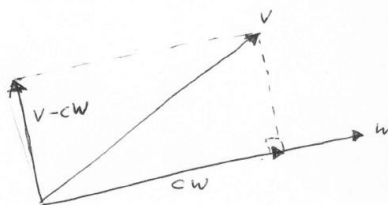
Homework: Proof Parallelogram Law.

Fourier Coefficients

Observation:

Let $w \in V$ such that $\|w\| \neq 0$.

For any $v \in V$ there exists a unique $c \in K$ such that $v - cw$ is perpendicular to w .



Now $v - cw$ perpendicular to w means that $\langle v - cw, w \rangle = 0$.

$$\Rightarrow \langle v, w \rangle - c \langle w, w \rangle = 0 \Rightarrow c = \frac{\langle v, w \rangle}{\langle w, w \rangle}$$

Conversely, if $c = \frac{\langle v, w \rangle}{\langle w, w \rangle}$ then $\langle w, w \rangle = \langle v, w \rangle \Rightarrow$

$$\langle v, w \rangle - c \langle w, w \rangle = 0 \Rightarrow \langle v - cw, w \rangle = 0$$

hence $v - cw$ perpendicular to w .

We call c the component of v along w , or the Fourier coefficient of v with respect to w .

Example Fourier Coefficients.

Let V be the space of continuous functions on $[-\pi, \pi]$.

Let $f: x \rightarrow \sin kx$, where $k \in \mathbb{Z}_{>0}$.

$$\text{Then } \|f\| = \sqrt{\langle f, f \rangle} = \left(\int_{-\pi}^{\pi} \sin^2 kx \, dx \right)^{1/2} = \sqrt{\pi}$$

In this case, if g is any continuous function on $[-\pi, \pi]$, then the Fourier coefficient of g with respect to f is

$$\frac{\langle g, f \rangle}{\langle f, f \rangle} = \frac{1}{\pi} \int_{-\pi}^{\pi} g(x) \sin kx \, dx$$

The Complex (\mathbb{C}) Case

Let V be a vector space over the complex numbers.

A hermitian product on V is a rule $\langle v, w \rangle$

satisfying. 1) $\langle v, w \rangle = \overline{\langle w, v \rangle}$ for all $v, w \in V$

2) $u, v, w \in V$, then $\langle u, v+w \rangle = \langle u, v \rangle + \langle u, w \rangle$

3) if $\alpha \in \mathbb{C}$, then $\langle \alpha u, v \rangle = \alpha \langle u, v \rangle$

$$\langle u, \alpha v \rangle = \overline{\alpha} \langle u, v \rangle.$$

$\langle \cdot, \cdot \rangle$ is positive definite

if $\langle v, v \rangle \geq 0$ for all $v \in V$ and

$\langle v, v \rangle > 0$ if $v \neq 0$.

Note

Orthogonal, perpendicular, orthogonal basis,
orthogonal complement, as before!

Also the Fourier coefficient and the
projection of v along w are as before.

Example

Let V be the space of continuous complex-valued functions on the interval $[-\pi, \pi]$.

- If $f, g \in V$, we define $\langle \cdot, \cdot \rangle$ as follows:

$$\langle f, g \rangle = \int_{-\pi}^{\pi} f(t) \overline{g(t)} dt$$

This can be shown, using standard properties of the integral, to be a positive definite hermitian product.

- Let $f_n(t) = e^{int}$

$$\text{A) if } n \neq m, \text{ then } \langle f_n, f_m \rangle = \int_{-\pi}^{\pi} e^{int} \overline{e^{imt}} dt = \int_{-\pi}^{\pi} e^{i(n-m)t} dt = 0$$

$$\text{if } n=m, \text{ then } \langle f_n, f_n \rangle = \int_{-\pi}^{\pi} e^{int} \overline{e^{int}} dt = \int_{-\pi}^{\pi} 1 dt = 2\pi$$

- If $f \in V$, then its Fourier coefficient with respect to f_n is equal to:

$$\frac{\langle f, f_n \rangle}{\langle f_n, f_n \rangle} = \frac{1}{2\pi} \int_{-\pi}^{\pi} f(t) e^{-int} dt.$$

Note: A) shows that f_n and f_m with $n \neq m$ are orthogonal.

Furthermore it can be shown that $\{f_n\}, n \in \mathbb{N}^+$ constitutes a basis for V .

Hence $\{e^{it}, e^{2it}, e^{3it}, \dots\}$ is an orthogonal

basis of V the vector space of continuous complex-valued functions on the interval $[-\pi, \pi]$. (Note, by dividing through $\langle f_n, f_n \rangle$ you get normalized basis.) \square

References

This presentation uses a selection of slides that are adapted from original slides by Dr M.E. Angoletta at DISP2003, a DSP course given by CERN and University of Lausanne (UNIL)

Resonances in Outer Ear

The **outer** ear consists of the external visible part and the auditory canal. The tube is about 2.5 cm long

Ear is closed tube (closed to one end):

⇒ resonance of 0.25 wavelength

⇒ Resonance frequencies f can be calculated with:

$$f = nv/(4L), \text{ where } n = 1, 3, 5, \dots, L=2.5\text{cm}$$

and $v = 343 \text{ m/s}$ speed of sound

For $n = 1$, $v = 343\text{m/s} = 34300 \text{ cm/s}$, $L = 2.5 \text{ cm}$, we have

$$f = 34300 \text{ (cm/s)} / 10 \text{ (cm/s)} = 3430 \text{ Hz}$$

Note wavelength of 3430 Hz equals $34300 / 3430 \text{ cm} = 10\text{cm}$