

Analog and Digital Signals

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Analog and Digital Signals

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

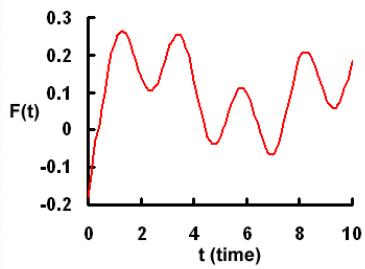
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Analog and Digital Signals

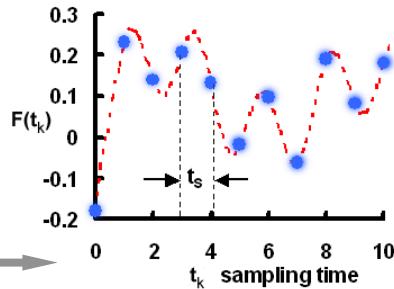
Analog Signals

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 sampling time is $t_s = 0.001$ sec => $f_s = 1000$ Hz

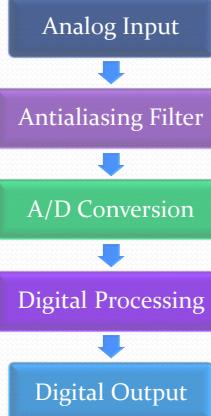
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ADC System Implementation

Important issues:

Analysis bandwidth, Dynamic range



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

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Sampling Rate

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 => wheels appear to go clockwise
- Train accelerates => wheels go counter clockwise

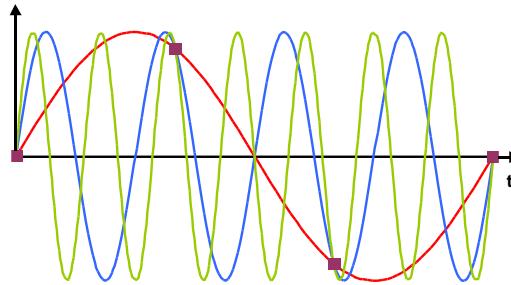
Rotating propeller of an airplane captured by a Mobile phone camera.



Both examples: Low sampling frequency leading to Frequency misidentification

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

Sampling a Sine Wave



— $s(t) = \sin(2\pi f_0 t)$

■ $s(t) @ f_{\text{Sample}}$

For example:

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

— $s_1(t) = \sin(2\pi 4t)$

— $s_2(t) = \sin(2\pi 7t)$

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

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

The sampling theorem

Theorem

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

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

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

Example

$$s(t) = 3 \cdot \cos(25 \cdot 2\pi t) + 10 \cdot \sin(150 \cdot 2\pi t) - \cos(50 \cdot 2\pi t) \quad \text{Condition on } f_s?$$

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

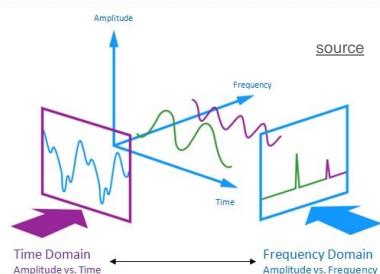
$$f_{MAX}$$

$$f_s > 300 \text{ Hz}$$

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.



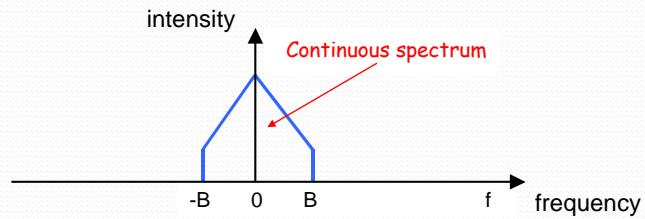
- Bandwidth indicates the width of a range in the frequency domain.
 - high bandwidth: a range located high up in the frequency domain
 - passband bandwidth: defined by a lower and upper cutoff frequency

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.

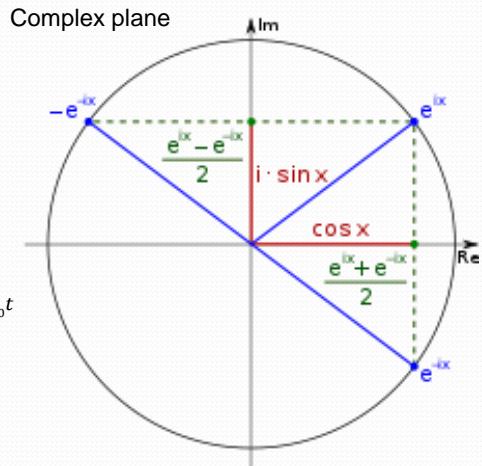
Spectrum of band-limited signal



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

Negative frequencies

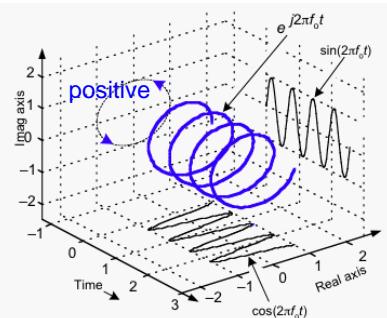
$$\cos(2\pi f_0 t) + i \cdot \sin(2\pi f_0 t) = e^{i2\pi f_0 t}$$



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

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

Negative frequencies

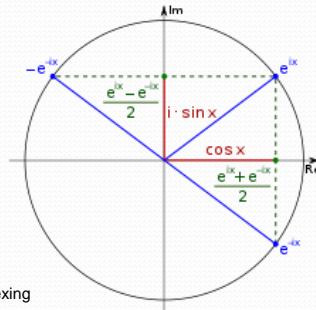


(Source: Richard Lyons)

$$\cos(2\pi f_0 t) + i \cdot \sin(2\pi f_0 t) = e^{i2\pi f_0 t}$$

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

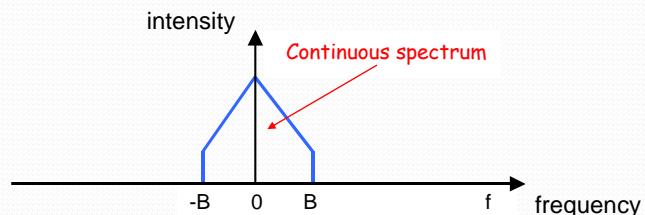
$$\cos(2\pi f_0 t) = \frac{e^{i2\pi f_0 t} + e^{-i2\pi f_0 t}}{2}$$



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Spectrum of band-limited signal

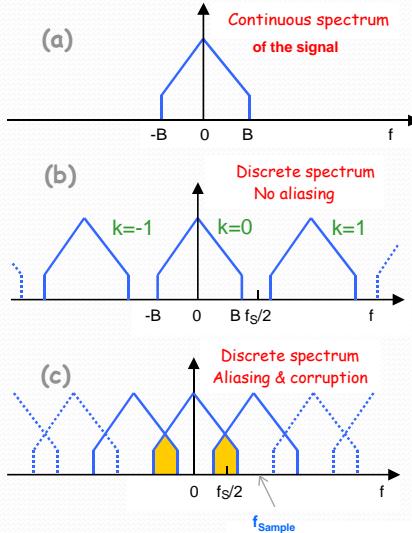


Spectrum of a band-limited signal:
the signal has frequency components $f \in [-B, B]$

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Sampling Low-Pass Signals



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

(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:

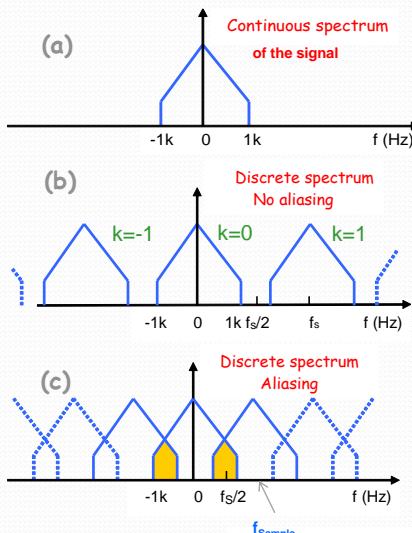
$$s_k(t) = \sin(2\pi(f_0 + k f_{Sample})t), |k| \in \mathbb{N}$$

(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 $[-1\text{kHz}, 1\text{kHz}]$ ($B = f_{MAX} = 1\text{kHz}$).

(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:

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

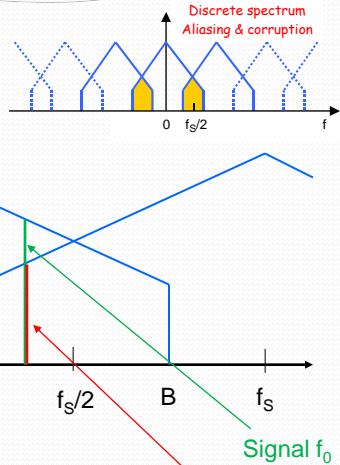
(c) $f_s \leq 2\text{kHz} \Rightarrow$ aliasing !

Aliasing: signal ambiguity in frequency domain

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

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Sampling Low-Pass Signals



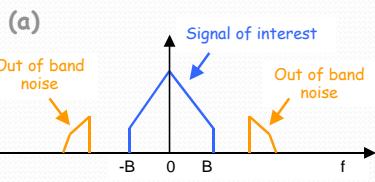
Aliasing: signal ambiguity in frequency domain

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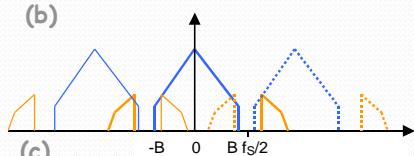
If the sample rate is too low for the bandwidth of the signal => + amplitude of signal component with freq. $f_0 + (-f_s)$

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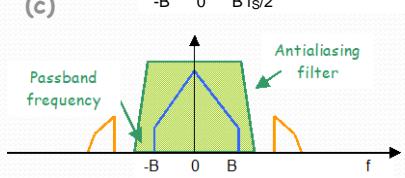
Antialiasing Filter



(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 contributions of frequencies within the band.



(c) Antialiasing filter

Passband: depends on bandwidth of interest.

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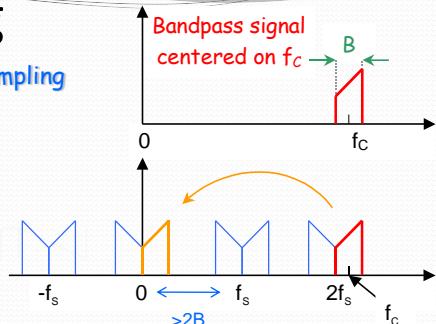
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Under-sampling

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 m such that $f_s > 2B = 10 \text{ MHz}$

Advantages

- Slower ADCs / electronics needed.

- Simpler antialiasing filters.

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

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

(Some) ADC parameters

1. Number of bits N (~resolution)
2. Sample rate (~speed)
3. Signal-to-noise ratio (SNR)
4. Signal-to-noise-&-distortion rate

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

5. Effective Number of Bits (ENOB)
6. ...

USB Audio Interface

- 24-bit
- 192 kHz

Software Defined Radio (SDR)

- 14-bit ADC
- 2 – 6 Msamples/sec
- Covers 1 kHz – 2 GHz
- Bandwidth 10MHz
- At 8-bit >9.2 MS/sec

Static distortion

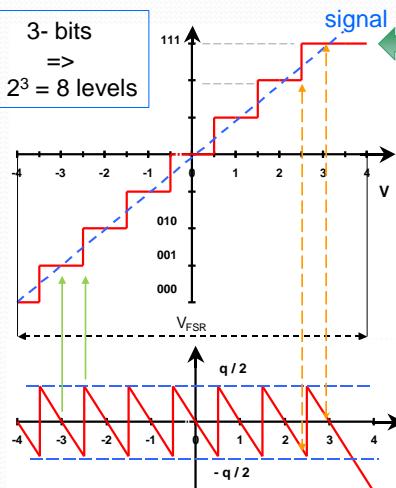
NB: Definitions may be slightly manufacturer-dependent!

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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)

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

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

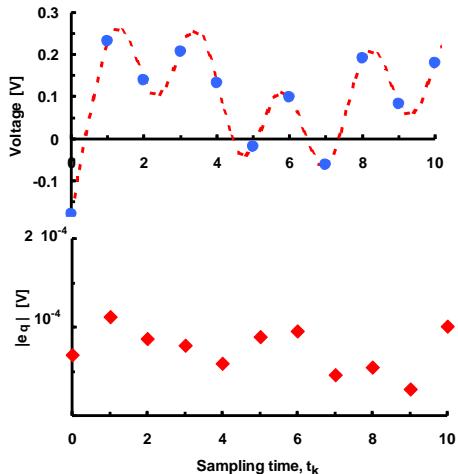
- Voltage (= q)
- Scale factor ($= 1 / 2^N$)
- Percentage ($= 100 / 2^N$)

Quantisation error digitized signal vs signal

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ADC - Quantisation error



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

- Quantisation Error e_q in $[-0.5 q, +0.5 q]$.
- 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

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

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

RMS = root mean square

FSR = Full Scale Range

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



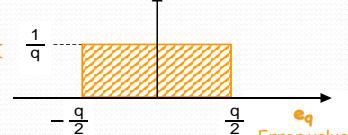
$$\text{Input}(t) = \frac{1}{2} V_{\text{FSR}} \sin(\omega t).$$

$p(e)$

quantisation error probability density

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

(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.

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:

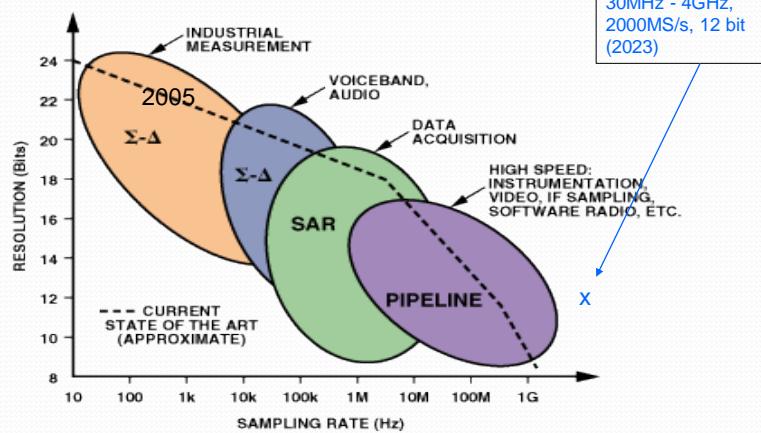
- Real signals have noise.
- Forcing input to full scale unwise.
- Real ADCs have additional noise (aperture jitter, non-linearities etc).

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

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ADC Performance



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

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Complex Numbers

The **complex numbers** are given by:

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

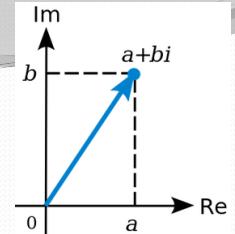
- here i is the imaginary unit that satisfies: $i^2 = -1$
- a is called the real part of c
- b is called the imaginary part of c

If $z=a+bi$, then the **complex conjugate** z^*

is defined as $z^*=a-bi$

Complex Numbers

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



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:

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

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

Complex Numbers

The **complex numbers** are given by:

$$\mathbb{C} = \{c \mid c = x + yi, \text{ where } x, y \in \mathbb{R}\}$$

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:

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

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

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

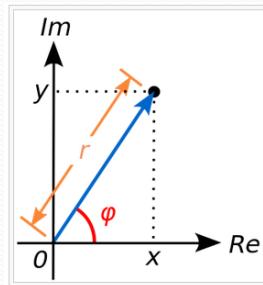


Figure 2: The argument φ and modulus r locate a point on an Argand diagram; $r(\cos\varphi + i\sin\varphi)$ or $re^{i\varphi}$ are polar expressions of the point.

Complex Numbers

Let:

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

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

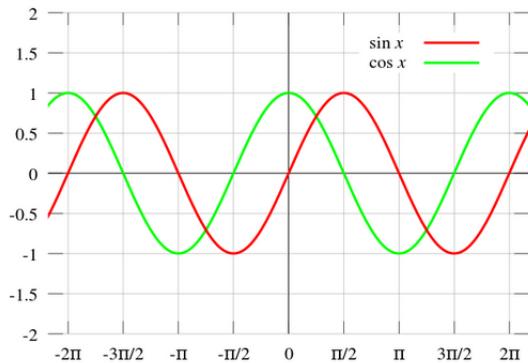
Note:

$$\begin{aligned}\cos(a)\cos(b) - \sin(a)\sin(b) &= \cos(a + b) \\ \cos(a)\sin(b) + \sin(a)\cos(b) &= \sin(a + b)\end{aligned}$$

Hence:

$$z_1z_2 = r_1r_2(\cos(\varphi_1+\varphi_2) + i\sin(\varphi_1+\varphi_2)) = r_1r_2e^{i(\varphi_1+\varphi_2)}$$

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 $xy \in K$

closed under
addition and
multiplication

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

has inverses

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

has null-elements

Examples of Fields:

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

Vector Spaces

a 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 $\lambda u \in V$
- associative** 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 $u+(-u) = 0$
- commutative** 4) $\forall u, v \in V: u+v = v+u$
- distributive** 5) $\forall c \in K: c(u+v) = cu+cv \quad \text{for all } u, v \in V$
- 6) $\forall a, b \in K: (a+b)v = av + bv \quad \text{for all } v \in V$
- 7) $\forall a, b \in K: (ab)v = a(bv) \quad \text{for all } v \in V$
- 8) $\forall u \in V \quad 1 \cdot u = u \quad (\text{where } 1 \in K)$

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

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).

- Then 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 $a_1v_1 + \dots + a_nv_n = 0$.

- If there do not exist such numbers, i.e., if $a_1, \dots, a_n \in K$ such that $a_1v_1 + \dots + a_nv_n = 0$, then $a_i = 0 \quad \forall i = 1, \dots, n$. then v_1, \dots, v_n are linearly independent.

Example:- Let $V = \mathbb{R}^n$, then $E_i = (1, 0, \dots, 0)$ 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 all $x \in V$,
- then $(x_1 - y_1)v_1 + \dots + (x_n - y_n)v_n = 0$
thus $x_1 = y_1, \dots, x_n = y_n$. => in a unique way

Scalar Products

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

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

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

3) let $c \in K$, then $\langle cu, v \rangle = c \langle u, v \rangle$

and $\langle u, cv \rangle = c \langle u, v \rangle$

A scalar product is non-degenerate, if also:

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

Examples of Scalar Products:

- $V = K^n \quad \langle x, y \rangle: x, y \rightarrow x \cdot y = x_1 y_1 + x_2 y_2 + \dots + x_n y_n$
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 I: Proof

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)

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

Proof: $\|v+w\|^2 = \langle v+w, v+w \rangle = \langle v, v \rangle + 2\langle v, w \rangle + \langle w, w \rangle$
 $= \|v\|^2 + \|w\|^2$ (as $\langle v, w \rangle = 0$). \square

Parallelogram law:

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

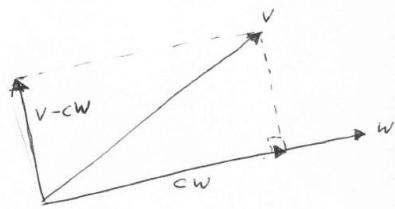
[Homework II: Proof Parallelogram Law.](#)

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 $c \langle w, w \rangle = \langle v, w \rangle \Rightarrow$

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

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.

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

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

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

$$\langle \cdot, \cdot \rangle \text{ is positive definite} \quad \langle u, \alpha v \rangle = \bar{\alpha} \langle u, v \rangle.$$

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

$$\langle v, v \rangle > 0 \quad \text{if } v \neq 0.$$

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

$$\text{1) if } n \neq m, \text{ then } \langle f_n, f_m \rangle = \int_{-\pi}^{\pi} e^{int} \overline{e^{imt}} dt = \int_{-\pi}^{\pi} e^{ikt} dt = 0$$

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

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

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

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 $\{e^{jnt}, e^{jmt}, e^{jst}, \dots\}$ 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.)

References

This presentation uses a selection of slides that are adapted from original slides by

Dr M.E. Angloletta 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 $34300 / 3430 \text{ cm} = 10\text{cm}$

*)

$$\begin{aligned} \int_{-\pi}^{\pi} \sin^2 kx \, dx &= \left[\frac{x}{2} - \frac{\sin(2kx)}{4k} \right]_{-\pi}^{\pi} \\ &= \frac{\pi}{2} - \frac{\sin(2k\pi)}{4k} - \left(-\frac{\pi}{2} - \frac{\sin(-2k\pi)}{4k} \right) \\ &= \pi \end{aligned}$$

Recap:

Complex numbers \mathbb{C} :

1) $\mathbb{R} \subset \mathbb{C}$; sum and products for these numbers $\in \mathbb{R} \subset \mathbb{C}$ as before.

2) There is complex number i such that $i^2 = -1$

3) Every complex number can be uniquely expressed as $a+bi$, with $a, b \in \mathbb{R}$

4) $\alpha, \beta, \gamma \in \mathbb{C}$, then $(\alpha\beta)\gamma = \alpha(\beta\gamma)$

$$(\alpha + \beta) + \gamma = \alpha + (\beta + \gamma)$$

$$\alpha(\beta + \gamma) = \alpha\beta + \alpha\gamma$$

$$(\beta + \gamma)\alpha = \beta\alpha + \gamma\alpha$$

$$\alpha\beta = \beta\alpha$$

$$\alpha + \beta = \beta + \alpha$$

If $1 \in \mathbb{R}$ then $1\alpha = \alpha$

If $0 \in \mathbb{R}$ then $0\alpha = 0$

Furthermore, $\alpha + (-1)\alpha = 0$.

$$\alpha = a+bi, \beta = c+di, \text{ then } \alpha + \beta = (a+bi) + (c+di) = (a+c) + (b+d)i$$

$$\text{and } \alpha \cdot \beta = (a+bi)(c+di)$$

$$= ac + adi + bci + bdi^2$$

$$= (ac - bd) + (ad + bc)i$$

$$\text{, and if } \lambda \in \mathbb{R} \quad \lambda\alpha = \lambda(a+bi) = \lambda a + \lambda bi$$

$$\bar{\alpha} = \overline{a+bi} = a-bi \Rightarrow \alpha\bar{\alpha} = a^2 + b^2 \in \mathbb{R}$$

$\bar{\alpha}$ is conjugate of α

$$\Rightarrow \alpha^{-1} = \frac{\bar{\alpha}}{\alpha\bar{\alpha}}$$

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