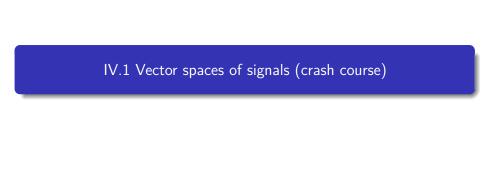


Chapter IV: The Fourier Transform and its

applications



Vector spaces

- **Vector space** = a set $V\{v_i\}$ with the following two properties:
 - ▶ one element + another element = still an element of the same space
 - ightharpoonup a scalar constant imes an element = still an element of the same space
- ► You can't escape a vector space by summing or scaling
- ▶ The elements of a vector space are called **vectors**

Examples of vector spaces

- Geometric spaces are great intuitive examples:
 - ightharpoonup a line, or the set \mathbb{R} (one-dimensional)
 - ightharpoonup a plane, or the set $\mathbb C$ (two-dimensional)
 - ► 3D space (three-dimensional)
 - ▶ 4D space (four-dimensional, like the spatio-temporal universe)
 - arrays with N numbers (N-dimensional)
 - lacktriangle space of continuous signals (∞ -dimensional)
- ► The **dimension** of the space = "how many numbers you need in order to specify one element" (informal)
- ➤ A "vector" like in maths = a sequence of N numbers = a "vector" like in programming
 - ightharpoonup e.g. a point in a plane has two coordinates = a vector of size N=2
 - e.g. a point in a 3D-space has three coordinates = a vector of size N=3

Inner product

- Many vector spaces have a fundamental operation: the (Euclidean) inner product
 - for discrete signals:

$$\langle \mathbf{x}, \mathbf{y} \rangle = \sum_{i} x_{i} y_{i}^{*}$$

for continuous signals:

$$\langle \mathbf{x}, \mathbf{y} \rangle = \int x(t) y^*(t)$$

- * represents complex conjugate (has no effect for real signals)
- ► The result is one number (real or complex)
- Also known as dot product or scalar product ("produs scalar")

Inner product

- ► Each entry in **x** times the complex conjugate of the one in **y**, all summed
- ightharpoonup For discrete signals, it can be understood as a row imes column multiplication
- Discrete vs continuous: just change sum/integral depending on signal type

Inner product properties

▶ Inner product is **linear** in both terms:

$$\begin{split} \langle \mathbf{x_1} + \mathbf{x_2}, \mathbf{y} \rangle &= \langle \mathbf{x_1}, \mathbf{y} \rangle + \langle \mathbf{x_2}, \mathbf{y} \rangle \\ \langle c \cdot \mathbf{x}, \mathbf{y} \rangle &= c \cdot \langle \mathbf{x_1}, \mathbf{y} \rangle \\ \langle \mathbf{x}, \mathbf{y_1} + \mathbf{y_2} \rangle &= \langle \mathbf{x}, \mathbf{y_1} \rangle + \langle \mathbf{x}, \mathbf{y_2} \rangle \\ \langle \mathbf{x}, c \cdot \mathbf{y} \rangle &= c^* \cdot \langle \mathbf{x_1}, \mathbf{y} \rangle \end{split}$$

The distance between two vectors

- ► An inner product induces a **norm** and a **distance** function
- ► The (Euclidean) distance between two vectors =

$$d(\mathbf{x},\mathbf{y}) = \sqrt{(x_1 - y_1)^2 + (x_2 - y_2)^2 + \dots + (x_N - y_N)^2}$$

- ► This distance is the **usual geometric distance** you know from geometry
- lt has the exact same intuition like in **normal geometry**:
 - if two vectors have small distance, they are close, they are similar
 - two vectors with large distance are far away, not similar
 - two identical vectors have zero distance

The norm of a vector

- ► An inner product induces a **norm** and a **distance** function
- ► The norm (length) of a vector = sqrt(inner product with itself)

$$\|\mathbf{x}\| = \sqrt{\langle \mathbf{x}, \mathbf{x} \rangle} = \sqrt{x_1^2 + x_2^2 + \dots + x_N^2}$$

- ► The norm of a vector is the distance from x to point 0.
- It has the exact same intuition like in normal geometry:
 - ightharpoonup vector has large norm = has big values, is far from $\mathbf{0}$
 - ightharpoonup vector has small norm = has small values, is close to $\bf 0$
 - vector has zero norm = it is the vector 0
- Norm of a vector = sqrt(the signal energy)

Norm and distance

- The norm and distance are related
- ightharpoonup The distance between ${f a}$ and ${f b}=$ norm (length) of their difference

$$d(\mathbf{x}, \mathbf{y}) = \|\mathbf{x} - \mathbf{y}\| = \sqrt{x_1^2 + x_2^2 + \dots + x_N^2}$$

Just like in geometry: distance = length of the difference vector

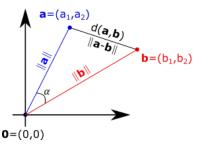


Figure 1: Norm and distance in vector spaces

Angle between vectors

► The **angle** between two vectors is:

$$cos(\alpha) = \frac{\langle x, y \rangle}{||x|| \cdot ||y||}$$

- ▶ is a value between -1 and 1
- ▶ **Otrhogonal vectors** = two vectors with $\langle x, y \rangle = 0$
 - ► their angle = 90 deg
 - in geometric language, the two vectors are **perpendicular**

Why vector space

- ► Why are all these useful?
- ► They are a very general **framework** for different kinds of signals
- ▶ We can have generic algorithms expressed in terms of distances, norms, angles, and they will work the same in all vector spaces
 - Example in DEDP class: ML decision with 1, 2, N samples

Vector spaces in DSP class

We deal mainly with the following vector spaces:

- ▶ The vector space of all infinitely-long real signals x[n]
- ▶ The vector space of all infinitely-long periodic signals x[n] with period N
 - ▶ for each *N* we have a different vector space
- ▶ The vector space of all finite-length signals x[n] with only N samples
 - ▶ for each *N* we have a different vector space

Bases

ightharpoonup A **basis** = a set of N linear independent elements from a vector space

$$\textit{B} = \{\textbf{b}^1, \textbf{b}^2...\textbf{b}^N\}$$

Any vector in a vector space is expressed as a linear combination of the basis elements:

$$\mathbf{x} = \alpha_1 \mathbf{b}^1 + \alpha_2 \mathbf{b}^2 + \dots + \alpha_N \mathbf{b}^N$$

▶ The vector is defined by these coefficients:

$$\mathbf{x} = (\alpha_1, \alpha_2, ... \alpha_N)$$

Bases and coordinate systems

- Bases are just like coordinate systems in a geometric space
 - any point is expressed w.r.t. a coordinate system

$$\mathbf{x} = x_1 \mathbf{i} + x_2 \mathbf{j}$$

any vector is expressed w.r.t. a basis

$$\mathbf{x} = \alpha_1 \mathbf{b}^1 + \alpha_2 \mathbf{b}^2 + \dots + \alpha_N \mathbf{b}^N$$

- ightharpoonup N = The number of basis elements = The dimension of the space
- Example: any color = RGB values (monitor) or Cyan-Yellow-Magenta values (printer)

Bases and coordinate systems

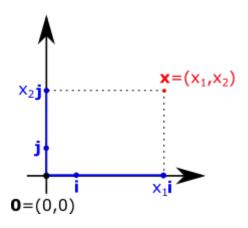


Figure 2: Basis expansion of a vector x

Choice of bases

- ▶ There is typically an infinite choice of bases
- ► The **canonical basis** = all basis vectors are full of zeros, just with one 1
- ▶ You used it already in an exercise:
 - ▶ any signal x[n] can be expressed of a sum of $\delta[n-k]$

$$\{\ldots,3,6,2,\ldots\} = \cdots + 3\delta[n] + 6\delta[n-1] + 2\delta[n-2] + \ldots$$

▶ the canonical basis is $B = \{..., \delta[n], \delta[n-1], \delta[n-2], ...\}$

Orthonormal bases

- ightharpoonup An **orthonormal basis** a basis where all elements \mathbf{b}^i are:
 - orthogonal to each other:

$$\langle \mathbf{b}^i, \mathbf{b}^j \rangle = 0, \forall i \neq j$$

normalized (their norm = 1):

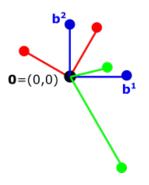
$$||\mathbf{b}^i|| = \sqrt{\langle \mathbf{b}^i, \mathbf{b}^i \rangle} = 1, \forall i$$

- **Example:** the canonical basis $\{\delta[n-k]\}$ is orthonormal:

 - $\langle \delta[n-k], \delta[n-k] \rangle = 1, \forall k$

Orthonormal bases

Orthonormal basis = like a coordinate system with orthogonal vectors, of length 1



blue = an orthonormal basis red = another orthonormal basis green = not an orthonormal basis

Figure 3: Sample bases in a 2D space

Basis expansion of a vector

- ▶ Suppose we have an **orthonormal basis** $B = \{\mathbf{b}^i\}$
- ► Suppose we have a vector **x**
- ► We can write (expand) x as:

$$\mathbf{x} = \alpha_1 \mathbf{b}^1 + \alpha_2 \mathbf{b}^2 + \dots + \alpha_N \mathbf{b}^N$$

▶ Question: how to **find** the coefficients α_i ?

Basis expansion of a vector

▶ If the basis is **orthonormal**, we have:

$$\langle \mathbf{x}, \mathbf{b}^{i} \rangle = \langle \alpha_{1} \mathbf{b}^{1} + \alpha_{2} \mathbf{b}^{2} + \dots + \alpha_{N} \mathbf{b}^{N}, \mathbf{b}^{i} \rangle$$

$$= \langle \alpha_{1} \mathbf{b}^{1}, \mathbf{b}^{i} \rangle + \langle \alpha_{2} \mathbf{b}^{2}, \mathbf{b}^{i} \rangle + \dots + \langle \alpha_{N} \mathbf{b}^{N}, \mathbf{b}^{i} \rangle$$

$$= \alpha_{1} \langle \mathbf{b}^{1}, \mathbf{b}^{i} \rangle + \alpha_{2} \langle \mathbf{b}^{2}, \mathbf{b}^{i} \rangle + \dots + \alpha_{N} \langle \mathbf{b}^{N}, \mathbf{b}^{i} \rangle$$

$$= \alpha_{i}$$

Basis expansion of a vector

Any vector **x** can be written as:

$$\mathbf{x} = \alpha_1 \mathbf{b}^1 + \alpha_2 \mathbf{b}^2 + \dots + \alpha_N \mathbf{b}^N$$

▶ For orthonormal basis: the coefficients α_i are found by inner product with the corresponding basis vector:

$$\alpha_i = \langle, \mathbf{x}, \mathbf{b}^i \rangle$$

Why bases

- How does all this talk about bases help us?
- ▶ To better understand the Fourier transform
- ▶ The signals $\{e^{j\omega n}\}$ form an **orthonormal basis**
- ▶ The Fourier Transform of a signal x =finding the coefficients of \mathbf{x} in this basis
- ► The Inverse Fourier Transform = expanding **x** with the elements of this basis
- ► Same **generic** thing every time, only the type of signals differ



Reminder

► Reminder:

$$e^{jx} = \cos(x) + j\sin(x)$$

$$\cos(x) = \frac{e^{jx} + e^{-jx}}{2}$$

$$\sin(x) = \frac{e^{jx} - e^{-jx}}{2j}$$

$$\sin(x) = \cos(x - \frac{\pi}{2})$$

$$\cos(x) = \sin(x + \frac{\pi}{2})$$

Why sinusoidal signals

- Why are sinusoidal signals sin() and cos() so prevalent in signal processing?
- Answer: because they are combinations of an e^{jx} and an e^{-jx}
- ▶ Why are these e^{jx} so special?
- Answer: because they are eigen-functions of linear and time-invariant (LTI) systems

Response of LTI systems to harmonic signals

- ▶ Consider an LTI system with h[n]
- ▶ Input signal = complex harmonic (exponential) signal $x[n] = Ae^{j\omega_0 n}$
- ► Output signal = convolution

$$y[n] = \sum_{k=-\infty}^{\infty} h[k]x[n-k]$$
$$= \sum_{k=-\infty}^{\infty} h[k]e^{-j\omega_0 k}Ae^{j\omega_0 n}$$
$$= H(\omega_0) \cdot x[n]$$

lacksquare Output signal imes a (complex) constant $(H(\omega_0))$

Eigen-function

Eigen-function of a system ("functie proprie") = a function f which, if input in a system, produces an output proportional to it

$$H{f} = \lambda \cdot f, \lambda \in \mathbb{C}$$

- lacktriangle just like **eigen-vectors** of a matrix (remember algebra): $A \mbox{$\stackrel{>}{\gtrsim}$} = \lambda \mbox{$\stackrel{>}{\approx}$}$
- we call the "functions" to allow for continuous signals as well
- ► Complex exponential signals $e^{j\omega t}$ (or $e^{j\omega n}$) are **eigen-functions** of Linear and Time Invariant (LTI) systems:
 - ightharpoonup output signal imes a (complex) constant

Representation with respect to eigen-functions (-vectors)

- We can understand the effect of a LTI system very easily if we decompose all signals x[n] as a combination of $\{e^{j\omega n}\}$
- ► Example: RGB color filter
 - suppose we have some photographic filters (lenses):
 - ▶ one reduces red to 50%
 - ▶ one reduces green to 25%
 - one reduces blue to 80%
 - RGB are eigen-functions of the system: input = 200 Blue, output = 0.8 * 200 Blue
 - what is the output color if input is "pink"?
 - Answer is easy if we represent all colors in RGB

Representation with respect to eigen-functions (-vectors)

- We can understand the effect of a LTI system **very easily** if we decompose all signals as a combination of $\{e^{j\omega n}\}$
- ► All vector space theory becomes useful now:
 - $ightharpoonup \{e^{j\omega n}\}$ is an **orthonormal basis**
 - decomposing signals = finding coefficients α_i
 - we know how to do this, just like for any orthonormal basis

$$x[n] = \sum \alpha_{\omega} \cdot e^{j\omega n}$$
$$\alpha_{\omega} = \langle x, e^{j\omega n} \rangle$$

- Consider the vector space of non-periodic infinitely-long signals
- ► This vector space is **infinite-dimensional**
- ► The signals $\{e^{j2\pi fn}\}, \forall f \in [-\frac{1}{2}, \frac{1}{2}]$ form an **orthonormal basis**
- ▶ We can expand (almost) any **x** in this basis:

$$x[n] = \int_{f=-1/2}^{1/2} \underbrace{X(f)}_{\alpha_{\omega}} e^{j2\pi f n} df$$

► The coefficient of every $e^{j2\pi fn}$ is found by inner product:

$$\alpha_{\omega} = X(f) = \langle x[n], e^{j2\pi fn} \rangle = \sum_{n} x[n]e^{-j2\pi fn}$$

Inverse Discrete-Time Fourier Transform (DTFT)

$$x[n] = \int_{f=-1/2}^{1/2} X(f)e^{j2\pi fn}df$$

▶ A signal x[n] can be written as a linear combination of $\{e^{j2\pi fn}\}, \forall f \in [-\frac{1}{2}, \frac{1}{2}],$ with some coefficients X(f)

Discrete-Time Fourier Transform (DTFT)

$$X(f) = \langle x[n], e^{j2\pi fn} \rangle = \sum_{n=-\infty}^{\infty} x[n]e^{-j2\pi fn}$$

► The coefficient X(f) of every $\{e^{j2\pi fn}\}$ is found using the inner product $\langle \mathbf{x}, e^{j2\pi fn} \rangle$

- ightharpoonup Alternative form with ω
- We can replace $2\pi f = \omega$, and $df = \frac{1}{2\pi} d\omega$

$$x[n] = \frac{1}{2\pi} \int_{\omega = -\pi}^{\pi} X(\omega) e^{j\omega n} d\omega$$

$$X(\omega) = \langle x[n], e^{j\omega n} \rangle = \sum_{n} x[n]e^{-j\omega n}$$

- ▶ A non-periodic signal x[n] has a **continuous spectrum** $X(\omega)$, with $f \in [-\frac{1}{2}, \frac{1}{2}]$
 - e.g. $\omega \in [-\pi, \pi]$

- Consider the vector space of periodic signals with period N
 - ightharpoonup for some fixed N=2, 3 or . . . etc
- ► This is a vector space of dimension N
 - we need N numbers to identify a signal (specify its period)
- ▶ We can consider x[n] only for **one period**, i.e. n = 0, ... N 1
- ► The signals $\{e^{j2\pi fn}\}, \forall f \in \{0, \frac{1}{N}, \dots \frac{N-1}{N}\}$ form an **orthonormal** basis with N elements
- ▶ It is a **discrete** set of frequencies: $f = \frac{k}{N}, \forall k \in \{0, 1, ..., N-1\}$

Discrete Fourier Transform (DFT)

Inverse Discrete Fourier Transform

$$x[n] = \frac{1}{N} \sum_{k=0}^{N-1} X_k e^{j2\pi kn/N}$$

A periodic signal x[n] can be written as a linear combination of k signals $\{e^{j2\pi kn/N}\}$, with some coefficients X_k

Discrete Fourier Transform

$$X_k = \langle x[n], e^{j2\pi fn} \rangle = \sum_{n=0}^{N-1} x[n]e^{-j2\pi kn/N}$$

► The coefficient X(f) of every $\{e^{j2\pi fn}\}$ is found using the inner product $\langle \mathbf{x}, e^{j2\pi fn} \rangle$

Discrete Fourier Transform (DFT)

- A periodic signal x[n] with period N has a **discrete spectrum** $X(\omega)$ composed of only N frequencies $\{0, \frac{1}{N} \dots \frac{N-1}{N}\}$
- ► Each frequency $\frac{k}{N}$ has a **coefficient** X_k
 - ightharpoonup also written as c_k
 - ▶ The N coefficients X_k are the equivalent of $X(\omega)$
- ▶ It is also known as the "Fourier Series for Discrete Signals"



Definition

Definitions (again):

Inverse Discrete-Time Fourier Transform (DTFT)

$$x[n] = \int_{f=-1/2}^{1/2} X(f)e^{j2\pi fn}df = \frac{1}{2\pi} \int_{\omega=-\pi}^{\pi} X(\omega)e^{j\omega n}d\omega$$

Discrete-Time Fourier Transform (DTFT)

$$X(f) = \langle x[n], e^{j2\pi fn} \rangle = \sum_{n=-\infty}^{\infty} x[n]e^{-j2\pi fn}$$

Basic properties of DTFT

- ▶ $X(\omega)$ is **defined** only for $\omega \in [-\pi, \pi]$
- ▶ $X(\omega)$ is **complex** (has $|X(\omega)|$, $\angle X(\omega)$)
- ▶ If the signal x[n] is real, $X(\omega)$ is **even**

$$x[n] \in \mathbb{R} \to X(-\omega) = X^*(\omega)$$

- ► This means:
 - ▶ modulus is even: $|X(\omega)| = |X(-\omega)|$
 - ▶ phase is odd: $X(\omega) = -X(-\omega)$

Expressing as sum of sinusoids

• Grouping terms with $e^{j\omega n}$ and $e^{j(-\omega)n}$ we get:

$$\begin{split} x[n] &= \frac{1}{2\pi} \int_{-\pi}^{0} X(\omega) e^{j\omega n} + \frac{1}{2\pi} \int_{0}^{\pi} X(\omega) e^{j\omega n} d\omega \\ &= \frac{1}{2\pi} \int_{0}^{\pi} (X(\omega) e^{j\omega n} + X(-\omega) e^{j(-\omega)n}) d\omega \\ &= \frac{1}{2\pi} \int_{0}^{\pi} 2|X(\omega)| (e^{j\omega n + \angle X(\omega)} + e^{-j\omega n - \angle X(\omega)}) d\omega \\ &= \frac{1}{2\pi} \int_{0}^{\pi} 2|X(\omega)| \cos(\omega n + \angle X(\omega)) d\omega \end{split}$$

Any signal x[n] is a sum of sinusoids with all frequencies $f \in [0, \frac{1}{2}]$, or $\omega \in [0, \pi]$

Expressing as sum of sinusoids

- ► The DTFT shows that any signal x[n] is a "sum" of sinusoids with all frequencies $f \in [0, \frac{1}{2}]$, or $\omega \in [0, \pi]$
 - ▶ this is the fundamental practical interpretation of the Fourier transform
 - not really a sum, because we have an integral
- ▶ The **modulus** $|X(\omega)|$ gives the **amplitude** of the sinusoids (× 2)
 - for $\omega = 0$, $|X(\omega = 0)| =$ the DC component
- ▶ The **phase** $\angle X(\omega)$ gives the initial phase

1. Linearity

$$a \cdot x_1[n] + b \cdot x_2[n] \leftrightarrow a \cdot X_1(\omega) + b \cdot X_2(\omega)$$

Proof: via definition

2. Shifting in time

$$x[n-n_0] \leftrightarrow e^{-j\omega n_0} X(\omega)$$

Proof: via definition

▶ The amplitudes $|X(\omega)|$ is not affected, shifting in time affects only the phase

3. Modulation in time

$$e^{j\omega_0 n}x[n] \leftrightarrow X(\omega-\omega_0)$$

4. Complex conjugation

$$x^*[n] \leftrightarrow X^*(-\omega)$$

5. Convolution

$$x_1[n] * x_2[n] \leftrightarrow X_1(\omega) \cdot X_2(\omega)$$

▶ Not circular convolution, this is the normal convolution

6. Product in time

Product in time \leftrightarrow convolution of Fourier transforms

$$x_1[n] \cdot x_2[n] \leftrightarrow \frac{1}{2\pi} \int_{-\pi}^{\pi} X_1(\lambda) X_2(\omega - \lambda) d\lambda$$

Correlation theorem

$$r_{X_1X_2}[I] \leftrightarrow X_1(\omega)X_2^{\star}(\omega)$$

Wiener Khinchin theorem

Autocorrelation of a signal \leftrightarrow Power spectral density

$$r_{xx}[I] \leftrightarrow S_{xx}(\omega) = |X(\omega)|^2$$

Parseval theorem

▶ Parseval theorem: energy of the signal is the same in time and frequency domains

$$E = \sum_{-\infty}^{\infty} |x[n]|^2 = \frac{1}{2\pi} \int_{-\pi}^{\pi} |X(\omega)|^2$$

Is true for all orthonormal bases

IV.4 The Discrete Fourier Transform (DFT)

Definitions

Definitions (again):

Inverse Discrete Fourier Transform (DFT)

$$x[n] = \frac{1}{N} \sum_{k=0}^{N-1} X_k e^{j2\pi kn/N}$$

Discrete Fourier Transform (DFT)

$$X_k = \langle x[n], e^{j2\pi fn} \rangle = \sum_{n=0}^{N-1} x[n]e^{-j2\pi kn/N}$$

- DFT is defined for periodical signals with period N
- there are exactly N terms in each sum

Definitions

To remember:

- ▶ DFT: takes a vector with N elements (x[n]), produces a vector with N elements (X_k)
 - ▶ for this reason, we can compute it e.g. with Matlab
- ▶ DTFT: takes a vector with ∞ elements (x[n]), produces a continuous function $(X(\omega))$ between $[-\pi, -pi]$

Periodicity and notation

- ▶ DTFT has only N coefficients X_k , each X_k corresponding to a frequency $f = \frac{k}{N}$
- ▶ In frequency domain, $f = \frac{N-k}{N} = \frac{-k}{N}$ (aliasing, we can always add/subtract 1 from f)
- ▶ So we can consider X_{N-k} as X_{-k} , due to periodicity
- \triangleright Example: a signal with period N=6 has 6 DFT coefficients
 - \blacktriangleright we can call them X_0 , X_1 , X_2 , X_3 , X_4 , X_5
 - we have $X_5 = X_{-1}$, $X_4 = X_{-2}$
 - \blacktriangleright we can also call them X_{-2} , X_{-1} , X_0 , X_1 , X_2 , X_3

Basic Properties of the DFT

- ightharpoonup Has only N coefficients X_k
- ► X_k are **complex** (has $|X_k|$, $\angle X_k$)
- ▶ If the signal x[n] is real, the coefficients are **even**

$$x[n] \in \mathbb{R} \to X_{-k} = X_k^*$$

- ► This means:
 - ightharpoonup modulus is even: $|X_k| = |X_{-k}|$
 - ▶ phase is odd: $\angle X_{-k} = -\angle X_k$

Expressing as sum of sinusoids, N = odd

- ightharpoonup Grouping terms with k and -k:
- ▶ If *N* is odd, we have X_0 and pairs (X_k, X_{-k}) :

$$x[n] = \frac{1}{N} X_0 e^{j0n} + \frac{1}{N} \sum_{k=-(N-1)/2}^{-1} X_k e^{j2\pi kn/N} + \frac{1}{N} \sum_{k=1}^{(N-1)/2} X_k e^{j2\pi kn/N}$$

$$= \frac{1}{N} X_0 + \frac{1}{N} \sum_{k=1}^{(N-1)/2} (X_k e^{j2\pi kn/N} + X_{-k} e^{-j2\pi kn/N})$$

$$= \frac{1}{N} X_0 + \frac{1}{N} \sum_{k=1}^{(N-1)/2} |X_k| (e^{j2\pi kn/N + \angle X(k)} + e^{-j2\pi kn/N - \angle X(\omega)})$$

$$= \frac{1}{N} X_0 + \frac{1}{N} \sum_{k=0}^{(N-1)/2} 2|X_k| \cos(2\pi k/Nn + \angle X_k)$$

 \triangleright A sum of sinusoids with frequencies up to 1/2

Expressing as sum of sinusoids, N = even

▶ If N is even, we have X_0 and pairs (X_k, X_{-k}) , with an extra term $X_{N/2}$ which has no pair

• e.g. N = 6:
$$X_{-2}, X_{-1}, X_0, X_1, X_2, X_3$$

- ▶ The term with no pair, $X_{N/2}$, must be a real number, because $X_{N/2} = X_{-N/2}^* = XN/2^*$
- ▶ The extra term will be $\frac{1}{N}X_{N/2}e^{j2\pi N/2n/N} = X_{N/2}\cos(n\pi)$
- Overall:

$$x[n] = \frac{1}{N}X_0 + \frac{1}{N}\sum_{k=0}^{(N-2)/2} 2|X_k|\cos(2\pi k/Nn + \angle X_k) + \frac{1}{N}X_{N/2}\cos(n\pi)$$

ightharpoonup A sum of sinusoids with frequencies up to 1/2

Expressing as sum of sinusoids

▶ DFT says that any periodic signal x[n], with period N, is a sum of N sinusoids with frequencies:

$$f=0,\frac{1}{N},\frac{2}{N},\dots$$
 up to $\frac{N-1}{2}$ or $\frac{N}{2}$

(not exceeding 1/2)

- ► The **modulus** $|X_k|$ gives the **amplitude** of the sinusoids (sometimes \times 2)
 - for $\omega = 0$, $|X_0| =$ the DC component
 - ightharpoonup when modulus = 0, that frequency has amplitude 0
- ▶ The **phase** $\angle X_k$ gives the initial phase

Example

► Consider a periodic signal x[n] with period N = 6 and the DFT coefficients:

$$\textit{X}_k = [15.0000 + 0.0000i \ , \ -2.5000 + 3.4410i \ , \ -2.5000 + 0.8123i \ , \ -2.5000 \ - \ 0.8123i \ , \ -2.5000 \ - \ 3.4410i]$$

Write x[n] as a sum of sinusoids.

▶ Do the same for a periodic signal x[n] with period N = 5 and the DFT coefficients:

$$X_k = [21.0000 + 0.0000i, -3.0000 + 5.1962i, -3.0000 + 1.7321i, -3.0000 + 0.0000i, -3.0000 - 1.7321i, -3.0000 - 5.1962i]$$

Write x[n] as a sum of sinusoids.

DFT matrix

- ► The DFT (and the inverse IDFT) is equivalent with a matrix multiplication:
 - on whiteboard
- In the world of discrete signals, there are many signal transforms possible, and many of them can be expressed as matrix multiplications, just like the DFT.

1. Linearity

If the signal $x_1[n]$ has the DFT coefficients $\{X_{\nu}^{(1)}\}$, and $x_2[n]$ has $\{X_{\nu}^{(2)}\}$, then their sum has

$$a \cdot x_1[n] + b \cdot x_2[n] \leftrightarrow \{a \cdot X_k^{(1)} + b \cdot X_k^{(2)}\}$$

Proof: via definition

2. Shifting in time

If $x[n] \leftrightarrow \{X_k\}$, then

$$x[n-n_0] \leftrightarrow \{e^{(-j2\pi k n_0/N)}X_k\}$$

Proof: via definition

▶ The amplitudes $|X_k|$ are not affected, shifting in time **affects only** the phase

3. Modulation in time

$$e^{j2\pi k_0 n/N} \leftrightarrow \{X_{k-k_0}\}$$

4. Complex conjugation

$$x^*[n] \leftrightarrow \{X_{-k}^*\}$$

5. Circular convolution

Circular convolution of two signals \leftrightarrow product of coefficients

$$x_1[n] \otimes x_2[n] \leftrightarrow \{N \cdot X_k^{(1)} \cdot X_k^{(2)}\}$$

Circular convolution definition:

$$x_1[n] \otimes x_2[n] = \sum_{k=0}^{N-1} x_1[k] x_2[(n-k)_N]$$

- takes two periodic signals of period N, result is also periodic with period N
- Example at the whiteboard: how it is computed

Example

Example (write on slides)

Circular convolution

- ▶ We are in the vector space of **periodic signals** with period N
- Linear (e.g. normal) convolution produces a result which is longer periodic with period N
- Circular convolution takes two sequences of length N and produces another sequence of length N
 - each sequence is a period of a periodic signal
 - circular convolution = like a convolution of periodic signals

6. Product in time

Product in time \leftrightarrow circular convolution of DFT coefficients

$$x_1[n] \cdot x_2[n] \leftrightarrow \sum_{m=0}^{N-1} X_m^{(1)} X_{(k-m)_N}^{(2)} = X_k^{(1)} \otimes X_k^{(2)}$$

► Parseval theorem: energy of the signal is the same in time and frequency domains

$$E = \sum_{0}^{N-1} |x[n]|^2 = \frac{1}{2\pi} \sum |X_k|^2$$

► Is true for all orthonormal bases

Plot / sketch DTFT and DFT of various signals

Let's plot / sketch DTFT and DFT of various signals

DTFT of:

- ightharpoonup a constant signal x[n] = A
- ▶ a rectangular signal x[n] = A between $-\tau$ and τ , 0 elsewhere
- ightharpoonup a cosine of frequency precisely f = k/N
- ightharpoonup a cosine of frequency not f = k/N

DFT, with N=20, of:

- ightharpoonup a constant signal x[n] = A
- ▶ a rectangular signal x[n] = A between $-\tau$ and τ , 0 elsewhere
- ightharpoonup a cosine of frequency precisely f = k/N
- ▶ a cosine of frequency not f = k/N

DFT matrix

► The DFT (and the inverse IDFT) is equivalent with a matrix multiplication:

DFT:

$$\mathbf{X} = \mathbf{W}_N \mathbf{x}$$

IDFT:

$$\mathbf{x} = \mathbf{W}_N^T \mathbf{X}$$

where

$$\mathbf{X} = \begin{bmatrix} X_0 \\ X_1 \\ \vdots \\ X_{N-1} \end{bmatrix}, \quad \mathbf{x} = \begin{bmatrix} x_0 \\ x_1 \\ \vdots \\ x_{N-1} \end{bmatrix},$$

DFT matrix (continued)

$$\mathbf{W}_{N} = \frac{1}{\sqrt{N}} \begin{bmatrix} W_{N}^{0\cdot0} & W_{N}^{0\cdot1} & \cdots & W_{N}^{0\cdot(N-1)} \\ W_{N}^{1\cdot0} & W_{N}^{1\cdot1} & \cdots & W_{N}^{1\cdot(N-1)} \\ \vdots & \vdots & \ddots & \vdots \\ W_{N}^{(N-1)\cdot0} & W_{N}^{(N-1)\cdot1} & \cdots & W_{N}^{(N-1)\cdot(N-1)} \end{bmatrix}$$

with an element $W_N^{kn}=e^{-j2\pi\frac{k}{N}n}$ (k= row index, n= column index)

- ▶ there might be small variations depending on whether we have $\frac{1}{\sqrt{N}}$ at both DFT and IDFT, or just put $\frac{1}{N}$ just for the IDFT
- ▶ note that for IDFT we have $W^{-1} = W^T$ (orthonormal basis)

DFT matrix multiplication

- Naive implementation of DFT, IDFT: use matrix multiplication with $W,\ W^{-1}$
- ▶ Number of multiplications necessary for a vector of length N is N^2
- In the world of algorithms, the computational complexity of an algorithm = number of multiplications necessary, depending on some variable N
 - only the dominant term matters, no coefficient, e.g $O(N^2)$ not $7.3N^2+4N$
- ▶ Naive DFT has computation complexity $\mathcal{O}(N^2)$
 - prohibitively large

FFT

- ► The Fast Fourier Transform (FFT) algorithm = a fast algorithm for computing the DFT, exploiting the particular nature (symmetries) in the DFT matrix
- ▶ FFT computational complecity: $\mathcal{O}(N \log_2(N))$
- ightharpoonup Exercise: for N=1024, how much faster is FFT compared to naive DFT multiplication?
- ▶ Invention and adoption of FFT (~'60s, Cooley & Tukey) = "the birth of Digital Signal Processing"

Other transforms

- ► In the world of discrete signals, there are many signal transforms possible, and many of them can be expressed as matrix multiplications, just like the DFT.
- ▶ Transform = expressing a N-dimensional vector x as a linear combination of a set of N basis vectors
- ► How:
 - 1. Put the N vectors of the basis as columns in a matrix A
 - 2. Solve the system x = AX (inverse transform)
 - 3. Which means $X = A^{-1}x$ (forward transform)
- ► Why:
 - compression: the discrete cosine transform is the basis for lossy JPEG image compression
 - **.**..

Example

Consider the exercise from Week 2:

$$x[n] = \{..., 0, \frac{1}{3}, \frac{2}{3}, 1, 1, 1, 1, 0, ...\}$$

Write the expression of x[n] based on the signal u[n].

Solve this in Matlab using a matrix approach

Another example: JPEG

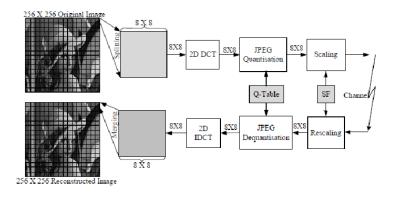


Figure 4: JPEG compression with DCT transform

image from: JPEG Picture Compression Using Discrete Cosine Transform, N. K. More, S. Dubey, 2012

Another example: JPEG (cont'd)

- ► Each 8x8 image block is a vector in a 64 dimensional space
- ► Each 8x8 image block is decomposed into 64 basis vectors

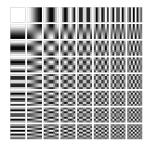


Figure 5: 8x8 DCT basis vectors

- ▶ Result: 64 coefficients, but many are small, negligible, quantizable => compression
- image from: Wikipedia

- ► How are DTFT and DFT related?
- Discrete Time Fourier Transform (DTFT):
 - ► for non-periodical signals
 - spectrum is continuous
- Discrete Fourier Transform (DFT):
 - for periodical signals
 - spectrum is discrete
- ▶ Duality: periodic in time ↔ discrete in frequency
- ► The Diracs of the DFT are samples from the continuous DTFT of a single period of the signal

Example: consider a sequence of 7 values

$$x = [6, 3, -4, 2, 0, 1, 2]$$

▶ If we consider a x surrounded by infinitely long zeros (x[n] non-periodical), we have a continuous spectrum $X(\omega)$ (DTFT)

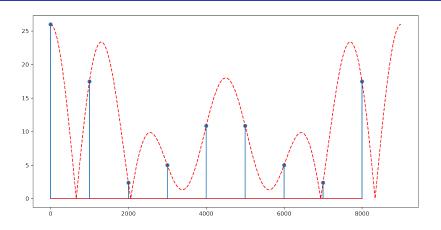
$$x = [...0, 6, 3, -4, 2, 0, 1, 2, 0, ...] \leftrightarrow X(\omega)$$

▶ If we consider that x is surrounded by repeating the sequences (x[n] periodical), we have a discrete spectrum X_k (DFT)

$$x = [..., -4, 2, 0, 1, 2, 0, 6, 3, -4, 2, 0, 1, 2, 0, 6, 3, -4, ...] \leftrightarrow X_k$$

▶ The discrete X_k are just **samples from** $X(\omega)$, at frequencies k/N:

$$X_k = X(2\pi(k/N)n)$$



$$x = [6, 5, 4, -3, 2, -3, 4, 5, 6]$$

- red line = DTFT of x (assuming surrounded by zeros)
 (actually run as fft(x, 10000), x is extended with 9991 zeros)
- ▶ blue = DFT of x (assumes periodic) = fft(x)

- ightharpoonup Consider a non-periodic signal x[n]
- lt has a continuous spectrum $X(\omega)$
- ▶ If we **periodize** it by repeating with period N:

$$x_N[n] = \sum_{k=-\infty}^{\infty} x[n-kN]$$

▶ then the Fourier transform is **discrete** (made of Diracs):

$$X_N(\omega) = 2\pi X_k \delta(\omega - k \frac{2\pi}{N})$$

▶ The coefficients of the Diracs = the DFT coefficients

$$X_k = X(2\pi k/Nn)$$

▶ They are **samples** from the continuous $X(\omega)$ of the non-periodized signal

Study case

```
n = 0:99;
f = 0.015;
x = cos(2*pi*f*n)
plot(abs(fft(x)))
```

Discuss:

- 1. Why is the spectrum not just 2 Diracs, like a normal cos()?
- 2. FFT assumes periodicity. Are there boundary problems?
- 3. What is the role of the rectangular window?
- 4. What happens if we run fft(x, 10000) instead of fft(x)?

- When you have finite-length cosine vector x, you have just a part of your signal
- ▶ The true signal x, infinitely long, is actually multiplied with a rectangular window w[n]

$$x = \cos(2\pi f n) \cdot w[n]$$

- Multiplication in time = convolution in frequency

 The spectrum of $x = x[n] \cdot w[n]$ is Diracs * $W(\omega)$
- ▶ Instead of Diracs, you have $W(\omega)$'s:
 - wide peak
 - secondary lobes

- ▶ Working with a piece of a signal always distorts a signal
 - every Dirac is "smudged" into a $W(\omega)$
- ► This is unavoidable
- If we have a segment of a cosine, in the DTFT (continuous) we never see Diracs, but $W(\omega)$
 - ▶ The longer the piece, the better

- ▶ So what's the problem if we see $W(\omega)$ instead of Diracs?
- ► Frequency resolution = the ability to distinguish between closely spaced frequency components in a signal
- ▶ Having $W(\omega)$ instead of a Dirac is bad because it **masks** the surrounding region
 - a two close Diracs with similar height are be impossible to differentiate, because of the wide central lobe
 - a Dirac further away but smaller is impossible to differentiate, because of secondary lobes
- Analyzing a short segment of a signal leads to low resolution in frequency
- ► Analyzing a longer segment leads to higher resolution in frequency
- ▶ Frequency resolution is proportional to the length of the signal

- We can change the window w[n]
 - ► Recangular window
 - Hamming window
 - ► Hann window
 - **.**..
- What they do: trade narrow peak vs small secondary lobes
 - ► Rectangular window: widest peak, smallest secondary lobes
 - Other windows : narrower peak, higher secondary lobes
- ▶ What they do: attenuate endings, to reduce boundary problems

- ▶ Remember: every time we work with a piece of a signal (e.g. we process an audio file in pieces of 1024 samples), we are applying windowing
- ▶ Even the rectangular window is still a window
- ▶ If you need to compute the spectrum, know that is is affected
- ► Always consider replacing the rectangular window with another one, if you use fft() or other frequency-based operations

STFT and Spectrogram

- ► How to analyze the frequency of a signal whose frequency components change in time (e.g. like a musical song)?
- ▶ Short-Time Fourier Transform (STFT) = a technique for analyzing the frequency content of local sections of a signal as it changes over time.
- ➤ STFT divides a longer time signal into shorter segments of equal length and then computes the Fourier Transform separately on each short segment.
 - ► Split the signal into pieces (e.g. 1024-samples long)
 - Compute the spectrum of every piece (e.g. fft())
 - Display the resulting sequence of spectra = "spectrogram"

STFT and Spectrogram

- ► The STFT is a **time-frequency representation** of a signal
- ▶ 2-Dimesional: time and frequency
- Examples: https://en.wikipedia.org/wiki/Spectrogram

STFT: Time and frequency resolution

- Imagine you look at the spectrogram of a music piece, and you want to pinpoint the moment where the bass guitar starts to play a chord of 100Hz
- ▶ Do a STFT and look for the moment where you see a high spectrum atoun 100Hz
- ▶ If the segments are short:
 - good time resolution
 - poor frequency resolution
- If the segments are long:
 - poor time resolution
 - high frequency resolution

STFT: Time and frequency resolution

Time-frequency Uncertainty Principle:

you cannot have very good time resolution and very good frequency resolution simultaneously

STFT: other issues

- Other issues with STFT
 - can change the window type, to alleviate boundary problems / artefacts
 - ▶ allow some overlap between segments (e.g. 10%)

How to compute the DTFT

- ► The DFT is computed with fft(x)
- ► How to compute the DTFT?
- ▶ You can't. You need to surround x with infinitely long zeros
- ▶ You can only surround it with many zeros, but still finite
- ▶ Do this with fft(x, 100000) (DFT in N=100000 points)
 - \triangleright x is surrounded with zeros unil total length = 100000
 - it's still just fft(), so DFT, so you have points and not the full continuous function
 - but it's many many points

How to compute the DTFT

- ➤ Computing the fft() in N=100000 points is unrelated with frequency resolution
- Frequency resolution is dependent on actual length of x
- lacktriangle Windowing changes the Diracs into $W(\omega)$'s
- fft() in N points is just taking N points from the resulting continuous spectrum

Relation between DTFT and Z transform

Z transform:

$$X(z) = \sum_{n} x[n]z^{-n}$$

► DTFT:

$$X(\omega) = \sum_{n} x[n]e^{-j\omega n}$$

▶ DTFT can be obtained from Z transform with

$$z=e^{j\omega}$$

- ► These $z = e^{j\omega}$ are **points on the unit circle**
 - $|z| = |e^{j\omega}| = 1 (modulus)$
 - $ightharpoonup \angle z = \angle e^{j\omega} = \omega(phase)$

Relation between DTFT and Z transform

- ► Fourier transform = Z transform evaluated **on the unit circle**
 - if the unit circle is in the convergence region of Z transform
 - otherwise, equivalence does not hold
- ▶ This is true for most usual signals we work with
 - some details and discussions are skipped

$$X(z) = C \cdot \frac{(z - z_1) \cdots (z - z_M)}{(z - p_1) \cdots (z - p_N)}$$
$$X(\omega) = C \cdot \frac{(e^{j\omega} - z_1) \cdots (e^{j\omega} - z_M)}{(e^{j\omega} - p_1) \cdots (e^{j\omega} - p_N)}$$

Modulus:

$$|X(\omega)| = |C| \cdot \frac{|e^{j\omega} - z_1| \cdots |e^{j\omega} - z_M|}{|e^{j\omega} - p_1| \cdots |e^{j\omega} - p_N|}$$

Phase:

$$\angle X = \angle C + \angle (e^{j\omega} - z_1) + \cdots + \angle (e^{j\omega} - z_M) - \angle (e^{j\omega} - p_1) - \cdots - \angle (e^{j\omega} - p_N)$$

- ► For complex numbers:
 - ightharpoonup modulus of |a-b| = the length of the segment between a and b
 - ▶ phase of |a b| = the angle of the segment from b to a (direction is important)
- ▶ So, for a point on the unit circle $z = e^{j\omega}$
 - ▶ modulus $|X(\omega)|$ is given by the distances to the zeros and to the poles
 - ▶ phase $\angle X(\omega)$ is given by the angles from the zeros and poles to z

- Consequences:
 - when a pole is very close to unit circle -> Fourier transform is large at this point
 - when a zero is very close to unit circle -> Fourier transform is small at this point
- Examples: . . .

- ▶ Simple interpretation for modulus $|X(\omega)|$:
 - ightharpoonup Z transform X(z) is like a landscape
 - poles = mountains of infinite height
 - zeros = valleys of zero height
 - ▶ Fourier transform $X(\omega) =$ "Walking over this landscape along the unit circle"
 - The height profile of the walk gives the amplitude of the Fourier transform
 - ▶ When close to a mountain -> road is high -> Fourier transform has large amplitude
 - When close to a valley -> road is low -> Fourier transform has small amplitude

- Note: X(z) might also have a constant C in front!
 - ▶ It does not appear in pole-zero plot
 - ▶ The value of |C| and $\angle C$ must be determined separately
- ► This "geometric method" can be applied for phase as well

Time-frequency duality

- ▶ **Duality** properties related to all Fourier transforms
- ▶ Discrete ↔ Periodic
 - discrete in time -> periodic in frequency
 - periodic in time -> discrete in frequency
- ► Continuous ↔ Non-periodic
 - continous in time -> non-periodic in frequency
 - non-periodic in time -> continuous in frequency

Terminology

- Based on frequency content:
 - low-frequency signals
 - mid-frequency signals (band-pass)
 - high-frequency signals
- **Band-limited** signals: spectrum is 0 beyond some frequency f_{max}
- ▶ **Bandwitdh** B: frequency interval $[F_1, F_2]$ which contains 95% of energy
 - ► $B = F_2 F_1$
- ▶ Based on bandwidth *B*:
 - ▶ Narrow-band signals: B << central frequency $\frac{F_1+F_2}{2}$
 - Wide-band signals: not narrow-band