Representing Images and Sounds

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Representing an Elephant

It was six men of Indostan,
To learning much inclined,
Who went to see the elephant,
(Though all of them were blind),
That each by observation
Might satisfy his mind.

The first approached the elephant,
And happening to fall
Against his broad and sturdy side,
At once began to bawl:
"God bless me! But the elephant
Is very like a wall!"

The second, feeling of the tusk,
Cried: "Ho! What have we here,
So very round and smooth and sharp?
To me 'tis very clear,
This wonder of an elephant
Is very like a spear!"

The third approached the animal,
And happening to take
The squirming trunk within his hands,
Thus boldly up and spake:
"I see," quoth he, "the elephant
Is very like a snake!"

The fourth reached out an eager hand,
And felt about the knee.
"What most this wondrous beast is like
Is might plain," quoth he;
"Tis clear enough the elephant
Is very like a tree."

The fifth, who chanced to touch the ear,
Said: "E'en the blindest man
Can tell what this resembles most:
Deny the fact who can,
This marvel of an elephant
Is very like a fan."

The sixth no sooner had begun
About the beast to grope,
Than seizing on the swinging tail
That fell within his scope,
"I see," quoth he, "the elephant
Is very like a rope."

And so these men of Indostan
Disputed loud and long,
Each in his own opinion
Exceeding stiff and strong.
Though each was partly right,
All were in the wrong.
Representation

Describe these images

Such that a listener can visualize what you are describing

More images
How do you describe them?

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Sounds are just sequences of numbers

When plotted, they just look like blobs

Which leads to the natural “sounds are blobs”

Or more precisely, “sounds are sequences of numbers that, when plotted, look like blobs”

Which wont get us anywhere
Representation

- Representation is description
- But in compact form
- Must describe the salient characteristics of the data
  - E.g. a pixel-wise description of the two images here will be completely different

- Must allow identification, comparison, storage..
The most common element in the image: background

Or rather large regions of relatively featureless shading

Uniform sequences of numbers

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Representing images using a “plain” image

Most of the figure is a more-or-less uniform shade
  Dumb approximation – a image is a block of uniform shade
    Will be mostly right!
  How much of the figure is uniform?

How? Projection
  Represent the images as vectors and compute the projection of the image on the “basis”

\[
BW \rightarrow \text{Image}
\]
\[
W = \text{pinv}(B)\text{Image}
\]
\[
\text{PROJECTION} = BW = B(B^T B)^{-1} B^T . \text{Image}
\]
Adding more bases

- Lets improve the approximation
- Images have some fast varying regions
  - Dramatic changes
  - Add a second picture that has very fast changes
    - A checkerboard where every other pixel is black and the rest are white

\[
\begin{align*}
\text{Image} & \rightarrow w_1 B_1 + w_2 B_2 \\
W & = \begin{bmatrix} w_1 \\ w_2 \end{bmatrix} \\
B & = \begin{bmatrix} B_1 & B_2 \end{bmatrix}
\end{align*}
\]

\[
BW \rightarrow \text{Image}
\]

\[
W = \text{pinv}(B) \text{Image}
\]

\[
\text{PROJECTION} = BW = B(B^T B)^{-1} B^T \cdot \text{Image}
\]
Adding still more bases

Regions that change with different speeds

\[
\text{Image} \rightarrow w_1 B_1 + w_2 B_2 + w_3 B_3 + \ldots
\]

\[
W = \begin{bmatrix}
w_1 \\
w_2 \\
w_3 \\
\vdots
\end{bmatrix}
\]

\[
B = [B_1 \ B_2 \ B_3]
\]

\[
BW \rightarrow \text{Image}
\]

\[
W = \text{pinv}(B) \text{Image}
\]

\[
\text{PROJECTION} = BW = B(B^TB)^{-1}B^T \text{Image}
\]

Getting closer at 625 bases!
Representation using checkerboards

A “standard” representation

Checker boards are the same regardless of what picture you’re trying to describe

As opposed to using “nose shape” to describe faces and “leaf colour” to describe trees.

Any image can be specified as (for example)

\[0.8 \times \text{checkerboard}(0) + 0.2 \times \text{checkerboard}(1) + 0.3 \times \text{checkerboard}(2)\]

The definition is sufficient to reconstruct the image to some degree

Not perfectly though
What about sounds?

Square wave equivalents of checker boards
Projecting sounds

\[ \begin{bmatrix} B_1 & B_2 & B_3 \end{bmatrix} \begin{bmatrix} w_1 \\ w_2 \\ w_3 \end{bmatrix} = \begin{bmatrix} \text{Signal} \rightarrow w_1 B_1 + w_2 B_2 + w_3 B_3 \\ \text{W} = \begin{bmatrix} w_1 \\ w_2 \\ w_3 \end{bmatrix} \\ \text{B} = [B_1 \ B_2 \ B_3] \end{bmatrix} \]

\[ BW \rightarrow \text{Signal} \]

\[ W = \text{pinv}(B)\text{Signal} \]

\[ \text{PROJECTION} = BW = B(B^T B)^{-1} B \text{Signal} \]

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Why checkerboards are great bases

- We cannot explain one checkerboard in terms of another
  - The two are orthogonal to one another!

- This means that we can find out the contributions of individual bases separately
  - Joint decomposition with multiple bases with give us the same result as separate decomposition with each of them
  - This never holds true if one basis can explain another

\[
\begin{align*}
W &= P_{\text{inv}}(B) \text{Image} \\
\text{Image} &= w_1 B_1 + w_2 B_2 \\
P_{\text{inv}}(B) \text{Image} &= \begin{bmatrix} P_{\text{inv}}(B_1) \text{Image} \\ P_{\text{inv}}(B_2) \text{Image} \end{bmatrix} = \begin{bmatrix} w_1 \\ w_2 \end{bmatrix}
\end{align*}
\]
Checker boards are not good bases

- Sharp edges
- Can *never* be used to explain rounded curves
Sinusoids ARE good bases

- They are orthogonal
- They can represent rounded shapes nicely
- Unfortunately, they cannot represent sharp corners
What are the frequencies of the sinusoids

Follow the same format as the checkerboard:
- DC
- The entire length of the signal is one period
- The entire length of the signal is two periods.
- And so on..

The k-th sinusoid:
- \( F(n) = \sin(2\pi kn/N) \)
  - \( N \) is the length of the signal
  - \( k \) is the number of periods in \( N \) samples
How many frequencies in all?

- A max of $L/2$ periods are possible
- If we try to go to $(L/2 + X)$ periods, it ends up being identical to having $(L/2 – X)$ periods
  - With sign inversion

Example for $L = 20$
- Red curve = sine with 9 cycles (in a 20 point sequence)
  - $Y(n) = \sin(2\pi 9n/20)$
- Green curve = sine with 11 cycles in 20 points
  - $Y(n) = -\sin(2\pi 11n/20)$
- The blue lines show the actual samples obtained
  - These are the only numbers stored on the computer
  - This set is the same for both sinusoids

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How to compose the signal from sinusoids

The sines form the vectors of the projection matrix

Pinv() will do the trick as usual

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How to compose the signal from sinusoids

$$\begin{bmatrix}
\sin(2\pi 0.0/L) & \sin(2\pi 1.0/L) & \cdot & \sin(2\pi (L/2).0/L) \\
\sin(2\pi 0.1/L) & \sin(2\pi 1.1/L) & \cdot & \sin(2\pi (L/2).1/L) \\
\cdot & \cdot & \cdot & \cdot \\
\sin(2\pi 0.(L-1)/L) & \sin(2\pi 1.(L-1)/L) & \cdot & \sin(2\pi (L/2).(L-1)/L)
\end{bmatrix}
\begin{bmatrix}
w_1 \\
w_2 \\
w_{L/2}
\end{bmatrix}
= 
\begin{bmatrix}
s[0] \\
s[1] \\
s[L-1]
\end{bmatrix}
$$

L/2 columns only

$$W = \begin{bmatrix}w_1 \\w_2 \\w_3\end{bmatrix} \quad B = [B_1 \ B_2 \ B_3]$$

$$Signal = \begin{bmatrix}s[0] \\s[1] \\
\cdot \\
s[L-1]\end{bmatrix}$$

$$BW \to Signal$$

$$W = \text{pinv}(B)Signal$$

$$PROJECTION = BW = B(B^T B)^{-1}B.Signal$$

The sines form the vectors of the projection matrix

Pinv() will do the trick as usual
Each sinusoid’s amplitude is adjusted until it gives us the least squared error.

The amplitude is the weight of the sinusoid.

This can be done independently for each sinusoid.
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- The amplitude is the weight of the sinusoid.
- This can be done independently for each sinusoid.
Every sine starts at zero
  Can never represent a signal that is non-zero in the first sample!

Every cosine starts at 1
  If the first sample is zero, the signal cannot be represented!
The need for phase

- Allow the sinusoids to move!

\[ \text{signal} = w_1 \sin(2\pi kn/N + f_1) + w_2 \sin(2\pi kn/N + f_2) + w_3 \sin(2\pi kn/N + f_3) + \ldots \]

- How much do the sines shift?

Sines are shifted: do not start with value = 0
Determining phase

Least squares fitting: move the sinusoid left / right, and at each shift, try all amplitudes
- Find the combination of amplitude and phase that results in the lowest squared error

We can still do this separately for each sinusoid
- The sinusoids are still orthogonal to one another
Determining phase

- Least squares fitting: move the sinusoid left / right, and at each shift, try all amplitudes
  - Find the combination of amplitude and phase that results in the lowest squared error
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Determining phase

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  - Find the combination of amplitude and phase that results in the lowest squared error
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The problem with phase

This can no longer be expressed as a simple linear algebraic equation

- The phase is integral to the bases
  - I.e. there’s a component of the basis itself that must be estimated!

- Linear algebraic notation can only be used if the bases are fully known
  - \textit{We can only (pseudo) invert a known matrix}
Complex Exponential to the rescue

\[ b[n] = \sin(freq \cdot n) \]

\[ b[n] = \exp(j \cdot freq \cdot n) = \cos(freq \cdot n) + j \sin(freq \cdot n) \]

\[ j = \sqrt{-1} \]

\[ \exp(j \cdot freq \cdot n + \phi) = \exp(j \cdot freq \cdot n)\exp(\phi) = \cos(freq \cdot n + \phi) + j \sin(freq \cdot n + \phi) \]

- The cosine is the real part of a complex exponential
- The sine is the imaginary part
- A phase term for the sinusoid becomes a multiplicative term for the complex exponential!!
Explaining with Complex Exponentials

\[ A \times + \]

\[ B \times + \]

\[ C \times = \]
Complex exponentials are well behaved

- Like sinusoids, a complex exponential of one frequency can never explain one of another
  - They are orthogonal
- They represent smooth transitions
- Bonus: They are complex
  - Can even model complex data!
- They can also model real data
  - $\exp(jx) + \exp(-jx)$ is real
    - $\cos(x) + j\sin(x) + \cos(x) - j\sin(x) = 2\cos(x)$
- More importantly
  - $\exp\left(j2\pi \frac{(L/2 - x)n}{L}\right) + \exp\left(j2\pi \frac{(L/2 + x)n}{L}\right)$ is real
  - The complex exponentials with frequencies equally spaced from L/2 are complex conjugates
Complex exponentials are well behaved

\[
\exp\left(j2\pi \frac{(L/2 - x)n}{L}\right) + \exp\left(j2\pi \frac{(L/2 + x)n}{L}\right) \quad \text{is real}
\]

The complex exponentials with frequencies equally spaced from \( L/2 \) are complex conjugates

“Frequency = \( k \)” \( \implies \) \( k \) periods in \( L \) samples

\[
a \exp\left(j2\pi \frac{(L/2 - x)n}{L}\right) + \text{conjugate}(a) \exp\left(j2\pi \frac{(L/2 + x)n}{L}\right)
\]

Is also real

If the two exponentials are multiplied by numbers that are conjugates of one another the result is real
Describe the data using L complex exponential bases.

The weights given to the \((L/2 + k)\)th basis and the \((L/2 - k)\)th basis should be complex conjugates, to make the result real.

Because we are dealing with real data.

Fortunately, a least squares fit will give us identical weights to both bases automatically; there is no need to impose the constraint externally.
Complex Exponential Bases: Algebraic Formulation

\[
\begin{bmatrix}
\exp(j2\varphi.0.0/L) & \exp(j2\varphi.(L/2).0/L) & \exp(j2\varphi.(L-1).0/L) \\
\exp(j2\varphi.0.1/L) & \exp(j2\varphi.(L/2).1/L) & \exp(j2\varphi.(L-1).1/L) \\
\vdots & \vdots & \vdots \\
\exp(j2\varphi.0.(L-1)/L) & \exp(j2\varphi.(L/2).(L-1)/L) & \exp(j2\varphi.(L-1).(L-1)/L)
\end{bmatrix}
\begin{bmatrix}
S_0 \\
S_{L/2} \\
S_{L-1}
\end{bmatrix} =
\begin{bmatrix}
s[0] \\
s[1] \\
s[L-1]
\end{bmatrix}
\]

Note that \( S_{L/2+x} = \text{conjugate}(S_{L/2-x}) \)
Shorthand Notation

\[ W_{L}^{k,n} = \frac{1}{\sqrt{L}} \exp(j2\rho kn/L) = \frac{1}{\sqrt{L}} \left( \cos(2\rho kn/L) + j\sin(2\rho kn/L) \right) \]

\[
\begin{bmatrix}
W_{L}^{0,0} & W_{L}^{L/2,0} & W_{L}^{L-1,0} \\
W_{L}^{0,1} & W_{L}^{L/2,1} & W_{L}^{L-1,1} \\
& \ddots & \ddots \\
W_{L}^{0,L-1} & W_{L}^{L/2,L-1} & W_{L}^{L-1,L-1}
\end{bmatrix}
\begin{bmatrix}
S_{0} \\
S_{L/2} \\
\vdots \\
S_{L-1}
\end{bmatrix}
= 
\begin{bmatrix}
s[0] \\
s[1] \\
\vdots \\
s[L-1]
\end{bmatrix}
\]

Note that \( S_{L/2+x} = \text{conjugate}(S_{L/2-x}) \)
A quick detour

- Real Orthonormal matrix:
  - $XX^T = X X^T = I$
  - But only if all entries are real
  - The inverse of $X$ is its own transpose

- Definition: Hermitian
  - $X^H = \text{Complex conjugate of } X^T$
  - Conjugate of a number $a + ib = a - ib$
  - Conjugate of $\exp(ix) = \exp(-ix)$

- Complex Orthonormal matrix
  - $XX^H = X^H X = I$
  - The inverse of a complex orthonormal matrix is its own Hermitian
Doing it in matrix form

\[ W_L^{k,n} = \frac{1}{\sqrt{L}} \exp(j2\pi kn/L) = \frac{1}{\sqrt{L}} \left( \cos(2\pi kn/L) + j \sin(2\pi kn/L) \right) \]

\[ W_L^{-k,n} = \text{conjugate}(W_L^{k,n}) = \frac{1}{\sqrt{L}} \exp(-j2\pi kn/L) = \frac{1}{\sqrt{L}} \left( \cos(2\pi kn/L) - j \sin(2\pi kn/L) \right) = \]

\[
\begin{bmatrix}
S_0 \\
. \\
S_{L/2} \\
. \\
S_{L-1}
\end{bmatrix} = \begin{bmatrix}
W_L^{0,0} & W_L^{-0,L/2} & . & W_L^{-0,L-1} \\
W_L^{-1,0} & W_L^{-1,L/2} & . & W_L^{-1,L-1} \\
. & . & . & . \\
W_L^{-(L-1),0} & W_L^{-(L-1),L/2} & . & W_L^{-(L-1),(L-1)}
\end{bmatrix} \begin{bmatrix}
s[0] \\
s[1] \\
. \\
s[L-1]
\end{bmatrix}
\]

The complex exponential basis matrix to the left is an orthonormal matrix.

- Its inverse is its own Hermition
- \( W^{-1} = W^H \)
The Discrete Fourier Transform

\[
\begin{bmatrix}
S_0 \\
. \\
S_{L/2} \\
. \\
S_{L-1}
\end{bmatrix}
= \\
\begin{bmatrix}
W_L^{0,0} & W_L^{-0,L/2} & \ldots & W_L^{-0,L-1} \\
W_L^{-1,0} & W_L^{-1,L/2} & \ldots & W_L^{-1,L-1} \\
. & . & . & . \\
W_L^{-(L-1),0} & W_L^{-(L-1),L/2} & \ldots & W_L^{-(L-1),(L-1)}
\end{bmatrix}
\begin{bmatrix}
s[0] \\
s[1] \\
\cdot \\
s[L-1]
\end{bmatrix}
\]

- The matrix to the right is called the “Fourier Matrix”
- The weights \((S_0, S_1, \ldots, \text{Etc.})\) are called the Fourier transform
The Inverse Discrete Fourier Transform

The matrix to the left is the inverse Fourier matrix

Multiplying the Fourier transform by this matrix gives us the signal right back from its Fourier transform
The Fourier Matrix

Left panel: The real part of the Fourier matrix for a 32-point signal

Right panel: The imaginary part of the Fourier matrix
The outcome of the transformation with the Fourier matrix is the **DISCRETE FOURIER TRANSFORM** (DFT)

The **FAST Fourier transform** is an algorithm that takes advantage of the symmetry of the matrix to perform the matrix multiplication really fast

The FFT computes the DFT

Is much faster if the length of the signal can be expressed as $2^N$
The complex exponential is two dimensional

- Has a separate X frequency and Y frequency
  - Would be true even for checker boards!

- The 2-D complex exponential must be unravelled to form one component of the Fourier matrix

- For a KxL image, we’d have K*L bases in the matrix
DFT: Properties

The DFT coefficients are complex
- Have both a magnitude and a phase
- EQUN

Simple linear algebra tells us that
- \( DFT(A + B) = DFT(A) + DFT(B) \)
- The DFT of the sum of two signals is the DFT of their sum

A horribly common approximation in sound processing
- \( \text{Magnitude}(DFT(A+B)) = \text{Magnitude}(DFT(A)) + \text{Magnitude}(DFT(B)) \)
- Utterly wrong
- Absurdly useful
The Fourier Transform and Perception: Sound

- The Fourier transform represents the signal analogously to a bank of tuning forks.
- Our ear has a bank of tuning forks.
- The output of the Fourier transform is perceptually very meaningful.
Symmetric signals

If a signal is symmetric around $L/2$, the Fourier coefficients are real!

$$A(L/2-k) * \exp(-j \cdot f^*(L/2-k)) + A(L/2+k) * \exp(-j \cdot f^*(L/2+k))$$

is always real if

$$A(L/2-k) = A(L/2+k)$$

We can pair up samples around the center all the way; the final summation term is always real.

Overall symmetry properties

- If the *signal* is real, the FT is symmetric.
- If the signal is symmetric, the FT is real.
- If the signal is real and symmetric, the FT is real and symmetric.

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The Discrete Cosine Transform

- Compose a symmetric signal or image
  - Images would be symmetric in two dimensions

- Compute the Fourier transform
  - Since the FT is symmetric, sufficient to store only half the coefficients (quarter for an image)
  - Or as many coefficients as were originally in the signal / image
DCT

\[
\begin{bmatrix}
\cos(2p(0.5).0/2L) & \cos(2p.(1 + 0.5).0/2L) & \cdots & \cos(2p.(L - 0.5).0/2L) \\
\cos(2p.(0.5).1/2L) & \cos(2p.(1 + 0.5).1/2L) & \cdots & \cos(2p.(L - 0.5).1/2L) \\
\vdots & \vdots & \ddots & \vdots \\
\cos(2p.(0.5).(L - 1)/2L) & \cos(2p.(1 + 0.5).(L - 1)/2L) & \cdots & \cos(2p.(L - 0.5).(L - 1)/2L)
\end{bmatrix}
\begin{bmatrix}
w_0 \\
w_1 \\
\vdots \\
w_{L-1}
\end{bmatrix}
= 
\begin{bmatrix}
s[0] \\
s[1] \\
\vdots \\
s[L - 1]
\end{bmatrix}
\]

- Not necessary to compute a 2xL sized FFT
- Enough to compute an L-sized cosine transform
- Taking advantage of the symmetry of the problem
- This is the Discrete Cosine Transform
Representing images

Most common coding is the DCT

JPEG: Each 8x8 element of the picture is converted using a DCT

The DCT coefficients are quantized and stored

- Degree of quantization = degree of compression

Also used to represent textures etc for pattern recognition and other forms of analysis
What does the DFT represent

\[
\begin{bmatrix}
\exp(j2\pi 0.0/L) & \exp(j2\pi (L/2).0/L) & \exp(j2\pi (L - 1).0/L) \\
\exp(j2\pi 0.1/L) & \exp(j2\pi (L/2).1/L) & \exp(j2\pi (L - 1).1/L) \\
. & . & . \\
. & . & . \\
\exp(j2\pi 0.(L - 1)/L) & \exp(j2\pi (L/2).(L - 1)/L) & \exp(j2\pi (L - 1).(L - 1)/L)
\end{bmatrix}
\begin{bmatrix}
S_0 \\
S_{L/2} \\
. \\
. \\
S_{L-1}
\end{bmatrix} =
\begin{bmatrix}
s[0] \\
s[1] \\
. \\
. \\
s[L - 1]
\end{bmatrix}
\]

\[s[n] = \sum_{k=0}^{L-1} S_k \exp(j2\pi kn/L)\]

- The DFT can be written formulaically as above
- There is no restriction on computing the formula for \( n < 0 \) or \( n > L-1 \)
  - It's just a formula
  - But computing these terms behind 0 or beyond \( L-1 \) tells us what the signal composed by the DFT looks like outside our narrow window
What does the DFT represent

If you extend the DFT-based representation beyond 0 (on the left) or L (on the right) it repeats the signal!

So what does the DFT really mean

$s[n] = \sum_{k=0}^{L-1} S_k \exp(j2\pi kn/L)$
What does the DFT represent

The DFT represents the properties of the infinitely long repeating signal that you can generate with it.

Of which the observed signal is ONE period.

This gives rise to some odd effects.
The discrete Fourier transform of the above signal actually computes the properties of the periodic signal shown below which extends from –infinity to +infinity. The period of this signal is 32 samples in this example.
The DFT of one period of the sinusoid shown in the figure computes the spectrum of the entire sinusoid from –infinity to +infinity.
The DFT of one period of the sinusoid shown in the figure computes the spectrum of the entire sinusoid from –infinity to +infinity.
The DFT of one period of the sinusoid shown in the figure computes the spectrum of the entire sinusoid from \(-\infty\) to \(+\infty\).

The DFT of a real sinusoid has only one non-zero frequency.

The second peak in the figure is the “reflection” around \(L/2\) (for real signals).
The DFT of any sequence computes the spectrum for an infinite repetition of that sequence.
Windowing

- The DFT of *any* sequence computes the spectrum for an infinite repetition of that sequence.

- The DFT of a partial segment of a sinusoid computes the spectrum of an infinite repetition of that segment, and not of the entire sinusoid.
The DFT of *any* sequence computes the spectrum for an infinite repetition of that sequence.

The DFT of a partial segment of a sinusoid computes the spectrum of an infinite repetition of that segment, and not of the entire sinusoid.

This will not give us the DFT of the sinusoid itself!
Windowing

Magnitude spectrum of segment

Magnitude spectrum of complete sine wave
The difference occurs due to two reasons:

- The transform cannot know what the signal actually looks like outside the observed window.
The difference occurs due to two reasons:

- The transform cannot know what the signal actually looks like outside the observed window.
- The implicit repetition of the observed signal introduces large discontinuities at the points of repetition.

  These are not part of the underlying signal.

- We only want to characterize the underlying signal.

  The discontinuity is an irrelevant detail.
While we can never know what the signal looks like outside the window, we can try to minimize the discontinuities at the boundaries.

We do this by multiplying the signal with a *window* function. We call this procedure windowing and refer to the resulting signal as a “windowed” signal.
While we can never know what the signal looks like outside the window, we can try to minimize the discontinuities at the boundaries. We do this by multiplying the signal with a *window* function. We call this procedure windowing. We refer to the resulting signal as a “windowed” signal. Windowing attempts to do the following:

- Keep the windowed signal similar to the original in the central regions.
Windowing

While we can never know what the signal looks like outside the window, we can try to minimize the discontinuities at the boundaries.

We do this by multiplying the signal with a window function.

We call this procedure windowing.

We refer to the resulting signal as a “windowed” signal.

Windowing attempts to do the following:

- Keep the windowed signal similar to the original in the central regions.
- Reduce or eliminate the discontinuities in the implicit periodic signal.
The DFT of the windowed signal does not have any artefacts introduced by discontinuities in the signal.

Often it is also a more faithful reproduction of the DFT of the complete signal whose segment we have analyzed.
Windowing

Magnitude spectrum of original segment

Magnitude spectrum of windowed signal

Magnitude spectrum of complete sine wave
Windowing is not a perfect solution

- The original (unwindowed) segment is identical to the original (complete) signal within the segment
- The windowed segment is often not identical to the complete signal anywhere

Several windowing functions have been proposed that strike different tradeoffs between the fidelity in the central regions and the smoothing at the boundaries.
Windowing

- **Cosine windows:**
  - Window length is $M$
  - Index begins at 0
  - Hamming: $w[n] = 0.54 - 0.46 \cos(2\pi n/M)$
  - Hanning: $w[n] = 0.5 - 0.5 \cos(2\pi n/M)$
  - Blackman: $0.42 - 0.5 \cos(2\pi n/M) + 0.08 \cos(4\pi n/M)$
Geometric windows:

- Rectangular (boxcar):

- Triangular (Bartlett):

- Trapezoid:

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We can pad zeros to the end of a signal to make it a desired length

- Useful if the FFT (or any other algorithm we use) requires signals of a specified length
- E.g. Radix 2 FFTs require signals of length $2^n$ i.e., some power of 2. We must zero pad the signal to increase its length to the appropriate number
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The consequence of zero padding is to change the periodic signal whose Fourier spectrum is being computed by the DFT
The DFT of the zero padded signal is essentially the same as the DFT of the unpadded signal, with additional spectral samples inserted in between.

- It does not contain any additional information over the original DFT.
- It also does not contain less information.
Magnitude spectra

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Zero padding windowed signals results in signals that appear to be less discontinuous at the edges.

- This is only illusory.
- Again, we do not introduce any new information into the signal by merely padding it with zeros.
The DFT of the zero padded signal is essentially the same as the DFT of the unpadded signal, with additional spectral samples inserted in between.

- It does not contain any additional information over the original DFT.
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Magnitude spectra
Zero padding a speech signal

128 samples from a speech signal sampled at 16000 Hz

The first 65 points of a 128 point DFT. Plot shows log of the magnitude spectrum

The first 513 points of a 1024 point DFT. Plot shows log of the magnitude spectrum
The process of parameterization

The signal is processed in segments of 25-64 ms

- Because the properties of audio signals change quickly
- They are “stationary” only very briefly
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The process of parameterization

Each segment is typically 25-64 milliseconds wide. Audio signals typically do not change significantly within this short time interval.

Segments shift every 10-16 milliseconds.
The process of parameterization

Each segment is windowed and a DFT is computed from it.
The process of parameterization

Each segment is windowed and a DFT is computed from it.
Computing a Spectrogram

Compute Fourier Spectra of segments of audio and stack them side-by-side

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Computing the Spectrogram

Compute Fourier Spectra of segments of audio and stack them side-by-side. The Fourier spectrum of each window can be inverted to get back the signal. Hence the spectrogram can be inverted to obtain a time-domain signal.

In this example each segment was 25 ms long and adjacent segments overlapped by 15 ms.
The result of parameterization

- Each column here represents the FT of a single segment of signal 64ms wide.
  - Adjacent segments overlap by 48 ms.
- DFT details
  - 1024 points (16000 samples a second).
  - 2048 point DFT – 1024 points of zero padding.
  - Only 1025 points of each DFT are shown
    - The rest are “reflections”
- The value shown is actually the magnitude of the complex spectral values
  - Most of our analysis / operations are performed on the magnitude
Magnitude and phase

\[
\begin{bmatrix}
W_{L}^{0,0} & W_{L}^{L/2,0} & W_{L}^{L-1,0} \\
W_{L}^{0,1} & W_{L}^{L/2,1} & W_{L}^{L-1,1} \\
. & . & . \\
W_{L}^{0,L-1} & W_{L}^{L/2,L-1} & W_{L}^{L-1,L-1}
\end{bmatrix}
\begin{bmatrix}
S_0 \\
S_1 \\
. \\
S_{L-1}
\end{bmatrix}
= 
\begin{bmatrix}
s[0] \\
s[1] \\
. \\
s[L-1]
\end{bmatrix}
\]

\[
S_k = |S_k| \exp(j \text{phase}(S_k))
\]

- All the operations (e.g. the examples shown in the previous class) are performed on the magnitude.
- The phase of the complex spectrum is needed to invert a DFT to a signal.
  - Where does that come from?
- Deriving phase is a serious, not-quite solved problem.
Common tricks: Obtain the phase from the original signal

- \( \text{Sft} = \text{DFT} (\text{signal}) \)
- \( \text{Phase1} = \text{phase} (\text{Sft}) \)
  - Each term is of the form \( \text{real} + j \text{imag} \)
  - For each element, compute \( \arctan(\text{imag}/\text{real}) \)
- \( \text{Smagnitude} = \text{magnitude} (\text{Sft}) \)
  - For each element compute \( \sqrt{\text{real} \times \text{real} + \text{imag} \times \text{imag}} \)
- \( \text{ProcessedSpectrum} = \text{Process} (\text{Smagnitude}) \)
- \( \text{New SFT} = \text{ProcessedSpectrum} \times \exp(j \times \text{Phase}) \)
- Recover signal from SFT

Some other tricks:

- Compute the FT of a different signal of the same length
- Use the phase from that signal
Returning to the speech signal

Actually a matrix of complex numbers

16ms (256 samples)

- For each complex spectral vector, compute a signal from the inverse DFT
  - Make sure to have the complete FT (including the reflected portion)
- If need be window the retrieved signal
- Overlap signals from adjacent vectors in exactly the same manner as during analysis
  - E.g. If a 48ms (768 sample) overlap was used during analysis, overlap adjacent segments by 768 samples
Additional tricks

- The basic representation is the magnitude spectrogram.
- Often it is transformed to a log spectrum:
  - By computing the log of each entry in the spectrogram matrix.
  - After processing, the entry is exponentiated to get back the magnitude spectrum.
  - To which phase may be factored in to get a signal.

- The log spectrum may be “compressed” by a dimensionality reducing matrix:
  - Usually a DCT matrix.