

Chapter 7

The Discrete Fourier Transform: Its Properties and Applications

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Introduction

- Frequency analysis of discrete-time signals is usually and most conveniently performed on a digital signal processor.
- To perform frequency analysis on a discrete-time signal, we convert the time-domain sequence to an equivalent frequency-domain representation.

$$\{x(n)\} \xrightarrow{F} X(\omega)$$

Frequency-Domain Sampling: The Discrete Fourier Transform

- Before we introduce the DFT, we consider the sampling of the Fourier transform of an aperiodic discrete-time sequence.
- Thus, we establish the relationship between the sampled Fourier transform and the DFT.

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3

Frequency-Domain Sampling and Reconstruction of Discrete-Time Signals

- We recall that aperiodic finite-energy signals have continuous spectra.
- Let us consider such an aperiodic discrete-time signal $x(n)$ with Fourier transform

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

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4

Frequency-Domain Sampling and Reconstruction of Discrete-Time Signals

- Suppose that we sample $X(\omega)$ periodically in frequency at a spacing of $\delta\omega$ radius between successive samples.
- Since $X(\omega)$ is periodic with period 2π , only samples in the fundamental frequency range are necessary.

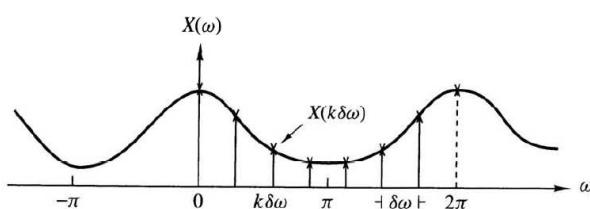
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5

Frequency-Domain Sampling and Reconstruction of Discrete-Time Signals

- For convenience, we take N equidistant samples in the interval $0 \leq \omega < 2\pi$ with spacing $\delta\omega = 2\pi / N$, as shown below:
 - Frequency-domain sampling of the Fourier transform



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6

Frequency-Domain Sampling and Reconstruction of Discrete-Time Signals

- First, we consider the selection of N , the number of samples in the frequency domain.
- If we evaluate $X(\omega) = \sum_{n=-\infty}^{\infty} x(n)e^{-j\omega n}$ at $\omega = 2\pi k / N$
- We obtain

$$X\left(\frac{2\pi}{N}k\right) = \sum_{n=-\infty}^{\infty} x(n)e^{-j2\pi kn/N}, \quad k = 0, 1, \dots, N-1$$

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7

Frequency-Domain Sampling and Reconstruction of Discrete-Time Signals

- The summation above equation can be subdivided into an infinite number of summations, where each sum contains N terms. Thus

$$\begin{aligned} X\left(\frac{2\pi}{N}k\right) &= \dots + \sum_{n=-N}^{-1} x(n)e^{-j2\pi kn/N} + \sum_{n=0}^{N-1} x(n)e^{-j2\pi kn/N} \\ &\quad + \sum_{n=N}^{2N-1} x(n)e^{-j2\pi kn/N} + \dots \\ &= \sum_{l=-\infty}^{\infty} \sum_{n=lN}^{lN+N-1} x(n)e^{-j2\pi kn/N} \end{aligned}$$

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8

Frequency-Domain Sampling and Reconstruction of Discrete-Time Signals

- If we change the index in the inner summation from n to $n - lN$ and interchange the order of the summation, we obtain the result

$$X\left(\frac{2\pi}{N}k\right) = \sum_{n=0}^{N-1} \left[\sum_{l=-\infty}^{\infty} x(n - lN) \right] e^{-j2\pi kn/N} \text{ for } k = 0, 1, 2, \dots, N-1.$$

Frequency-Domain Sampling and Reconstruction of Discrete-Time Signals

- The signal $x_p(n) = \sum_{l=-\infty}^{\infty} x(n - lN)$ obtained by the periodic repetition of $x(n)$ every N samples, is clearly periodic with fundamental period N .
- It can be expanded in a Fourier series as

$$x_p(n) = \sum_{k=0}^{N-1} c_k e^{j2\pi kn/N}, \quad n = 0, 1, \dots, N-1$$

Frequency-Domain Sampling and Reconstruction of Discrete-Time Signals

- With Fourier coefficients

$$c_k = \frac{1}{N} \sum_{n=0}^{N-1} x_p(n) e^{-j2\pi kn/N}, \quad k = 0, 1, \dots, N-1$$

- We can conclude that

$$c_k = \frac{1}{N} X\left(\frac{2\pi}{N}k\right), \quad k = 0, 1, \dots, N-1$$

$$x_p(n) = \frac{1}{N} \sum_{k=0}^{N-1} X\left(\frac{2\pi}{N}k\right) e^{j2\pi kn/N}, \quad n = 0, 1, \dots, N-1$$

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11

Frequency-Domain Sampling and Reconstruction of Discrete-Time Signals

- The relationship in above equation provides the reconstruction of the periodic signal $x_p(n)$ from the samples of the spectrum $X(\omega)$.
- However, it does not imply that we can recover $X(\omega)$ or $x(n)$ from the samples.
- To accomplish this, we need to consider the relationship between $x_p(n)$ and $x(n)$.

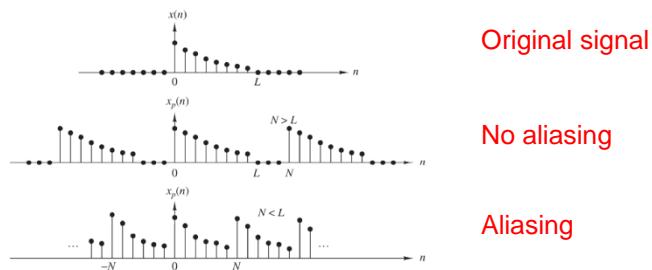
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12

Frequency-Domain Sampling and Reconstruction of Discrete-Time Signals

- Since $x_p(n)$ is the periodic extension of $x(n)$. It is clear that $x(n)$ can be recovered from $x_p(n)$ if there is no **aliasing** in the time domain, that is, if $x(n)$ is time-limited to less than the period N of $x_p(n)$.



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13

Frequency-Domain Sampling and Reconstruction of Discrete-Time Signals

- We conclude that the spectrum of an aperiodic discrete-time signal with finite duration L can be exactly recovered from its samples at frequencies

$$\omega_k = 2\pi k / N, \text{ if } N \geq L$$

- The procedure is to compute $x_p(n)$, $n = 0, 1, \dots, N-1$

then
$$x(n) = \begin{cases} x_p(n), & 0 \leq n \leq N-1 \\ 0, & \text{elsewhere} \end{cases}$$

and finally, $X(\omega)$ can be computed.

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14

Frequency-Domain Sampling and Reconstruction of Discrete-Time Signals

- As in the case of continuous-time signals, it is possible to express the spectrum $X(\omega)$ directly in terms of its samples $X(2\pi k / N)$, $k = 0, 1, \dots, N-1$. To derive such an interpolation formula for $X(\omega)$, we assume that $N \geq L$.
- Since $x(n) = x_p(n)$ for $0 \leq n \leq N-1$

$$x(n) = \frac{1}{N} \sum_{k=0}^{N-1} X\left(\frac{2\pi}{N}k\right) e^{j2\pi kn/N}, \quad 0 \leq n \leq N-1$$

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15

Frequency-Domain Sampling and Reconstruction of Discrete-Time Signals

- If we use $X(\omega) = \sum_{n=-\infty}^{\infty} x(n) e^{-j\omega n}$ and substitute for $x(n)$, we obtain

$$\begin{aligned} X(\omega) &= \sum_{n=0}^{N-1} \left[\frac{1}{N} \sum_{k=0}^{N-1} X\left(\frac{2\pi}{N}k\right) e^{j2\pi kn/N} \right] e^{-j\omega n} \\ &= \sum_{k=0}^{N-1} X\left(\frac{2\pi}{N}k\right) \left[\frac{1}{N} \sum_{n=0}^{N-1} e^{-j(\omega - 2\pi k/N)n} \right] \end{aligned}$$

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16

Frequency-Domain Sampling and Reconstruction of Discrete-Time Signals

- The inner summation term in the brackets of above represents the basic **interpolation function** shifted by $2\pi k / N$ in frequency. Indeed, if we define

$$\begin{aligned} P(\omega) &= \frac{1}{N} \sum_{n=0}^{N-1} e^{-j\omega n} = \frac{1}{N} \frac{1 - e^{-j\omega N}}{1 - e^{-j\omega}} \\ &= \frac{\sin(\omega N / 2)}{N \sin(\omega / 2)} e^{-j\omega(N-1)/2} \end{aligned}$$

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17

Frequency-Domain Sampling and Reconstruction of Discrete-Time Signals

- Then, we can obtain

$$X(\omega) = \sum_{k=0}^{N-1} X\left(\frac{2\pi}{N} k\right) P\left(\omega - \frac{2\pi}{N} k\right), \quad N \geq L$$

- The interpolation function $P(\omega)$ is not the familiar $(\sin \theta) / \theta$ but instead, it is a periodic counterpart of it, and it is due to the periodic nature of $X(\omega)$.

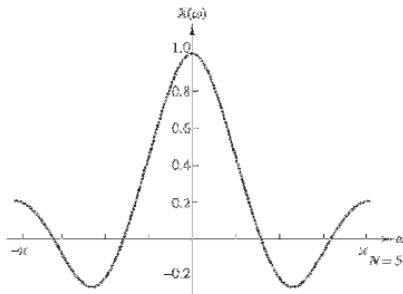
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18

Frequency-Domain Sampling and Reconstruction of Discrete-Time Signals

- The phase shift reflects the fact that the signal $x(n)$ is a causal, finite-duration sequence of length N . The function $(\sin \omega N / 2) / (N \sin(\omega / 2))$ is plotted for $N = 5$.



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19

Frequency-Domain Sampling and Reconstruction of Discrete-Time Signals

- We observe that the function $P(\omega)$ has the property

$$P\left(\frac{2\pi}{N}k\right) = \begin{cases} 1, & k = 0 \\ 0, & k = 1, 2, \dots, N-1 \end{cases}$$

- Consequently, the interpolation formula gives exactly the sample values $X(2\pi k / N)$ for $\omega = 2\pi k / N$. At all other frequencies, the formula provides a properly weighted linear combination of the original spectral samples.

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20

Frequency-Domain Sampling and Reconstruction of Discrete-Time Signals

- **Example:** Consider the signal $x(n) = a^n u(n)$, $0 < a < 1$. The spectrum of this signal is sampled at frequencies $\omega_k = 2\pi k / N$ for $k = 0, 1, \dots, N-1$.
Determine the reconstructed spectra for $a = 0.8$ when $N=5$ and $N=50$.
- **Solution:** The Fourier transform of the sequence $x(n)$ is

$$X(\omega) = \sum_{n=0}^{\infty} a^n e^{-j\omega n} = \frac{1}{1 - ae^{-j\omega}}$$

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21

Frequency-Domain Sampling and Reconstruction of Discrete-Time Signals

- Suppose that we sample $X(\omega)$ at N equidistant frequencies $\omega_k = 2\pi k / N$, $k = 0, 1, \dots, N-1$. Thus we obtain the spectral samples
$$X(\omega k) \equiv X\left(\frac{2\pi k}{N}\right) = \frac{1}{1 - ae^{-j2\pi k/N}}, \quad k = 0, 1, \dots, N-1$$
- The periodic sequence $x_p(n)$, corresponding to the frequency samples $X(2\pi k / N)$, $k = 0, 1, \dots, N-1$ can be obtained.

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22

Frequency-Domain Sampling and Reconstruction of Discrete-Time Signals

■ Hence $x_p(n) = \sum_{l=-\infty}^{\infty} x(n-lN) = \sum_{l=-\infty}^0 a^{n-lN}$
 $= a^n \sum_{l=0}^{\infty} a^{lN} = \frac{a^n}{1-a^N}, \quad 0 \leq n \leq N-1$

where the factor $1/(1-a^N)$ represents the effect of aliasing. Since $0 < a < 1$, the aliasing error tends toward zero as $N \rightarrow \infty$.

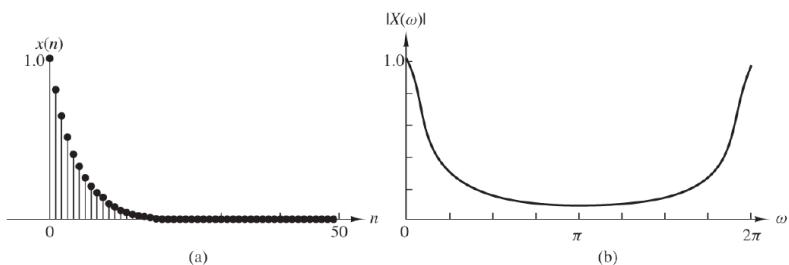
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23

Frequency-Domain Sampling and Reconstruction of Discrete-Time Signals

- For $a = 0.8$, the sequence $x(n)$ and its spectrum $X(\omega)$ are shown below (a)(b):



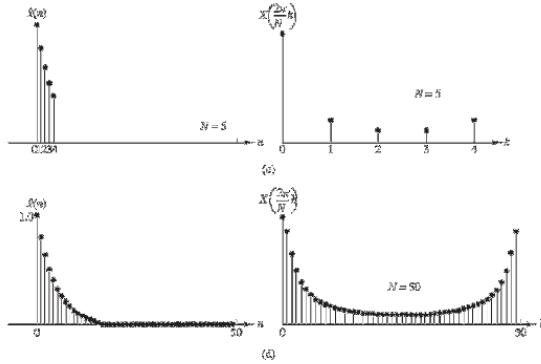
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24

Frequency-Domain Sampling and Reconstruction of Discrete-Time Signals

- The aliased sequences $x_p(n)$ for $N = 5$ and $N = 50$ and the corresponding spectral samples are shown below (c)(d).



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25

Frequency-Domain Sampling and Reconstruction of Discrete-Time Signals

- We note that the aliasing effects are negligible for $N=50$.
- If we define the aliased finite-duration sequence $x(n)$ as

$$\hat{x}(n) = \begin{cases} x_p(n), & 0 \leq n \leq N-1 \\ 0, & \text{otherwise} \end{cases}$$

- Then its Fourier transform is

$$\hat{X}(\omega) = \sum_{n=0}^{N-1} \hat{x}(n) e^{-j\omega n} = \sum_{n=0}^{N-1} x_p(n) e^{-j\omega N} = \frac{1}{1-a^N} \cdot \frac{1-a^N e^{-j\omega N}}{1-a e^{-j\omega}}$$

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26

Frequency-Domain Sampling and Reconstruction of Discrete-Time Signals

- Note that although $\hat{X}(\omega) \neq X(\omega)$, the sample values at $\omega_k = 2\pi k / N$ are identical. That is,

$$\hat{X}\left(\frac{2\pi}{N}k\right) = \frac{1}{1-a^N} \cdot \frac{1-a^N}{1-ae^{-j2\pi kN}} = X\left(\frac{2\pi}{N}k\right)$$

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27

The Discrete Fourier Transform (DFT)

- The development in the preceding section is concerned with the frequency-domain sampling of an aperiodic finite-energy sequence $x(n)$.
- In general, the equally spaced frequency samples

$$X(2\pi k / N), \quad k = 0, 1, \dots, N-1,$$

do not uniquely represent the original sequence $x(n)$ when $x(n)$ has infinite duration.

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28

The Discrete Fourier Transform (DFT)

- Instead, the frequency samples

$$X(2\pi k / N), k = 0, 1, \dots, N-1,$$

correspond to a periodic sequence $x_p(n)$ of period N , where $x_p(n)$ is an aliased version of $x(n)$, as indicated by the relation in the preceding equation, that is,

$$x_p(n) = \sum_{l=-\infty}^{\infty} x(n-lN).$$

The Discrete Fourier Transform (DFT)

- When the sequence $x(n)$ has a finite duration of length $L \leq N$, then $x_p(n)$ is simply a periodic repetition of $x(n)$, where $x_p(n)$ over a single period is given as

$$x_p(n) = \begin{cases} x(n), & 0 \leq n \leq L-1 \\ 0, & L \leq n \leq N-1 \end{cases}$$

- Consequently, the frequency samples

$$X(2\pi k / N), k = 0, 1, \dots, N-1,$$

uniquely represent the finite-duration sequence $x(n)$.

The Discrete Fourier Transform (DFT)

- Since $x(n) \equiv x_p(n)$ over a single period (padded by $N-L$ zeros), the original finite-duration sequence $x(n)$ can be obtained from the frequency samples $\{X(2\pi k / N)\}$ by means of the formula

$$x_p(n) = \frac{1}{N} \sum_{k=0}^{N-1} X\left(\frac{2\pi}{N} k\right) e^{j2\pi kn/N}, \quad n = 0, 1, \dots, N-1.$$

The Discrete Fourier Transform (DFT)

- Note that *zero padding* does not provide any additional information about the spectrum $X(\omega)$ of the sequence $\{x(n)\}$.
- In summary, a finite-duration sequence $x(n)$ of length L has a Fourier transform

$$X(\omega) = \sum_{n=0}^{L-1} x(n) e^{-j\omega n}, \quad 0 \leq \omega \leq 2\pi$$

- Where the upper and lower indices in the summation reflect the fact that $x(n) = 0$ outside the range $0 \leq n \leq L-1$.

The Discrete Fourier Transform (DFT)

- When we sample $X(\omega)$ at equally spaced frequencies

$$\omega_k = 2\pi k / N, \quad k = 0, 1, 2, \dots, N-1, \text{ where } N \geq L$$

the resultant samples are

$$X(k) \equiv X\left(\frac{2\pi k}{N}\right) = \sum_{n=0}^{L-1} x(n) e^{-j2\pi kn/N}$$

$$X(k) = \sum_{n=0}^{N-1} x(n) e^{-j2\pi kn/N}, \quad k = 0, 1, 2, \dots, N-1$$

where for convenience, the upper index in the sum has been increased from $L-1$ to $N-1$ since $x(n)=0$ for $n \geq L$.

The Discrete Fourier Transform (DFT)

- The relation

$$X(k) \equiv X\left(\frac{2\pi k}{N}\right) = \sum_{n=0}^{L-1} x(n) e^{-j2\pi kn/N}$$

$$X(k) = \sum_{n=0}^{N-1} x(n) e^{-j2\pi kn/N}, \quad k = 0, 1, 2, \dots, N-1$$

is called the **discrete Fourier transform (DFT)** of $x(n)$.

The Discrete Fourier Transform (DFT)

- To summarize, the formulas for the DFT and IDFT are

$$\text{DFT: } X(k) = \sum_{n=0}^{N-1} x(n)e^{-j2\pi kn/N}, \quad k = 0, 1, 2, \dots, N-1$$

$$\text{IDFT: } x(n) = \frac{1}{N} \sum_{k=0}^{N-1} X(k)e^{j2\pi kn/N}, \quad n = 0, 1, 2, \dots, N-1$$

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35

The Discrete Fourier Transform (DFT)

- **Example:** A finite-duration sequence of length L is given

as

$$x(n) = \begin{cases} 1, & 0 \leq n \leq L-1 \\ 0, & \text{otherwise} \end{cases}$$

Determine the N -point DFT of this sequence for $N \geq L$.

- **Solution:** The Fourier transform of this sequence is

$$\begin{aligned} X(\omega) &= \sum_{n=0}^{L-1} x(n)e^{-j\omega n} = \sum_{n=0}^{L-1} e^{-j\omega n} \\ &= \frac{1 - e^{-j\omega L}}{1 - e^{-j\omega}} = \frac{\sin(\omega L/2)}{\sin(\omega/2)} e^{-j\omega(L-1)/2} \end{aligned}$$

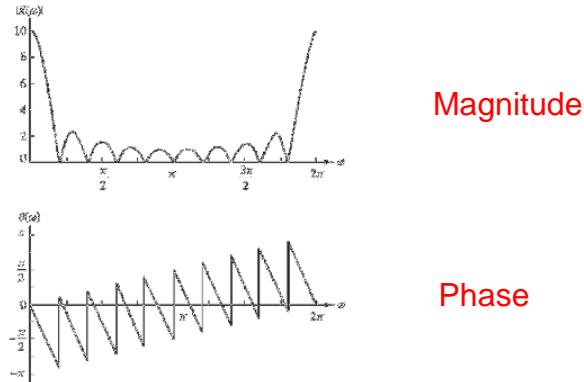
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36

The Discrete Fourier Transform (DFT)

- The magnitude and phase of $X(\omega)$ are illustrated in the below for $L=10$.



2010/6/12

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37

The Discrete Fourier Transform (DFT)

- The N -point DFT of $x(n)$ is simply $X(\omega)$ evaluated at the set of N equally spaced frequencies

$$\omega_k = 2\pi k / N, \quad k = 0, 1, \dots, N-1.$$

- Hence

$$\begin{aligned} X(k) &= \frac{1 - e^{-j2\pi kL/N}}{1 - e^{-j2\pi k/N}}, \quad k = 0, 1, \dots, N-1 \\ &= \frac{\sin(\pi kL / N)}{\sin(\pi k / N)} e^{-j\pi k(L-1)/N} \end{aligned}$$

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38

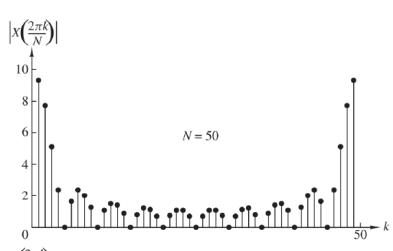
The Discrete Fourier Transform (DFT)

- If N is selected such that $N=L$, then the DFT becomes

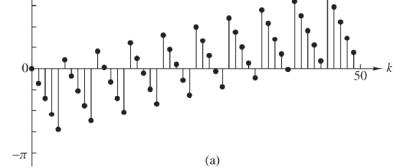
$$X(k) = \begin{cases} L, & k = 0 \\ 0, & k = 1, 2, \dots, L-1 \end{cases}$$

The Discrete Fourier Transform (DFT)

N=50

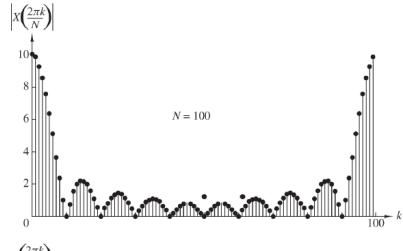


N=50

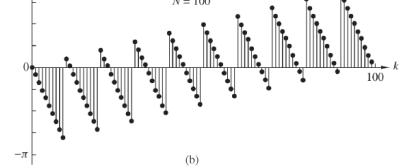


(a)

N=100



N=100



(b)

The DFT as a Linear Transformation

- The formulas for the DFT and IDFT may be expressed as

$$X(k) = \sum_{n=0}^{N-1} x(n)W_N^{kn}, \quad k = 0, 1, 2, \dots, N-1$$

$$x(n) = \frac{1}{N} \sum_{k=0}^{N-1} X(k)W_N^{-kn}, \quad n = 0, 1, 2, \dots, N-1$$

where $W_N = e^{-j2\pi/N}$

which is an N^{th} root of unity.

2010/6/12

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41

The DFT as a Linear Transformation

- We note that the computation of each point of the DFT can be accomplished by N complex multiplications and $(N-1)$ complex additions.
- Hence the N -point DFT values can be computed in a total of N^2 complex multiplications and $N(N-1)$ complex additions.
- It is instructive to view the DFT and IDFT as linear transformations on sequences $\{x(n)\}$ and $\{X(k)\}$, respectively.

2010/6/12

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42

The DFT as a Linear Transformation

- Let us define an N -point vector \mathbf{x}_N of the signal sequence $x(n)$, $n=0,1,\dots,N-1$, an N -point vector \mathbf{X}_N of frequency samples, and an $N \times N$ matrix \mathbf{W}_N as

$$\mathbf{x}_N = \begin{bmatrix} x(0) \\ x(1) \\ \vdots \\ x(N-1) \end{bmatrix}, \mathbf{X}_N = \begin{bmatrix} \mathbf{X}(0) \\ \mathbf{X}(1) \\ \vdots \\ \mathbf{X}(N-1) \end{bmatrix}, \mathbf{W}_N = \begin{bmatrix} 1 & 1 & 1 & \cdots & 1 \\ 1 & W_N & W_N^2 & \cdots & W_N^{N-1} \\ \vdots & W_N^2 & W_N^4 & \cdots & W_N^{2(N-1)} \\ \vdots & \vdots & \vdots & \cdots & \vdots \\ 1 & W_N^{N-1} & W_N^{2(N-1)} & \cdots & W_N^{(N-1)(N-1)} \end{bmatrix}$$

- With these definitions, the N -point DFT may be expressed in matrix forms as $\mathbf{X}_N = \mathbf{W}_N \mathbf{x}_N$.
- Where \mathbf{W}_N is the matrix of the linear transformation.

The DFT as a Linear Transformation

- We observe that \mathbf{W}_N is a symmetric matrix.
- If we assume that the inverse of \mathbf{W}_N exists, then

$$\mathbf{x}_N = \mathbf{W}_N^{-1} \mathbf{X}_N$$

- But this is just an expression for the IDFT.
- In fact, the IDFT can be expressed in matrix form as

$$\mathbf{x}_N = \frac{1}{N} \mathbf{W}_N^* \mathbf{X}_N$$

The DFT as a Linear Transformation

- Where \mathbf{W}_N^* denotes the complex conjugate of the matrix \mathbf{W}_N . Then we can conclude that

$$\mathbf{W}_N^{-1} = \frac{1}{N} \mathbf{W}_N^*$$

- Which, in turn, implies that

$$\mathbf{W}_N \mathbf{W}_N^* = N \mathbf{I}_N$$

- Where \mathbf{I}_N is an $N \times N$ identity matrix.

2010/6/12

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45

The DFT as a Linear Transformation

- Therefore, the matrix \mathbf{W}_N in the transformation is an orthogonal (unitary) matrix.
- Furthermore, its inverse exists and is given as \mathbf{W}_N^* / N .
- Of course, the existence of the inverse of \mathbf{W}_N was established previously from our derivation of the IDFT.

2010/6/12

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46

The DFT as a Linear Transformation

- The DFT and IDFT are computational tools that play a very important role in many digital signal processing applications, such as frequency analysis (spectrum analysis) of signals, power spectrum estimation, and linear filtering.
- The importance of the DFT and IDFT in such practical applications is due to a large extent to the existence of computationally efficient algorithms, known collectively as fast Fourier transform (FFT) algorithms, for computing the DFT and IDFT.

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47

Relationship of the DFT to Other Transforms

- **Relationship to the Fourier series coefficients of a periodic sequence**
- A periodic sequence $\{x_p(n)\}$ with fundamental period N can be represented in a Fourier series of the form
$$x_p(n) = \sum_{k=0}^{N-1} c_k e^{j2\pi kn/N}, \quad -\infty < n < \infty$$
$$c_k = \frac{1}{N} \sum_{n=0}^{N-1} x_p(n) e^{-j2\pi kn/N}, \quad k = 0, 1, \dots, N-1$$
- Where c_k is the Fourier series coefficients.

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48

Relationship of the DFT to Other Transforms

- **Relationship to the Fourier transform of an aperiodic sequence**

- If $x(n)$ is an aperiodic finite energy sequence with Fourier transform $X(\omega)$, which is sampled at N equally spaced frequencies $\omega_k = 2\pi k / N$, $k = 0, 1, \dots, N-1$, the spectral components

$$X(k) = X(\omega) \big|_{\omega=2\pi k/N} = \sum_{n=-\infty}^{\infty} x(n) e^{-j2\pi kn/N}, \quad k = 0, 1, \dots, N-1$$

are the DFT coefficients of the periodic sequence of period N .

2010/6/12

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49

Relationship of the DFT to Other Transforms

- **Relationship to the Fourier transform of an aperiodic sequence**

$$x_p(n) = \sum_{l=-\infty}^{\infty} x(n-lN)$$

- Thus $x_p(n)$ is determined by aliasing $\{x(n)\}$ over the interval $0 \leq n \leq N-1$. The finite-duration sequence

$$\hat{x}(n) = \begin{cases} x_p(n), & 0 \leq n \leq N-1 \\ 0, & \text{otherwise} \end{cases}$$

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50

Relationship of the DFT to Other Transforms

- Relationship to the Fourier transform of an aperiodic sequence
- Bears no resemblance to the original sequence $\{x(n)\}$, unless $x(n)$ is of finite duration and length $L \leq N$, in which case $x(n) = \hat{x}(n)$, $0 \leq n \leq N - 1$
- Only in this case will the IDFT of $\{X(k)\}$ yield the original sequence $\{x(n)\}$.

2010/6/12

Introduction to Digital Signal Processing

51

Relationship of the DFT to Other Transforms

- Relationship to the Fourier transform of an aperiodic sequence
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2010/6/12

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52

Relationship of the DFT to Other Transforms

- **Relationship to the z-transform**
- Consider a sequence $x(n)$ having the z-transform

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

with an ROC that includes the unit circle.

2010/6/12

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53

Relationship of the DFT to Other Transforms

- **Relationship to the z-transform**
- If $X(z)$ is sampled at the N equally spaced points on the unit circle $z_k = e^{j2\pi k/N}$, $k = 0, 1, 2, \dots, N-1$, we obtain

$$\begin{aligned} X(k) &\equiv X(z) \Big|_{z=e^{j2\pi k/N}}, \quad k = 0, 1, \dots, N-1 \\ &= \sum_{n=-\infty}^{\infty} x(n)e^{-j2\pi nk/N} \end{aligned}$$

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54

Relationship of the DFT to Other Transforms

- Relationship to the Fourier series coefficients of a continuous-time signal
- Suppose that $x_a(t)$ is a continuous-time periodic signal with fundamental period $T_p = 1/F_0$. The signal can be expressed in a Fourier series

$$x_a(t) = \sum_{k=-\infty}^{\infty} c_k e^{j2\pi k t F_0}$$

- Where $\{c_k\}$ are the Fourier coefficients.

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55

Relationship of the DFT to Other Transforms

- Relationship to the Fourier series coefficients of a continuous-time signal
- If we sample $x_a(t)$ at a uniform rate $F_s = N/T_p = 1/T$, we obtain the discrete-time sequence

$$\begin{aligned} x(n) \equiv x_a(nt) &= \sum_{k=-\infty}^{\infty} c_k e^{j2\pi k F_0 n T} = \sum_{k=-\infty}^{\infty} c_k e^{j2\pi k n / N} \\ &= \sum_{k=0}^{N-1} \left[\sum_{l=-\infty}^{\infty} c_{k-lN} \right] e^{j2\pi k n / N} \end{aligned}$$

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56

Relationship of the DFT to Other Transforms

- Relationship to the Fourier series coefficients of a continuous-time signal

- It is clear the above equation is in the form of an IDFT formula, where

$$X(k) = N \sum_{l=-\infty}^{\infty} c_{k-lN} \equiv N \tilde{c}_k \quad \text{and} \quad \tilde{c}_k = \sum_{l=-\infty}^{\infty} c_{k-lN}$$

- Thus the $\{\tilde{c}_k\}$ sequence is an aliased version of the sequence $\{c_k\}$.

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57

Properties of the DFT

- The DFT as a set of N samples $\{X(k)\}$ of the Fourier transform $X(\omega)$ for a finite-duration sequence $\{x(n)\}$ of length $L \leq N$.
- The sampling of $X(\omega)$ occurs at the N equally spaced frequencies $\omega_k = 2\pi k / N$, $k = 0, 1, 2, \dots, N-1$.

$$\text{DFT: } X(k) = \sum_{n=0}^{N-1} x(n) W_N^{kn}, \quad k = 0, 1, 2, \dots, N-1$$

$$\text{IDFT: } x(n) = \frac{1}{N} \sum_{k=0}^{N-1} X(k) W_N^{-kn}, \quad n = 0, 1, 2, \dots, N-1$$

$$\text{where } W_N = e^{-j2\pi/N}$$

$$x(n) \xrightarrow[N]{DFT} X(k)$$

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58

Periodicity, Linearity, and Symmetry Properties

- **Periodicity.** If $x(n)$ and $X(k)$ are an N -point DFT pair, then

$$x(n+N) = x(n) \quad \forall n$$

$$X(k+N) = X(k) \quad \forall k$$

- **Linearity.** If $x_1(n) \xrightarrow[N]{DFT} X_1(k)$ and $x_2(n) \xrightarrow[N]{DFT} X_2(k)$

then for any real-valued or complex-valued constants a_1 and a_2 ,

$$a_1 x_1(n) + a_2 x_2(n) \xrightarrow[N]{DFT} a_1 X_1(k) + a_2 X_2(k)$$

2010/6/12

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59

Periodicity, Linearity, and Symmetry Properties

- **Circular Symmetries of a Sequence.** As we have seen, the N -point DFT of a finite duration sequence $x(n)$, of length $L \leq N$, is equivalent to the N -point DFT of a periodic sequence $x_p(n)$, of period N , which is obtained by periodically extending $x(n)$, that is

$$x_p(n) = \sum_{l=-\infty}^{\infty} x(n-lN)$$

2010/6/12

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60

Periodicity, Linearity, and Symmetry Properties

- **Circular Symmetries of a Sequence.** Suppose that we shift the periodic sequence $x_p(n)$ by k units to the right. Thus we obtain another periodic sequence

$$x_p(n) = x_p(n - k) = \sum_{l=-\infty}^{\infty} x(n - k - lN)$$

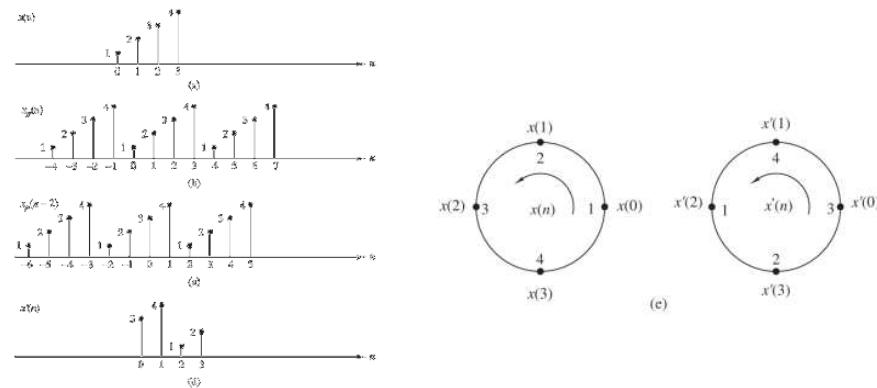
- The finite-duration sequence

$$x'(n) = \begin{cases} x_p(n), & 0 \leq n \leq N-1 \\ 0, & \text{otherwise} \end{cases}$$

is related to the original sequence $x(n)$ by a circular shift.

Periodicity, Linearity, and Symmetry Properties

- **Circular Symmetries of a Sequence.** This relationship is illustrated as below for $N=4$.



Periodicity, Linearity, and Symmetry Properties

- **Circular Symmetries of a Sequence.**
- In general, the circular shift of the sequence can be represented as the index modulo N . thus we can write

$$\begin{aligned}x'(n) &= x(n-k, \text{ modulo } N) \\&= x((n-k))_N\end{aligned}$$

- Time reversal of N -point sequence

$$x((-n))_N = x(N-n), \quad 0 \leq n \leq N-1$$

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63

Periodicity, Linearity, and Symmetry Properties

- **Circular Symmetries of a Sequence.** An equivalent definition of even and odd sequences for the associated periodic sequence $x_p(n)$ is given as follows

$$\text{Even: } x_p(n) = x_p(-n) = x_p(N-n)$$

$$\text{Odd: } x_p(n) = -x_p(-n) = -x_p(N-n)$$

- If the periodic sequence is complex valued, we have

$$\text{Conjugate even: } x_p(n) = x_p^*(N-n)$$

$$\text{Conjugate odd: } x_p(n) = -x_p^*(N-n)$$

2010/6/12

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64

Periodicity, Linearity, and Symmetry Properties

- **Circular Symmetries of a Sequence.** These relationships suggest that we decompose the sequence $x_p(n)$ as

$$x_p(n) = x_{pe}(n) + x_{po}(n)$$

where
$$x_{pe}(n) = \frac{1}{2}[x_p(n) + x_p^*(N-n)]$$
$$x_{po}(n) = \frac{1}{2}[x_p(n) - x_p^*(N-n)]$$

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65

Periodicity, Linearity, and Symmetry Properties

- **Symmetry properties of the DFT.** The sequences can be expressed as

$$x(n) = x_R(n) + jx_I(n), \quad 0 \leq n \leq N-1$$

$$X(k) = X_R(k) + jX_I(k), \quad 0 \leq k \leq N-1$$

- We can obtain

$$X_R(k) = \sum_{n=0}^{N-1} \left[x_R(n) \cos \frac{2\pi kn}{N} + x_I(n) \sin \frac{2\pi kn}{N} \right]$$
$$X_I(k) = - \sum_{n=0}^{N-1} \left[x_R(n) \sin \frac{2\pi kn}{N} - x_I(n) \cos \frac{2\pi kn}{N} \right]$$

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66

Periodicity, Linearity, and Symmetry Properties

- Symmetry properties of the DFT.

- Similarly,

$$x_R(n) = \frac{1}{N} \sum_{k=0}^{N-1} \left[X_R(k) \cos \frac{2\pi kn}{N} - X_I(k) \sin \frac{2\pi kn}{N} \right]$$
$$x_I(k) = \frac{1}{N} \sum_{k=0}^{N-1} \left[X_R(k) \sin \frac{2\pi kn}{N} + X_I(k) \cos \frac{2\pi kn}{N} \right]$$

Periodicity, Linearity, and Symmetry Properties

- **Real-valued sequences.** If the sequence $x(n)$ is real, it follows

$$X(N-k) = X^*(k) = X(-k)$$

- Consequently,

$$|X(N-k)| = |X(k)|$$
$$\angle X(N-k) = -\angle X(k)$$

Periodicity, Linearity, and Symmetry Properties

- **Real and even sequences.** If $x(n)$ is real and even, that is

$$x(n) = x(N-n), \quad 0 \leq n \leq N-1$$

- And $X_I(k) = 0$. Hence the DFT reduces to

$$X(k) = \sum_{n=0}^{N-1} x(n) \cos \frac{2\pi kn}{N}, \quad 0 \leq k \leq N-1$$

- IDFT $x(n) = \frac{1}{N} \sum_{k=0}^{N-1} X(k) \cos \frac{2\pi kn}{N}, \quad 0 \leq n \leq N-1$

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69

Periodicity, Linearity, and Symmetry Properties

- **Real and odd sequences.** If $x(n)$ is real and odd, that is

$$x(n) = -x(N-n), \quad 0 \leq n \leq N-1$$

- And $X_R(k) = 0$. Hence the DFT reduces to

$$X(k) = -j \sum_{n=0}^{N-1} x(n) \sin \frac{2\pi kn}{N}, \quad 0 \leq k \leq N-1$$

- IDFT $x(n) = j \frac{1}{N} \sum_{k=0}^{N-1} X(k) \sin \frac{2\pi kn}{N}, \quad 0 \leq n \leq N-1$

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70

Periodicity, Linearity, and Symmetry Properties

- **Purely imaginary sequences.** $x(n) = jx_I(n)$

$$X_R(k) = \sum_{n=0}^{N-1} x_I(n) \sin \frac{2\pi kn}{N}$$

$$X_I(k) = \sum_{n=0}^{N-1} x_I(n) \cos \frac{2\pi kn}{N}$$

- We observe that $X_R(k)$ is odd and $X_I(k)$ is even.

2010/6/12

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71

Periodicity, Linearity, and Symmetry Properties

N -Point Sequence $x(n)$, $0 \leq n \leq N-1$	N -Point DFT
$x(n)$	$X(k)$
$x^*(n)$	$X^*(N-k)$
$x^*(N-n)$	$X^*(k)$
$x_R(n)$	$X_{ce}(k) = \frac{1}{2}[X(k) + X^*(N-k)]$
$jX_I(n)$	$X_{co}(k) = \frac{1}{2}[X(k) - X^*(N-k)]$
$x_{ce}(n) = \frac{1}{2}[x(n) + x^*(N-n)]$	$X_R(k)$
$x_{co}(n) = \frac{1}{2}[x(n) - x^*(N-n)]$	$jX_I(k)$
Real Signals	
Any real signal	$X(k) = X^*(N-k)$
$x(n)$	$X_R(k) = X_R(N-k)$
	$X_I(k) = -X_I(N-k)$
	$ X(k) = X(N-k) $
	$\angle X(k) = -\angle X(N-k)$
$x_{ce}(n) = \frac{1}{2}[x(n) + x(N-n)]$	$X_R(k)$
$x_{co}(n) = \frac{1}{2}[x(n) - x(N-n)]$	$jX_I(k)$

2010/6/12

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72

Multiplication of Two DFTs and Circular Convolution

- Suppose that we have two finite-duration sequences of length N , $x_1(n)$ and $x_2(n)$. Their respective N -point DFTs are

$$X_1(k) = \sum_{n=0}^{N-1} x_1(n) e^{-j2\pi nk/N}, \quad k = 0, 1, \dots, N-1$$

$$X_2(k) = \sum_{n=0}^{N-1} x_2(n) e^{-j2\pi nk/N}, \quad k = 0, 1, \dots, N-1$$

- If we multiply the two DFTs together, the result is a DFT, say $X_3(k)$, of a sequence $x_3(n)$ of length N .

$$X_3(k) = X_1(k)X_2(k), \quad k = 0, 1, \dots, N-1$$

2010/6/12

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73

Multiplication of Two DFTs and Circular Convolution

- The IDFT of $\{X_3(k)\}$ is

$$x_3(m) = \frac{1}{N} \sum_{k=0}^{N-1} X_3(k) e^{j2\pi km/N} = \frac{1}{N} \sum_{k=0}^{N-1} X_1(k)X_2(k) e^{j2\pi km/N}$$

- Suppose that we substitute for $X_1(k)$ and $X_2(k)$ using the DFTs, thus we obtain

$$\begin{aligned} x_3(m) &= \frac{1}{N} \sum_{k=0}^{N-1} \left[\sum_{n=0}^{N-1} x_1(n) e^{-j2\pi kn/N} \right] \left[\sum_{l=0}^{N-1} x_2(l) e^{-j2\pi ln/N} \right] e^{j2\pi km/N} \\ &= \frac{1}{N} \sum_{n=0}^{N-1} x_1(n) \sum_{l=0}^{N-1} x_2(l) \left[\sum_{k=0}^{N-1} e^{j2\pi k(m-n-l)/N} \right] \end{aligned}$$

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74

Multiplication of Two DFTs and Circular Convolution

- The inner sum in the brackets in the above equation has the form

$$\sum_{k=0}^{N-1} a^k = \begin{cases} N, & a = 1 \\ \frac{1-a^N}{1-a}, & a \neq 1 \end{cases}$$

where a is defined as $a = e^{j2\pi(m-n-l)/N}$

- We observe that $a = 1$ when $m - n - l$ is a multiple of N .

2010/6/12

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75

Multiplication of Two DFTs and Circular Convolution

- On the other hand, $a^N = 1$ for any value of $a \neq 0$.

- Consequently, the above equation reduces to

$$\sum_{k=0}^{N-1} a^k = \begin{cases} N, & l = m - n + pN = ((m - n))_N, p \text{ an integer} \\ 0, & \text{otherwise} \end{cases}$$

- Then we obtain the desired expression for $x_3(m)$ in the form

$$x_3(m) = \sum_{n=0}^{N-1} x_1(n)x_2((m - n))_N, \quad m = 0, 1, \dots, N - 1$$

2010/6/12

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76

Multiplication of Two DFTs and Circular Convolution

- **Circular convolution.** If

$$x_1(n) \xrightarrow[N]{DFT} X_1(k) \text{ and } x_2(n) \xrightarrow[N]{DFT} X_2(k)$$

then $x_1(n) \odot x_2(n) \xrightarrow[N]{DFT} X_1(k)X_2(k)$

where $x_1(n) \odot x_2(n)$ denotes the circular convolution of the sequence $x_1(n)$ and $x_2(n)$.

2010/6/12

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77

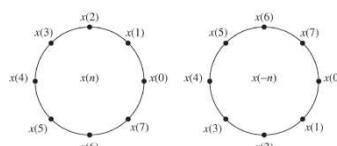
Additional DFT Properties

- **Time reversal of a sequence.** If

$$x(n) \xrightarrow[N]{DFT} X(k)$$

then $x((-n))_N = x(N-n) \xrightarrow[N]{DFT} X((-k))_N = X(N-k)$

- Hence reversing the N -point sequence in time is equivalent to reversing the DFT values.



2010/6/12

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78

Additional DFT Properties

- **Circular time shift of a sequence.** If

$$x(n) \xrightarrow[N]{DFT} X(k)$$

then $x((n-l))_N \xrightarrow[N]{DFT} X(k)e^{-j2\pi kl/N}$

- **Circular frequency shift.** If

$$x(n) \xrightarrow[N]{DFT} X(k)$$

$$x(n)e^{j2\pi ln/N} \xrightarrow[N]{DFT} X((k-l))_N$$

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79

Additional DFT Properties

- **Complex-conjugate properties.** If

$$x(n) \xrightarrow[N]{DFT} X(k)$$

then $x^*(n) \xrightarrow[N]{DFT} X^*((-k))_N = X^*(N-k)$

- The IDFT of $X^*(k)$ is

$$\frac{1}{N} \sum_{k=0}^{N-1} X^*(k) e^{j2\pi kn/N} = \left[\frac{1}{N} \sum_{k=0}^{N-1} X(k) e^{j2\pi k(N-n)/N} \right]$$

therefore, $x^*((-n))_N = x^*(N-n) \xrightarrow[N]{DFT} X^*(k)$

2010/6/12

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80

Additional DFT Properties

- **Circular correlation.** If

$$x(n) \xrightarrow[N]{DFT} X(k) \text{ and } y(n) \xrightarrow[N]{DFT} Y(k)$$

then $\tilde{r}_{xy}(l) \xrightarrow[N]{DFT} \tilde{R}_{xy}(k) = X(k)Y^*(k)$

where $\tilde{r}_{xy}(l)$ is the (unnormalized) circular crosscorrelation sequence, defined as

$$\tilde{r}_{xy}(l) = \sum_{n=0}^{N-1} x(n)y^*((n-l))_N$$

Additional DFT Properties

- **Multiplication of two sequences.** If

$$x_1(n) \xrightarrow[N]{DFT} X_1(k) \text{ and } x_2(n) \xrightarrow[N]{DFT} X_2(k)$$

then $x_1(n)x_2(n) \xrightarrow[N]{DFT} \frac{1}{N} X_1(k) \bigcircledast X_2(k)$

- **Parseval's Theorem.** If

$$x(n) \xrightarrow[N]{DFT} X(k) \text{ and } y(n) \xrightarrow[N]{DFT} Y(k)$$

then $\sum_{n=0}^{N-1} x(n)y^*(n) = \frac{1}{N} \sum_{k=0}^{N-1} X(k)Y^*(k)$

Linear Filtering Methods based on the DFT

- The DFT provides a discrete frequency representation of a finite-duration sequence in the frequency domain, it explores its use as a computational tool for linear system analysis and, especially, for linear filtering.
- We have already established that a system with frequency response $H(\omega)$, when excited with an input signal that has a spectrum $X(\omega)$, possesses an output spectrum $Y(\omega) = X(\omega)H(\omega)$.
- The output sequence $y(n)$ is determined from its spectrum via the inverse Fourier transform.

2010/6/12

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83

Use of the DFT in Linear Filtering

- Suppose that we have a finite-duration sequence $x(n)$ of length L which excites an FIR filter of length M . Without loss of generality, let

$$x(n) = 0, \quad n < 0 \text{ and } n \geq L$$

$$h(n) = 0, \quad n < 0 \text{ and } n \geq M$$

where $h(n)$ is the impulse response of the FIR filter.

- The output sequence

$$y(n) = \sum_{k=0}^{M-1} h(k)x(n-k)$$

2010/6/12

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84

Use of the DFT in Linear Filtering

- The frequency-domain equivalent to the above is

$$Y(\omega) = X(\omega)H(\omega)$$

- If
$$Y(k) \equiv Y(\omega) \big|_{\omega=2\pi k/N}, \quad k = 0, 1, \dots, N-1$$
$$= X(\omega)H(\omega) \big|_{\omega=2\pi k/N}, \quad k = 0, 1, \dots, N-1$$

then
$$Y(k) = X(k)H(k), \quad k = 0, 1, \dots, N-1$$

where $\{X(k)\}$ and $\{H(k)\}$ are the N -point DFTs of the corresponding sequences $x(n)$ and $h(n)$.

2010/6/12

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85

Filtering of Long Data Sequences

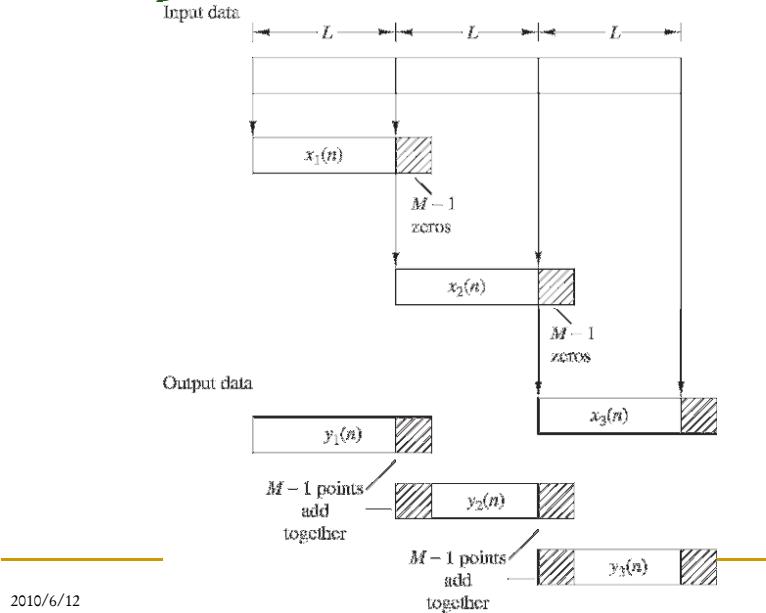
- In practical applications involving linear filtering of signals, the input sequence $x(n)$ is often a very long sequence. This is especially true in some real-time signal processing applications concerned with signal monitoring and analysis.
- Since linear filtering performed via the DFT involves operations on a block of data, which by necessity must be limited in size due to limited memory of a digital computer, a long input signal sequence must be segmented to fixed-size blocks prior to processing.

2010/6/12

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86

Overlap-Add Method



Overlap-Add Method

- We first segment $x(n)$, assumed to be a causal sequence here without any loss of generality, into a set of contiguous finite-length subsequences of length L each:

$$x(n) = \sum_{m=0}^{\infty} x_m(n - mL)$$

where

$$x_m(n) = \begin{cases} x(n + mL), & 0 \leq n \leq L-1 \\ 0, & \text{otherwise} \end{cases}$$

- Thus we can write

$$y(n) = h(n) * x(n) = \sum_{m=0}^{\infty} y_m(n - mL)$$

where

$$y_m(n) = h(n) * x_m(n)$$

- Since $h(n)$ is of length M and $x_m(n)$ is of length L , $y_m(n)$ is of length $L + M - 1$

Overlap-Add Method

- The desired linear convolution $y(n) = h(n) * x(n)$ is broken up into a sum of infinite number of short-length linear convolutions of length $L + M - 1$ each: $y_m(n) = h(n) * x_m(n)$
 - Consider implementing the following convolutions using the DFT-based method, where now the DFTs (and the IDFT) are computed on the basis of $(L + M - 1)$ points

$$y(n) = \sum_{m=0}^{\infty} y_m(n - mL)$$

- The first convolution in the above sum, $y_0(n) = h(n) * x_0(n)$, is of length $L + M - 1$ and is defined for $0 \leq n \leq L + M - 2$
 - The second short convolution $y_1[n] = h(n) * x_1(n)$, is also of length $L + M - 1$ but is defined for $L \leq n \leq 3L + M - 2$

⇒ There is an overlap of samples between these two short linear convolutions

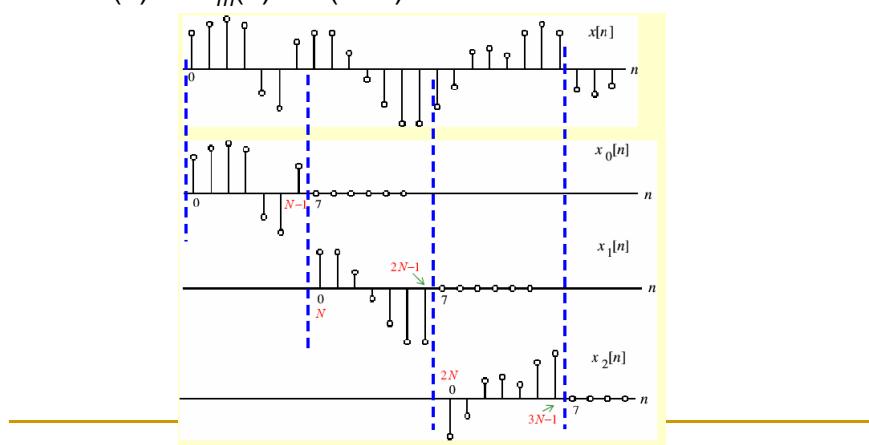
2010/6/12

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89

Overlap-Add Method

- In general, there will be an overlap of $M - 1$ samples between the samples of the short convolutions $h(n) * x_{r-1}(n)$ and $h(n) * x_m(n)$ for $(r-1)L \leq n \leq rL + M - 2$

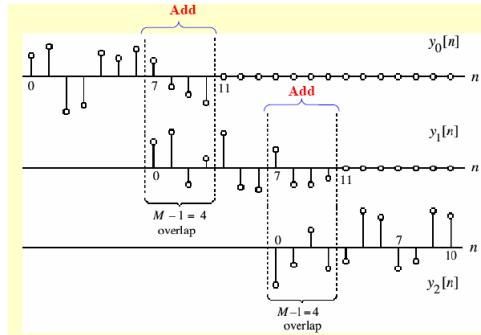


2010/6/12

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90

Overlap-Add Method



- Therefore, $y(n)$ obtained by a linear convolution of $x(n)$ and $h(n)$ is given by

$$y(n) = y_0(n), \quad 0 \leq n \leq 6$$

$$y(n) = y_1(n) + y_1(n-7), \quad 7 \leq n \leq 10$$

$$y(n) = y_1(n-7), \quad 11 \leq n \leq 13$$

⋮

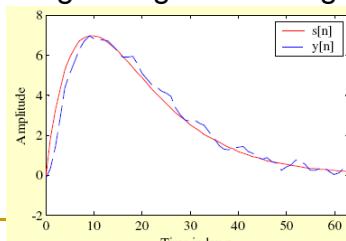
2010/6/12

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91

Overlap-Add Method

- The above procedure is called the **overlap-add method** since the results of the short linear convolutions overlap and the overlapped portions are added to get the correct final result
- The MATLAB function **fftfilt** can be used to implement the above method
- The following illustrates an example of filtering of a noise-corrupted signal using a length-3 moving average filter:

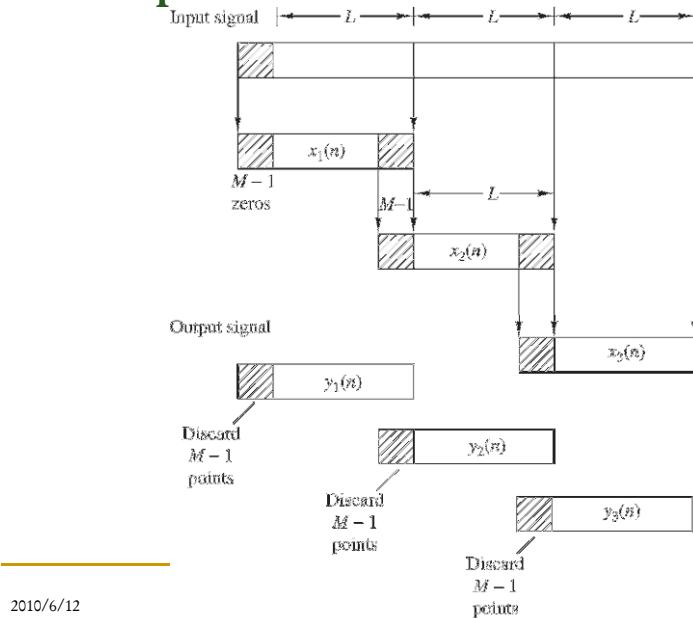


2010/6/12

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92

Overlap-Save Method



2010/6/12

93

Overlap-Save Method

- In implementing the overlap-add method using the DFT, we need to compute two $(L + M - 1)$ -point DFTs and one $(L + M - 1)$ -point IDFT for each short linear convolution
- It is possible to implement the overall linear convolution by performing instead circular convolution of length shorter than $(L + M - 1)$
- To this end, it is necessary **to segment $x(n)$ into overlapping blocks $x_m(n)$** , keep the terms of the circular convolution of $h(n)$ with that corresponds to the terms obtained by a linear convolution of $h(n)$ and $x_m(n)$, and throw away the other parts of the circular convolution

2010/6/12

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94

Overlap-Save Method

- To understand the correspondence between the linear and circular convolutions, consider a length-4 sequence $x(n)$ and a length-3 sequence $h(n)$
- Let $y_L(n)$ denote the result of a linear convolution of $x(n)$ with $h(n)$
- The six samples of $y_L(n)$ are given by

$$\begin{aligned}y_L(0) &= h(0)x(0) \\y_L(1) &= h(0)x(1) + h(1)x(0) \\y_L(2) &= h(0)x(2) + h(1)x(1) + h(2)x(0) \\y_L(3) &= h(0)x(3) + h(1)x(2) + h(2)x(1) \\y_L(4) &= \textcolor{blue}{h(1)x(3) + h(2)x(2)} \\y_L(5) &= \textcolor{blue}{h(2)x(3)}\end{aligned}$$

2010/6/12

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95

Overlap-Save Method

- If we append $h(n)$ with a single zero-valued sample and convert it into a length-4 sequence $h_e(n)$, the 4-point circular convolution $y_C(n)$ of $h_e(n)$ and $x(n)$ is given by
- $$\begin{aligned}y_C(0) &= h(0)x(0) + \textcolor{blue}{h(1)x(3) + h(2)x(2)} \\y_C(1) &= h(0)x(1) + h(1)x(0) + \textcolor{blue}{h(2)x(3)} \\y_C(2) &= h(0)x(2) + h(1)x(1) + h(2)x(0) \\y_C(3) &= h(0)x(3) + h(1)x(2) + h(2)x(1)\end{aligned}$$
- If we compare the expressions for the samples $y_L(n)$ with those of $y_C(n)$, we observe that the first 2 terms of $y_C(n)$ do not correspond to the first 2 terms of $y_L(n)$, whereas the last 2 terms of $y_C(n)$ are precisely the same as the 3rd and 4th terms of $y_L(n)$, i.e.

$$y_L(0) \neq y_C(0), y_L(1) \neq y_C(1), y_L(2) = y_C(2), y_L(3) = y_C(3)$$

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96

Overlap-Save Method

- **General case:** N -point circular convolution of a length- M sequence $h(n)$ with a length- L sequence $x(n)$ with $N > M$
- First $M - 1$ samples of the circular convolution are incorrect and are rejected
- Remaining $L - M + 1$ samples correspond to the correct samples of the linear convolution of $h(n)$ with $x(n)$
- Now, consider an infinitely long or very long sequence $x(n)$
- Break it up as a collection of smaller length (length-4) overlapping sequences $x_m(n)$ as $x_m(n) = x(n + 2m)$, $0 \leq n \leq 3$, $0 \leq m \leq \infty$
- Next, form

$$w_m(n) = h(n) \circledast x_m(n)$$

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97

Overlap-Save Method

- Or, equivalently,
 - $w_m(0) = h(0)x_m(0) + h(1)x_m(3) + h(2)x_m(2)$
 - $w_m(1) = h(0)x_m(1) + h(1)x_m(0) + h(2)x_m(3)$
 - $w_m(2) = h(0)x_m(2) + h(1)x_m(1) + h(2)x_m(0)$
 - $w_m(3) = h(0)x_m(3) + h(1)x_m(2) + h(2)x_m(1)$

- Computing the above for $m = 0, 1, 2, 3, \dots$, and substituting the values of $x_m[n]$ we arrive at

$$\begin{aligned} w_0(0) &= h(0)x(0) + h(1)x(3) + h(2)x(2) && \leftarrow \text{Reject} \\ w_0(1) &= h(0)x(1) + h(1)x(0) + h(2)x(3) && \leftarrow \text{Reject} \\ w_0(2) &= h(0)x(2) + h(1)x(1) + h(2)x(0) = y[2] && \leftarrow \text{Save} \\ w_0(3) &= h(0)x(3) + h(1)x(2) + h(2)x(1) = y[3] && \leftarrow \text{Save} \end{aligned}$$

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98

Overlap-Save Method

$$\begin{aligned}
 w_1(0) &= h(0)x(2) + h(1)x(5) + h(2)x(4) && \leftarrow \text{Reject} \\
 w_1(1) &= h(0)x(3) + h(1)x(2) + h(2)x(5) && \leftarrow \text{Reject} \\
 w_1(2) &= h(0)x(4) + h(1)x(3) + h(2)x(2) = y(4) && \leftarrow \text{Save} \\
 w_1(3) &= h(0)x(5) + h(1)x(4) + h(2)x(3) = y(5) && \leftarrow \text{Save}
 \end{aligned}$$

$$\begin{aligned}
 w_2(0) &= h(0)x(4) + h(1)x(7) + h(2)x(6) && \leftarrow \text{Reject} \\
 w_2(1) &= h(0)x(5) + h(1)x(4) + h(2)x(7) && \leftarrow \text{Reject} \\
 w_2(2) &= h(0)x(6) + h(1)x(5) + h(2)x(4) = y(6) && \leftarrow \text{Save} \\
 w_2(3) &= h(0)x(7) + h(1)x(6) + h(2)x(5) = y(7) && \leftarrow \text{Save}
 \end{aligned}$$

- It should be noted that to determine $y(0)$ and $y(1)$, we need to form $x_{-1}(n)$: $x_{-1}(0) = 0$, $x_{-1}(1) = 0$, $x_{-1}(2) = x(0)$, $x_{-1}(3) = x(1)$ and compute $w_{-1}(n) = h(n) \circledast x_{-1}(n)$ for $0 \leq n \leq 3$, reject $w_{-1}(0)$ and $w_{-1}(1)$, and save $w_{-1}(2) = y(0)$, and $w_{-1}(3) = y(1)$

Overlap-Save Method

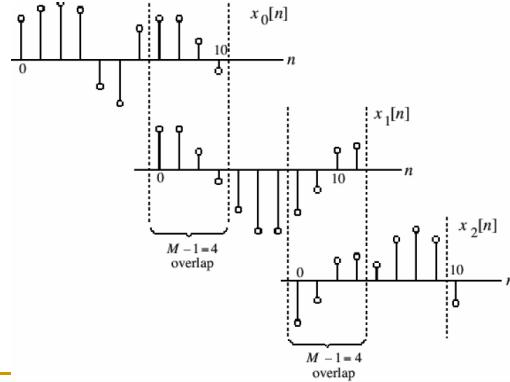
- General Case:** Let $h(n)$ be a length- L sequence
 - Let $x_m(n)$ denote the m -th section of an infinitely long sequence $x[n]$ of length L and defined by
- $$x_m(n) = x(n + m(L - m + 1)), \quad 0 \leq n \leq L - 1 \text{ with } M < L$$
- Let $w_m(n) = h(n) \circledast x_m(n)$
 - Then, we reject the first $M - 1$ samples of $w_m(n)$ and “**abut**” the remaining $L - M + 1$ samples of $w_m(n)$ to form $y_L(n)$, the linear convolution of $h(n)$ and $x(n)$
 - If $y_m[n]$ denotes the saved portion of $w_m(n)$, i.e.,

$$y(n) = \begin{cases} 0, & 0 \leq n \leq M - 2 \\ w_m(n), & M - 1 \leq n \leq L - 2 \end{cases}$$

- Then $y_L(n + m(L - M + 1)) = y_m(n), \quad M - 1 \leq n \leq L - 1$

Overlap-Save Method

- The approach is called **overlap-save method** since the input is segmented into overlapping sections and parts of the results of the circular convolutions are saved and abutted to determine the linear convolution result

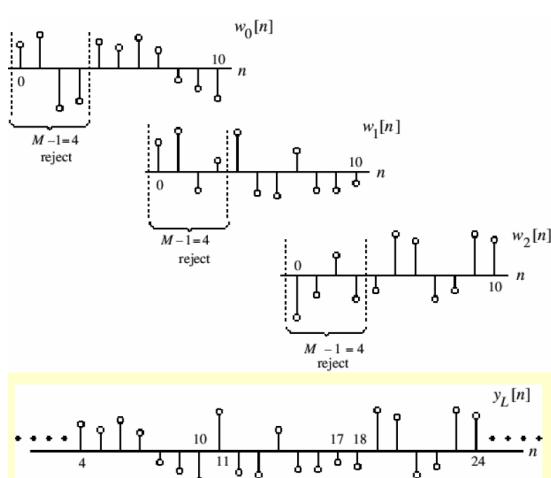


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101

Overlap-Save Method



2010/6/12

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102

Frequency Analysis of Signals Using the DFT

- Let $\{x(n)\}$ denote the sequence to be analyzed. Limiting the duration of the sequence to L samples, in the interval $0 \leq n \leq L-1$, is equivalent to multiplying $\{x(n)\}$ by a rectangular window $w(n)$ of length L . That is

$$\hat{x}(n) = x(n)w(n)$$

where

$$w(n) = \begin{cases} 1, & 0 \leq n \leq L-1 \\ 0, & \text{otherwise} \end{cases}$$

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103

Frequency Analysis of Signals Using the DFT

- Suppose that the sequence $x(n)$ consists of a signal sinusoid, that is

$$x(n) = \cos \omega_0 n$$

- Then the Fourier transform of the finite-duration sequence $x(n)$ can be expressed as

$$\hat{X}(\omega) = \frac{1}{2} [W(\omega - \omega_0) + W(\omega + \omega_0)]$$

where $W(\omega)$ is the Fourier transform of the window sequence, which is

$$W(\omega) = \frac{\sin(\omega L/2)}{\sin(\omega/2)} e^{-j\omega(L-1)/2}$$

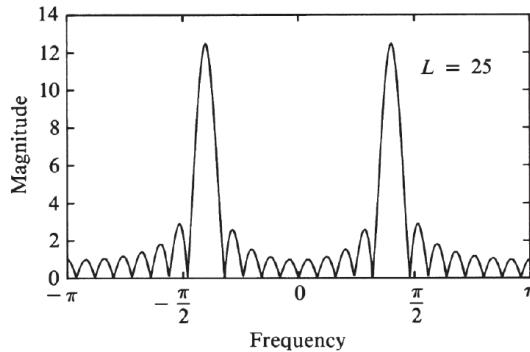
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104

Frequency Analysis of Signals Using the DFT

- Magnitude spectrum for $L=25$ and $N=2048$, illustrating the occurrence of leakage.



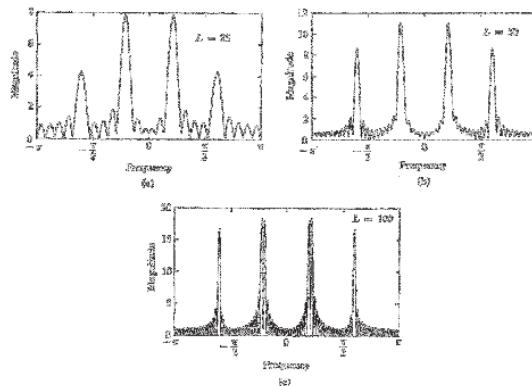
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105

Frequency Analysis of Signals Using the DFT

- $x(n) = \cos \omega_0 n + \cos \omega_1 n + \cos \omega_2 n$



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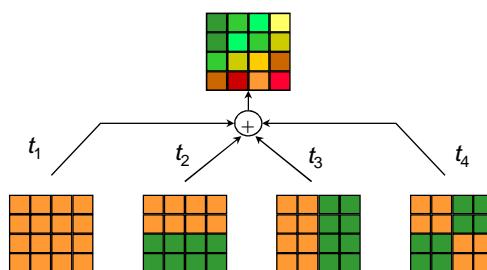
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106

Signal Transform

- Motivation:

- Represent a vector (e.g. a block of image samples) as the superposition of some typical vectors (block patterns)

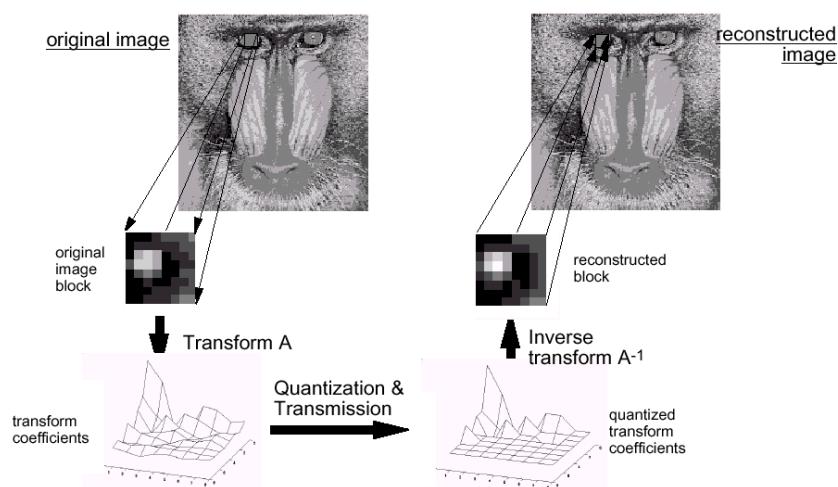


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107

Transform Coding of a Image

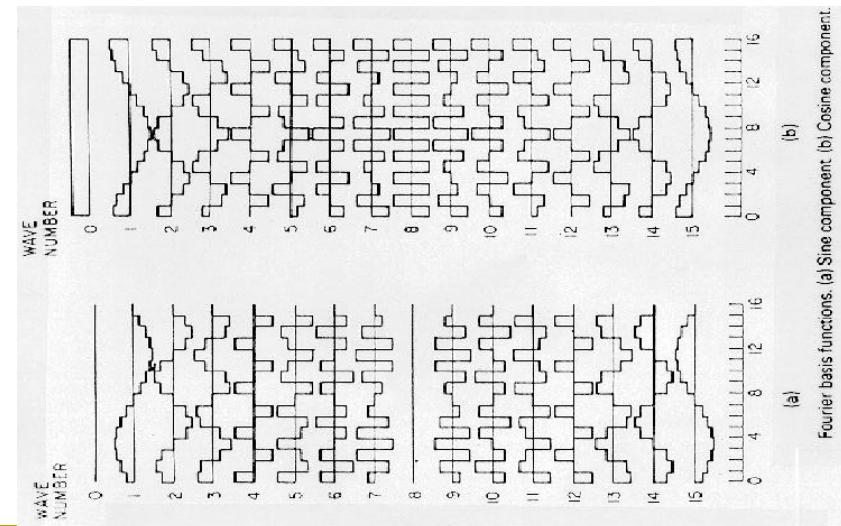


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108

1-D 16-Pont DFT Basis Vectors



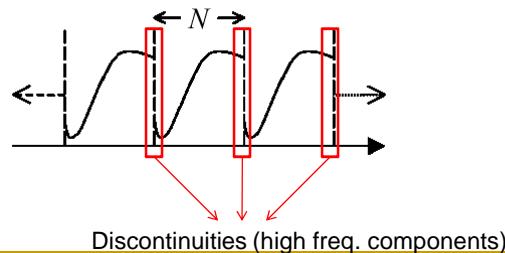
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109

Disadvantages of DFT in Signal Coding

- Fourier Transform of a real function results in complex numbers
- May result in artifacts due to discontinuity at the block boundary



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110

From DFT to DCT

- DFT of any real and symmetric sequence contains only real coefficients corresponding to the cosine terms of the series
- Construct a new symmetric sequence $y(n)$ of length $2N$ out of $x(n)$ of length N

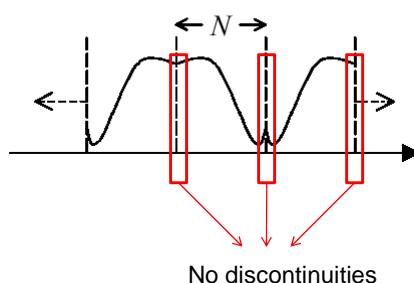
$$y(n) = x(n), 0 \leq n \leq N-1,$$

$$y(n) = x(2N-1-n), N \leq n \leq 2N-1.$$

- $Y(n)$ is symmetrical about $n = N - (1/2)$

From DFT to DCT

- DCT has a higher compression ration than DFT
 - DCT avoids the generation of spurious spectral components



From DFT to DCT

$$\begin{aligned}
 Y(k) &= \sum_{n=0}^{2N-1} y(n) W_{2N}^{(n+\frac{1}{2})k}, \\
 &= \sum_{n=0}^{N-1} y(n) W_{2N}^{(n+\frac{1}{2})k} + \sum_{n=N}^{2N-1} y(n) W_{2N}^{(n+\frac{1}{2})k} \\
 &= \sum_{n=0}^{N-1} x(n) W_{2N}^{(n+\frac{1}{2})k} + \sum_{n=N}^{2N-1} x(2N-1-n) W_{2N}^{(n+\frac{1}{2})k} \\
 &= \sum_{n=0}^{N-1} x(n) W_{2N}^{(n+\frac{1}{2})k} + \sum_{n=0}^{N-1} x(n) W_{2N}^{[2N-(n+\frac{1}{2})k]} \\
 &= \sum_{n=0}^{N-1} 2x(n) \cos \frac{\pi(2n+1)k}{2N}, \\
 0 \leq k \leq 2N-1, \text{ and } W_{2N} &= e^{-j\frac{2\pi}{2N}}
 \end{aligned}$$

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113

1-D N -Point DCT

$$F(k) = C(k) \sum_{n=0}^{N-1} f(n) \cos \left[\frac{(2n+1)k\pi}{2N} \right],$$

$$k = 0, 1, \dots, N-1,$$

$$f(n) = \sum_{k=0}^{N-1} C(k) F(k) \cos \left[\frac{(2n+1)k\pi}{2N} \right],$$

$$n = 0, 1, \dots, N-1,$$

where

$$C(0) = \sqrt{\frac{1}{N}}, \quad C(k) = \sqrt{\frac{2}{N}}, \quad k = 1, 2, \dots, N-1$$

- The constants are often defined differently

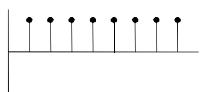
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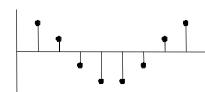
114

1-D N-Point DCT

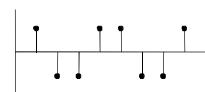
$$\cos((2n+1)0\pi/16)$$



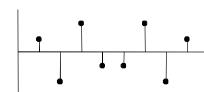
$$\cos((2n+1)2\pi/16)$$



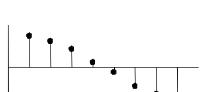
$$\cos((2n+1)4\pi/16)$$



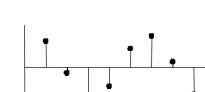
$$\cos((2n+1)6\pi/16)$$



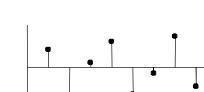
$$\cos((2n+1)1\pi/16)$$



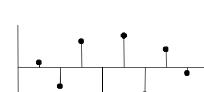
$$\cos((2n+1)3\pi/16)$$



$$\cos((2n+1)5\pi/16)$$



$$\cos((2n+1)7\pi/16)$$

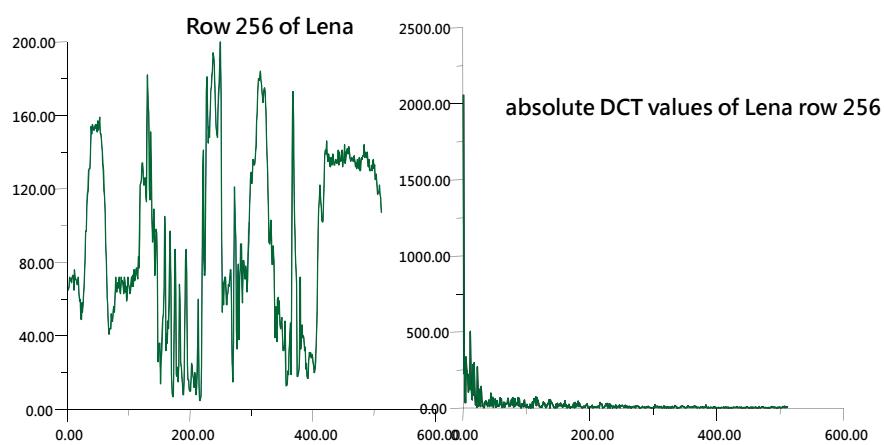


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115

Example of 1-D DCT

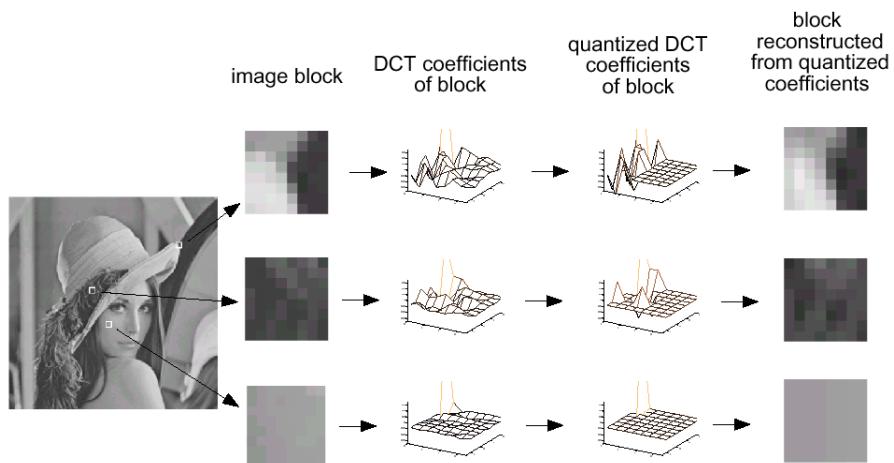


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116

Illustration of Image Coding Using 2-D DCT



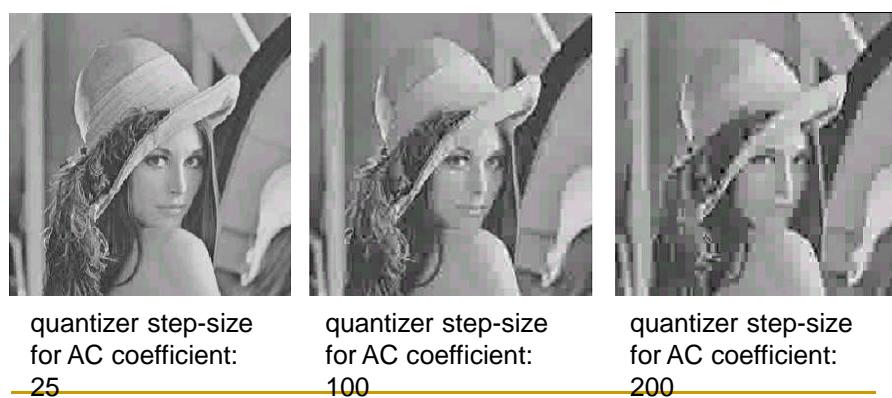
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117

DCT-Based Image Coding with Different Quantization Levels

- DCT coding with increasingly coarse quantization, block size 8x8



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118

Image Coding with Different Numbers of DCT Coefficients

original



with 16/64 coefficients

with 8/64 coefficients



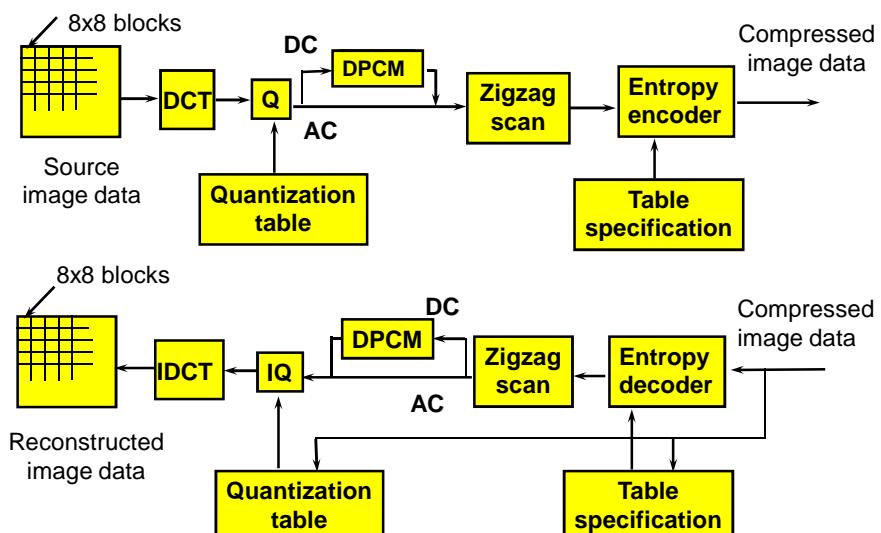
with 4/64 coefficients

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119

DCT-Based Coding: JPEG



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120