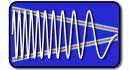


EE123



# **Digital Signal Processing**

Lecture 7 Block Convolution, Overlap and Add, FFT

based on slides by J.M. Kahn

M. Lustig, EECS UC Berkeley

### Last Time

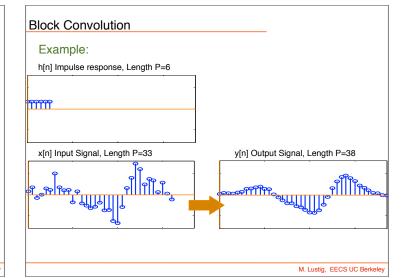
- Discrete Fourier Transform
  - Properties of the DFT
  - Linear convolution through circular
- Today
  - -Linear convolution with DFT
    - · Overlap and add
    - · Overlap and save
  - Fast Fourier Transform (start)

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### **Block Convolution**

- Problem:
  - An input signal x[n], has very long length (could be considered infinite)
  - An impulse response h[n] has length P
  - We want to take advantage of DFT/FFT and compute convolutions in blocks that are shorter than the signal
- Approach:
  - Break the signal into small blocks
  - Compute convolutions
  - Combine the results

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We can compute each output segment  $x_r[n] * h[n]$  with linear

• Zero-pad input segment  $x_r[n]$  to obtain  $x_{r,zp}[n]$ , of length M ullet Zero-pad the impulse response h[n] to obtain  $h_{zp}[n]$ , of length

 $x_r[n] * h[n] = \mathcal{DFT}^{-1} \{ \mathcal{DFT} \{ x_{r,zp}[n] \} \cdot \mathcal{DFT} \{ h_{zp}[n] \} \}$ 

Since output segment  $x_r[n] * h[n]$  starts offset from its neighbor  $x_{r-1}[n] * h[n]$  by L, neighboring output segments overlap at P-1

Finally, we just add up the output segments using (1) to obtain the

DFT-based circular convolution is usually more efficient:

N (this needs to be done only once). • Compute each output segment using:

### Overlap-Add Method

We decompose the input signal x[n] into non-overlapping segments  $x_r[n]$  of length L:

$$x_r[n] = \begin{cases} x[n] & rL \le n \le (r+1)L - 1 \\ 0 & \text{otherwise} \end{cases}$$

The input signal is the sum of these input segments:

$$x[n] = \sum_{r=0}^{\infty} x_r[n]$$

The output signal is the sum of the output segments  $x_r[n] * h[n]$ :

$$y[n] = x[n] * h[n] = \sum_{r=0}^{\infty} x_r[n] * h[n]$$
 (1)

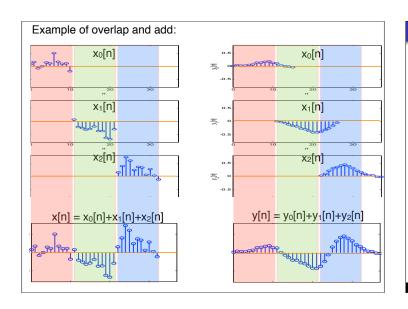
Each of the output segments  $x_r[n] * h[n]$  is of length M = L + P - 1.

Overlap-Add Method

convolution.

points.

output.



# Overlap-Save Method

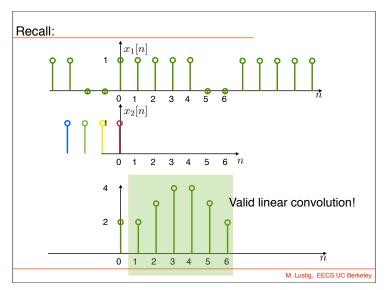
#### Basic Idea

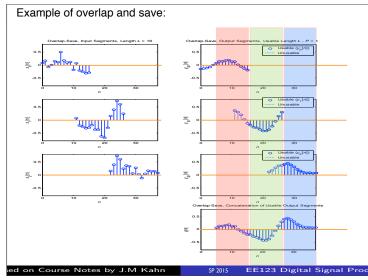
We split the input signal x[n] into overlapping segments  $x_r[n]$  of length L + P - 1.

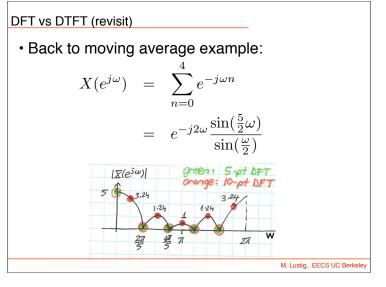
Perform a circular convolution of each input segment  $x_r[n]$  with the impulse response h[n], which is of length P using the DFT. Identify the L-sample portion of each circular convolution that corresponds to a linear convolution, and save it.

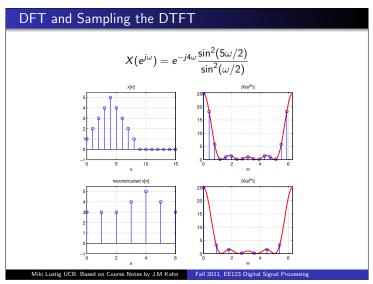
This is illustrated below where we have a block of L samples circularly convolved with a P sample filter.

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# Circular Convolution as Matrix Operation

Circular convolution:

$$h[n] @x[n] = \begin{bmatrix} h[0] & h[N-1] & \cdots & h[1] \\ h[1] & h[0] & & h[2] \\ & & & \vdots \\ h[N-1] & h[N-2] & & h[0] \end{bmatrix} \begin{bmatrix} x[0] \\ x[1] \\ \vdots \\ x[N-1] \end{bmatrix}$$

$$= H_c x$$

- $H_c$  is a circulant matrix
- The columns of the DFT matrix are Eigen vectors of circulant
- Eigen vectors are DFT coefficients. How can you show?

Proof in HW

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# Circular Convolution as Matrix Operation

Diagonalize:

$$W_N H_c W_n^{-1} = \begin{bmatrix} H[0] & 0 \cdots & 0 \\ 0 & H[1] \cdots & 0 \\ \vdots & 0 & H[N-1] \end{bmatrix}$$

• Right-multiply by  $W_N$ 

$$W_N H_c = \begin{bmatrix} H[0] & 0 \cdots & 0 \\ 0 & H[1] \cdots & 0 \\ \vdots & 0 & H[N-1] \end{bmatrix} W_N$$

Multiply both sides by x

$$W_N H_{cX} = \begin{bmatrix} H[0] & 0 \cdots & 0 \\ 0 & H[1] \cdots & 0 \\ \vdots & 0 & H[N-1] \end{bmatrix} W_{NX}$$

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### Fast Fourier Transform Algorithms

• We are interested in efficient computing methods for the DFT and inverse DFT:

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

where

$$W_N = e^{-j\left(\frac{2\pi}{N}\right)}$$
.

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• Recall that we can use the DFT to compute the inverse DFT:

$$\mathcal{DFT}^{-1}\{X[k]\} = \frac{1}{N} (\mathcal{DFT}\{X^*[k]\})^*$$

Hence, we can just focus on efficient computation of the DFT.

• Straightforward computation of an N-point DFT (or inverse DFT) requires  $N^2$  complex multiplications.

• Fast Fourier transform algorithms enable computation of an N-point DFT (or inverse DFT) with the order of just  $N \cdot \log_2 N$  complex multiplications.

This can represent a huge reduction in computational load, especially for large N.

N	$N^2$	$N \cdot \log_2 N$	$\frac{N^2}{N \cdot \log_2 N}$
16	256	64	4.0
128	16,384	896	18.3
1,024	1,048,576	10,240	102.4
8,192	67,108,864	106,496	630.2
$6 \times 10^6$	$36 \times 10^{12}$	$135 \times 10^{6}$	$2.67 \times 10^{5}$

\* 6Mp image size

- Most FFT algorithms exploit the following properties of  $W_N^{kn}$ :
  - Conjugate Symmetry

$$W_N^{k(N-n)} = W_N^{-kn} = (W_N^{kn})^*$$

Periodicity in n and k:

$$W_N^{kn} = W_N^{k(n+N)} = W_N^{(k+N)n}$$

Power:

$$W_N^2 = W_{N/2}$$

- Most FFT algorithms decompose the computation of a DFT into successively smaller DFT computations.
  - Decimation-in-time algorithms decompose x[n] into successively smaller subsequences.
  - Decimation-in-frequency algorithms decompose X[k] into successively smaller subsequences.
- We mostly discuss decimation-in-time algorithms here.

Assume length of x[n] is power of 2 (  $N=2^{\nu}$  ). If smaller zero-pad to closest power.

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# Decimation-in-Time Fast Fourier Transform

We start with the DFT

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

• Separate the sum into even and odd terms:

$$X[k] = \sum_{n \text{ even}} x[n]W_N^{kn} + \sum_{n \text{ odd}} x[n]W_N^{kn}$$

These are two DFT's, each with half of the samples.

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### Decimation-in-Time Fast Fourier Transform

Let n = 2r (n even) and n = 2r + 1 (n odd):

$$X[k] = \sum_{r=0}^{(N/2)-1} x[2r] W_N^{2rk} + \sum_{r=0}^{(N/2)-1} x[2r+1] W_N^{(2r+1)k}$$
$$= \sum_{r=0}^{(N/2)-1} x[2r] W_N^{2rk} + W_N^k \sum_{r=0}^{(N/2)-1} x[2r+1] W_N^{2rk}$$

• Note that:

$$W_N^{2rk} = e^{-j\left(\frac{2\pi}{N}\right)(2rk)} = e^{-j\left(\frac{2\pi}{N/2}\right)rk} = W_{N/2}^{rk}$$

Remember this trick, it will turn up often.

### Decimation-in-Time Fast Fourier Transform

• Hence:

$$X[k] = \sum_{r=0}^{(N/2)-1} x[2r] W_{N/2}^{rk} + W_N^k \sum_{r=0}^{(N/2)-1} x[2r+1] W_{N/2}^{rk}$$

$$\triangleq G[k] + W_N^k H[k], \quad k = 0, \dots, N-1$$

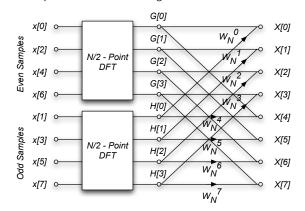
where we have defined:

$$\begin{array}{ll} G[k] & \stackrel{\triangle}{=} & \displaystyle\sum_{r=0}^{(N/2)-1} x[2r] W_{N/2}^{rk} & \Rightarrow \text{DFT of even idx} \\ \\ H[k] & \stackrel{\triangle}{=} & \displaystyle\sum_{r=0}^{(N/2)-1} x[2r+1] W_{N/2}^{rk} & \Rightarrow \text{DFT of odd idx} \end{array}$$

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# Decimation-in-Time Fast Fourier Transform

An 8 sample DFT can then be diagrammed as



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### Decimation-in-Time Fast Fourier Transform

• Both G[k] and H[k] are periodic, with period N/2. For example

$$G[k+N/2] = \sum_{r=0}^{(N/2)-1} x[2r] W_{N/2}^{r(k+N/2)}$$

$$= \sum_{r=0}^{(N/2)-1} x[2r] W_{N/2}^{rk} W_{N/2}^{r(N/2)}$$

$$= \sum_{r=0}^{(N/2)-1} x[2r] W_{N/2}^{rk}$$

$$= G[k]$$

so

$$G[k + (N/2)] = G[k]$$
  
 $H[k + (N/2)] = H[k]$ 

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### Decimation-in-Time Fast Fourier Transform

- The periodicity of G[k] and H[k] allows us to further simplify.
- For the first N/2 points we calculate G[k] and  $W_N^k H[k]$ , and then compute the sum

$$X[k] = G[k] + W_N^k H[k] \qquad \forall \{k: 0 \le k < \frac{N}{2}\}.$$

How does periodicity help for  $\frac{N}{2} \le k < N$ ?

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# Decimation-in-Time Fast Fourier Transform

$$X[k] = G[k] + W_N^k H[k] \qquad \forall \{k : 0 \le k < \frac{N}{2}\}.$$

• for  $\frac{N}{2} \le k < N$ :

$$W_N^{k+(N/2)} = ?$$

$$X[k + (N/2)] = ?$$

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## Decimation-in-Time Fast Fourier Transform

$$X[k + (N/2)] = G[k] - W_N^k H[k]$$

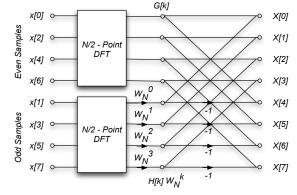
We previously calculated G[k] and  $W_N^k H[k]$ .

Now we only have to compute their difference to obtain the second half of the spectrum. No additional multiplies are required.

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# Decimation-in-Time Fast Fourier Transform

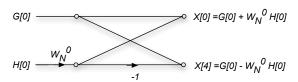
ullet The N-point DFT has been reduced two N/2-point DFTs, plus N/2 complex multiplications. The 8 sample DFT is then:



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# Decimation-in-Time Fast Fourier Transform

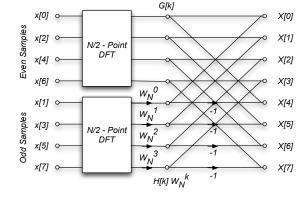
- Note that the inputs have been reordered so that the outputs come out in their proper sequence.
- We can define a butterfly operation, e.g., the computation of X[0] and X[4] from G[0] and H[0]:



This is an important operation in DSP.

# Decimation-in-Time Fast Fourier Transform

• Still  $O(N^2)$  operations.... What shall we do?

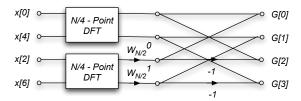


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### Decimation-in-Time Fast Fourier Transform

• We can use the same approach for each of the N/2 point DFT's. For the N=8 case, the N/2 DFTs look like



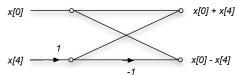
\*Note that the inputs have been reordered again.

# Decimation-in-Time Fast Fourier Transform

• At this point for the 8 sample DFT, we can replace the N/4 = 2 sample DFT's with a single butterfly. The coefficient is

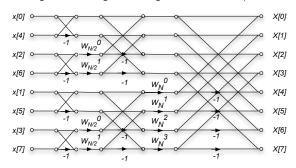
$$W_{N/4} = W_{8/4} = W_2 = e^{-j\pi} = -1$$

The diagram of this stage is then



### Decimation-in-Time Fast Fourier Transform

Combining all these stages, the diagram for the 8 sample DFT is:



This the decimation-in-time FFT algorithm.

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### Decimation-in-Time Fast Fourier Transform

- In general, there are  $\log_2 N$  stages of decimation-in-time.
- $\bullet$  Each stage requires N/2 complex multiplications, some of which are trivial.
- The total number of complex multiplications is  $(N/2)\log_2 N$ .
- The order of the input to the decimation-in-time FFT algorithm must be permuted.
  - First stage: split into odd and even. Zero low-order bit first
  - Next stage repeats with next zero-lower bit first.
  - Net effect is reversing the bit order of indexes

### Decimation-in-Time Fast Fourier Transform

This is illustrated in the following table for N=8.

Decimal	Binary	Bit-Reversed Binary	Bit-Reversed Decimal
0	000	000	0
1	001	100	4
2	010	010	2
3	011	110	6
4	100	001	1
5	101	101	5
6	110	011	3
7	111	111	7

# Decimation-in-Frequency Fast Fourier Transform

The DFT is

$$X[k] = \sum_{n=0}^{N-1} x[n] W_N^{nk}$$

If we only look at the even samples of X[k], we can write k = 2r,

$$X[2r] = \sum_{n=0}^{N-1} x[n] W_N^{n(2r)}$$

We split this into two sums, one over the first N/2 samples, and the second of the last N/2 samples.

$$X[2r] = \sum_{n=0}^{(N/2)-1} x[n]W_N^{2rn} + \sum_{n=0}^{(N/2)-1} x[n+N/2]W_N^{2r(n+N/2)}$$

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### Decimation-in-Frequency Fast Fourier Transform

But  $W_N^{2r(n+N/2)} = W_N^{2rn} W_N^N = W_N^{2rn} = W_{N/2}^{rn}$ . We can then write

$$X[2r] = \sum_{n=0}^{(N/2)-1} x[n] W_N^{2m} + \sum_{n=0}^{(N/2)-1} x[n+N/2] W_N^{2r(n+N/2)}$$

$$= \sum_{n=0}^{(N/2)-1} x[n] W_N^{2m} + \sum_{n=0}^{(N/2)-1} x[n+N/2] W_N^{2m}$$

$$= \sum_{n=0}^{(N/2)-1} (x[n] + x[n+N/2]) W_{N/2}^{m}$$

This is the N/2-length DFT of first and second half of x[n]summed.

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# Decimation-in-Frequency Fast Fourier Transform

$$\begin{array}{rcl} X[2r] & = & \mathsf{DFT}_{\frac{N}{2}} \left\{ \left( x[n] + x[n+N/2] \right) \right\} \\ X[2r+1] & = & \mathsf{DFT}_{\frac{N}{2}} \left\{ \left( x[n] - x[n+N/2] \right) W_N^n \right\} \end{array}$$

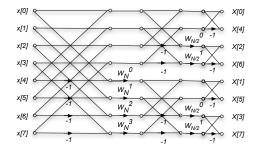
(By a similar argument that gives the odd samples)

Continue the same approach is applied for the N/2 DFTs, and the N/4 DFT's until we reach simple butterflies.

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### Decimation-in-Frequency Fast Fourier Transform

The diagram for and 8-point decimation-in-frequency DFT is as follows



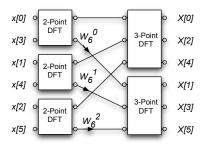
This is just the decimation-in-time algorithm reversed! The inputs are in normal order, and the outputs are bit reversed.

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# Non-Power-of-2 FFT's

A similar argument applies for any length DFT, where the length N is a composite number.

For example, if N = 6, a decimation-in-time FFT could compute three 2-point DFT's followed by two 3-point DFT's



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### Non-Power-of-2 FFT's

Good component DFT's are available for lengths up to 20 or so. Many of these exploit the structure for that specific length. For example, a factor of

$$W_N^{N/4} = e^{-j\frac{2\pi}{N}(N/4)} = e^{-j\frac{\pi}{2}} = -j$$
 Why?

just swaps the real and imaginary components of a complex number, and doesn't actually require any multiplies.

Hence a DFT of length 4 doesn't require any complex multiplies. Half of the multiplies of an 8-point DFT also don't require

Composite length FFT's can be very efficient for any length that factors into terms of this order.

For example N = 693 factors into

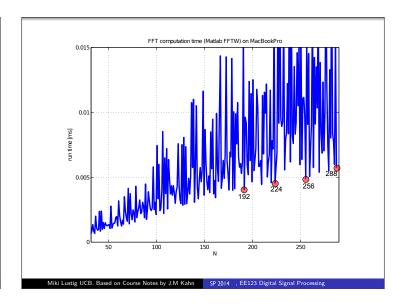
$$N = (7)(9)(11)$$

each of which can be implemented efficiently. We would perform

- $\bullet$  9  $\times$  11 DFT's of length 7
- $\bullet$  7  $\times$  11 DFT's of length 9, and
- 7 × 9 DFT's of length 11

- Historically, the power-of-two FFTs were much faster (better written and implemented).
- For non-power-of-two length, it was faster to zero pad to power of two.
- Recently this has changed. The free FFTW package implements very efficient algorithms for almost any filter length. Matlab has used FFTW since version 6

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# FFT as Matrix Operation

$$\begin{pmatrix} X[0] \\ \vdots \\ X[k] \\ \vdots \\ X[N-1] \end{pmatrix} = \begin{pmatrix} W_N^{00} & \cdots & W_N^{0n} & \cdots & W_N^{0(N-1)} \\ \vdots & \ddots & \vdots & \ddots & \vdots \\ W_N^{k0} & \cdots & W_N^{kn} & \cdots & W_N^{k(N-1)} \\ \vdots & \ddots & \vdots & \ddots & \vdots \\ W_N^{(N-1)0} & \cdots & W_N^{(N-1)n} & \cdots & W_N^{(N-1)(N-1)} \end{pmatrix} \begin{pmatrix} x[0] \\ \vdots \\ x[n] \\ \vdots \\ x[N-1] \end{pmatrix}$$

•  $W_N$  is fully populated  $\Rightarrow N^2$  entries.

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# FFT as Matrix Operation

$$\begin{pmatrix} x_{[0]} \\ \vdots \\ x_{[k]} \\ \vdots \\ x_{[N-1]} \end{pmatrix} = \begin{pmatrix} w_N^{00} & \cdots & w_N^{0n} & \cdots & w_N^{0(N-1)} \\ \vdots & \ddots & \vdots & \ddots & \vdots \\ w_N^{k0} & \cdots & w_N^{kn} & \cdots & w_N^{k(N-1)} \\ \vdots & \ddots & \vdots & \ddots & \vdots \\ w_N^{(N-1)0} & \cdots & w_N^{(N-1)n} & \cdots & w_N^{(N-1)(N-1)} \end{pmatrix} \begin{pmatrix} x_{[0]} \\ \vdots \\ x_{[n]} \\ \vdots \\ x_{[N-1]} \end{pmatrix}$$

- $W_N$  is fully populated  $\Rightarrow N^2$  entries.
- ullet FFT is a decomposition of  $W_N$  into a more sparse form:

$$F_N = \left[ egin{array}{cc} I_{N/2} & D_{N/2} \ I_{N/2} & -D_{N/2} \ \end{array} 
ight] \left[ egin{array}{cc} W_{N/2} & 0 \ 0 & W_{N/2} \ \end{array} 
ight] \left[ egin{array}{cc} {
m Even-Odd \ Perm.} \ {
m Matrix} \ \end{array} 
ight]$$

 $\bullet$   $I_{N/2}$  is an identity matrix.  $D_{N/2}$  is a diagonal with entries 1,  $W_N$ ,  $\cdots$ ,  $W_N^{N/2-1}$ 

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## FFT as Matrix Operation

Example: N = 4

$$F_4 = \left[ \begin{array}{cccc} 1 & 0 & 1 & 0 \\ 0 & 1 & 0 & W_4 \\ 1 & 0 & -1 & 0 \\ 0 & 1 & 0 & -W_4 \end{array} \right] \left[ \begin{array}{cccc} 1 & 1 & 0 & 0 \\ 1 & -1 & 0 & 0 \\ 0 & 0 & 1 & 1 \\ 0 & 0 & 1 & -1 \end{array} \right] \left[ \begin{array}{cccc} 1 & 0 & 0 & 0 \\ 0 & 0 & 1 & 0 \\ 0 & 1 & 0 & 0 \\ 0 & 0 & 0 & 1 \end{array} \right]$$

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## Beyond NlogN

- What if the signal x[n] has a k sparse frequency
  - A. Gilbert et. al, "Near-optimal sparse Fourier representations via sampling
  - H. Hassanieh et. al, "Nearly Optimal Sparse Fourier Transform"
  - Others.....

O(K Log N) instead of O(N Log N)

