- A digital filter structure can be described in the time-domain by a set of equations relating the output sequence to the input sequence and, in some cases, one or more internally generated sequences
- Consider



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• This structure, in the time-domain, is described by the set of equations

 $w_{1}[n] = x[n] - \alpha w_{5}[n]$ $w_{2}[n] = w_{1}[n] - \delta w_{3}[n]$ $w_{3}[n] = w_{2}[n-1]$ $w_{4}[n] = w_{3}[n] + \varepsilon w_{2}[n]$ $w_{5}[n] = w_{4}[n-1]$ $y[n] = \beta w_{1}[n] + \gamma w_{5}[n]$

- The equations cannot be implemented in the order shown with each variable on the left side computed before the variable below is computed
- For example, computation of $w_1[n]$ in the 1st step requires the knowledge of $w_5[n]$ which is computed in the 5th step
- Likewise, computation of $w_2[n]$ in the 2nd step requires the knowledge of $w_3[n]$ that is computed in the 3rd step

- This ordered set of equations is said to be **noncomputable**
- Suppose we reorder these equations $w_{3}[n] = w_{2}[n-1]$ $w_{5}[n] = w_{4}[n-1]$ $w_{1}[n] = x[n] - \alpha w_{5}[n]$ $w_{2}[n] = w_{1}[n] - \delta w_{3}[n]$ $y[n] = \beta w_{1}[n] + \gamma w_{5}[n]$ $w_{4}[n] = w_{3}[n] + \varepsilon w_{2}[n]$

- This ordered set of equations is **computable**
- In most practical applications, equations describing a digital filter structure can be put into a computable order by inspection
- A simple way to examine the computability of equations describing a digital filter structure is by writing the equations in a matrix form

• A matrix representation of the first ordered set of equations is



• In compact form

$$\mathbf{y}[n] = \mathbf{x}[n] + \mathbf{F} \mathbf{y}[n] + \mathbf{G} \mathbf{y}[n-1]$$

where

 $\mathbf{y}[n] = \begin{bmatrix} w_1[n] & w_2[n] & w_3[n] & w_4[n] & w_5[n] & y[n] \end{bmatrix}^T$ $\mathbf{x}[n] = \begin{bmatrix} x[n] & 0 & 0 & 0 & 0 \end{bmatrix}^T$



- For the computation of present value of a particular signal variable, nonzero entries in the corresponding rows of matrices F and G determine the variables whose present and previous values are needed
- If a diagonal element of **F** is nonzero, then computation of present value of the corresponding variable requires the knowledge of its present value implying presence of a delay-free loop

- Any nonzero entries in the same row above the main diagonal of F imply that the computation of present value of the corresponding variable requires present values of other variables not yet computed, making the set of equations noncomputable
- Hence, for computability all elements of **F** matrix on the diagonal and above diagonal must be zeros

- In the **F** matrix for the first ordered set of equations, diagonal elements are all zeros, indicating absence of delay-free loops
- However, there are nonzero entries above the diagonal in the first and second rows of
 F indicating that the set of equations are not in proper order for computation

• The **F** matrix for the second ordered set of equations is

which is seen to satisfy the computability condition

- The **precedence graph** can be used to test the computability of a digital filter structure and to develop the proper ordering sequence for a set of equations describing a computable structure
- It is developed from the **signal-flow graph** description of the digital filter structure in which independent and dependent signal variables are represented by **nodes**, and the multiplier and delay branches are represented by **directed branches**

- The directed branch has an attached symbol denoting the **branch gain** or **transmittance**
- For a multiplier branch, the branch gain is the multiplier coefficient value
- For a delay branch, the branch gain is simply z^{-1}

• The signal-flow graph representation of



is shown below



- A reduced signal-flow graph is then developed by removing the delay branches and all branches going out of the input node
- The reduced signal-flow graph of the example digital filter structure is shown below



- The remaining nodes in the reduced signalflow graph are grouped as follows:
- All nodes with only outgoing branches are grouped into one set labeled $\{\mathcal{N}_1\}$
- Next, the set {N₂} is formed containing nodes coming in only from one or more nodes in the set {N₁} and have outgoing branches to the other nodes

- Then, form the set {N₃} containing nodes that have branches coming in only from one or more nodes in the sets {N₁} and {N₂}, and have outgoing branches to other nodes
- Continue the process until there is a set of nodes $\{\mathcal{N}_f\}$ containing only incoming branches
- The rearranged signal-flow graph is called a **precedence graph**

- Since signal variables belonging to $\{\mathcal{N}_1\}$ do not depend on the present values of other signal variables, these variables should be computed first
- Next, signal variables belonging to {N₂} can be computed since they depend on the present values of signal variables contained in {N₁} that have already been computed

- This is followed by the computation of signal variables in $\{\mathcal{N}_3\}, \{\mathcal{N}_4\}$, etc.
- Finally, in the last step the signal variables in $\{\mathcal{N}_f\}$ are computed
- This process of sequential computation ensures the development of a valid computational algorithm
- If there is no final set $\{\mathcal{N}_f\}$ containing only incoming branches, the digital filter structure is noncomputable



• For the example precedence graph, pertinent groupings of node variables are: $\{N_1\} = \{w_3[n], w_5[n]\}$ $\{\mathcal{N}_2\} = \{w_1[n]\}$ $\{\mathcal{N}_3\} = \{w_2[n]\}$ $\{\mathcal{N}_4\} = \{w_4[n], y[n]\}$

• Precedence graph redrawn according to the above groupings is as shown below



• Since the final node set $\{\mathcal{N}_4\}$ has only incoming branches, the structure is computable

- A simple method to verify that the structure developed is indeed characterized by the prescribed transfer function *H*(*z*)
- Consider for simplicity a causal 3rd order IIR transfer function

$$H(z) = \frac{P(z)}{D(z)} = \frac{p_0 + p_1 z^{-1} + p_2 z^{-2} + p_3 z^{-3}}{1 + d_1 z^{-1} + d_2 z^{-2} + d_3 z^{-3}}$$

• If $\{h[n]\}$ denotes its impulse response, then

$$H(z) = \sum_{n=0}^{\infty} h[n] z^{-n}$$

- Note P(z) = H(z)D(z)which is equivalent to $p_n = \sum_{k=1}^{n} h[k]d_{n-k}, d_0 = 1$
- Evaluate above convolution sum for $0 \le n \le 6$: $p_0 = h[0]$ $p_1 = h[1] + h[0]d_1$ $p_2 = h[2] + h[1]d_1 + h[0]d_2$ $p_3 = h[3] + h[2]d_1 + h[1]d_2 + h[0]d_3$ $0 = h[4] + h[3]d_1 + h[2]d_2 + h[1]d_3$ $0 = h[5] + h[4]d_1 + h[3]d_2 + h[2]d_3$ $0 = h[6] + h[5]d_1 + h[4]d_2 + h[3]d_3$

• In matrix form we get



• In partitioned form above matrix equation can be written as

$$\begin{bmatrix} \mathbf{p} \\ \cdots \\ \mathbf{0} \end{bmatrix} = \begin{bmatrix} \mathbf{H}_1 \\ \cdots \\ \mathbf{h} \end{bmatrix} \begin{bmatrix} 1 \\ \cdots \\ \mathbf{H}_2 \end{bmatrix} \begin{bmatrix} 1 \\ \cdots \\ \mathbf{d} \end{bmatrix}$$

Solution set to be a set of the set of the

• Solving second equation we get

$$\mathbf{d} = -\mathbf{H}_2^{-1}\mathbf{h}$$

- Substituting above in the first equation we get $\mathbf{p} = \mathbf{H}_1 \begin{bmatrix} 1 \\ -\mathbf{H}_2^{-1} \mathbf{h} \end{bmatrix}$
- In the case of an *N*-th order IIR filter, the coefficients of its transfer function can be determined from the first 2*N*+1 impulse response samples

• <u>Example</u> - Consider the causal transfer function

$$H(z) = \frac{2+6z^{-1}+3z^{-2}}{1+z^{-1}+2z^{-2}} = 2+4z^{-1}-5z^{-2}-3z^{-3}+13z^{-4}+\cdots$$

Here

$$h[0] = 2, h[1] = 4, h[2] = -5, h[3] = -3, h[4] = 13$$

• Hence

$$\begin{bmatrix} p_0 \\ p_1 \\ p_2 \\ 0 \\ 0 \end{bmatrix} = \begin{bmatrix} 2 & 0 & 0 \\ 4 & 2 & 0 \\ -5 & 4 & 2 \\ -3 & -5 & 4 \\ 13 & -3 & -5 \end{bmatrix} \begin{bmatrix} 1 \\ d_1 \\ d_2 \end{bmatrix}$$

• Solving we get

$$\begin{bmatrix} d_1 \\ d_2 \end{bmatrix} = \begin{bmatrix} -5 & 4 \\ -3 & -5 \end{bmatrix}^{-1} \begin{bmatrix} -3 \\ 13 \end{bmatrix} = \begin{bmatrix} 1 \\ 2 \end{bmatrix}$$

and

$$\begin{bmatrix} p_0 \\ p_1 \\ p_2 \end{bmatrix} = \begin{bmatrix} 2 & 0 & 0 \\ 4 & 2 & 0 \\ -5 & 4 & 2 \end{bmatrix} \begin{bmatrix} 1 \\ 1 \\ 2 \end{bmatrix} = \begin{bmatrix} 2 \\ 6 \\ 3 \end{bmatrix}$$

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Structure Simulation and Verification Using MATLAB

- For computer simulation, the structure is described in the form of a set of equations
- These equations must be ordered properly to ensure computability
- The procedure is to express the output of each adder and filter output variable in terms of all incoming signal variables

Structure Simulation and Verification Using MATLAB

• Consider the structure



• A valid computational algorithm involving the least number of equations is $w_1[n] = x[n] - \alpha w_4[n-1],$ $w_2[n] = w_1[n] - \delta w_2[n-1],$ $w_4[n] = w_2[n-1] + \varepsilon w_2[n],$ $y[n] = \beta w_1[n] + \gamma w_4[n-1]$

Structure Simulation and Verification Using MATLAB

- This set of equations is evaluated for increasing values of *n* starting at *n* = 0
- At the beginning, the initial conditions w₂[-1] and w₄[-1] can be set to any desired values, which are typically zero
- From the computed impulse response samples, the structure can be verified by determining the transfer function coefficients using the M-file strucver

Simulation of IIR Filters

• The M-file filter implements the IIR filter in the transposed direct form II structure shown below for a 3rd order filter



• As indicated in the figure, d(1) has been assumed to be equal to 1

Simulation of IIR Filters

- Basic forms of this function are
 - y = filter(num,den,x)

[y,sf]=filter(num,den,x,si)

where x is the input vector, y is the output vector, si is the vector of initial conditions of the delay variables, and sf is the vector of final values of the delay variables

• For the simulation of a causal IIR filter realized in direct form II structure use the M-file direct2

Simulation of IIR Filters

- For the simulation of overlap-add filtering method use the M-file fftfilt or the second form of the M-file filter
- For the simulation of tapped cascaded lattice filter structures, use the M-file latcfilt
- The M-files filter, direct2 and latcfilt can also be used to simulate FIR filters
- The M-file filtfilt implements the zero-phase filtering

Discrete Fourier Transform Computation

• The *N*-point DFT X[k] of a length-*N* sequence $x[n], 0 \le n \le N-1$, is defined by $X[k] = \sum_{n=0}^{N-1} x[n] W_N^{kn}, 0 \le k \le N-1$

where

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

 Direct computation of all N samples of {X[k]} requires N²complex multiplications and N(N-1)complex additions

Goertzel's Algorithm

• A recursive DFT computation scheme that makes use of the identity $W_N^{-kN} = 1$

obtained using the periodicity of W_N^{-kn}

• Using this identity we can write

$$\begin{aligned} X[k] &= \sum_{\ell=0}^{N-1} x[\ell] W_N^{k\ell} \\ &= W_N^{-kN} \sum_{\ell=0}^{N-1} x[\ell] W_N^{k\ell} = \sum_{\ell=0}^{N-1} x[\ell] W_N^{-k(N-\ell)} \end{aligned}$$

Goertzel's Algorithm

- Define $y_k[n] = \sum_{\ell=0}^n x_e[\ell] W_N^{-k(n-\ell)}$
- <u>Note</u>: *y_k*[*n*] is the direct convolution of the causal sequence

$$x_e[n] = \begin{cases} x[n], & 0 \le n \le N - 1 \\ 0, & n < 0, n \ge N \end{cases}$$

with a causal sequence

$$h_k[n] = \begin{cases} W_N^{-kn}, & n \ge 0\\ 0, & n < 0 \end{cases}$$

• Observe $X[k] = y_k[n]_{n=N}$
Goertzel's Algorithm • *z*-transform of $y_k[n] = \sum_{\ell=0}^n x_e[\ell] W_N^{-k(n-\ell)}$ yields

$$Y_k(z) = \mathcal{Z}\{y_k[n]\} = \frac{X_e(z)}{1 - W_N^{-k} z^{-1}} = H_k(z) X_e(z)$$

where $H_k(z) = Z\{h_k[n]\} = 1/(1 - W_N^{-k} z^{-1})$ and $X_e(z) = Z\{x_e[n]\}$

• Thus, $y_k[n]$ is the output of an initially relaxed LTI digital filter $H_k(z)$ with an input $x_e[n]$ and, when n = N, $y_k[N] = X[k]$

• Structural interpretation of the algorithm -



• Thus a recursive DFT computation scheme is $y_k[n] = x_e[n] + W_N^{-k} y_k[n-1], \quad 0 \le n \le N$

with $y_k[-1] = 0$ and $x_e[N] = 0$

- Since a complex multiplication can be implemented with 4 real multiplications and 2 real additions, computation of each new value of y_k[n] requires 4 real multiplications and 4 real additions
- Thus computation of $X[k] = y_k[N]$ involves 4N real multiplications and 4N real additions

Computation of all *N* DFT samples requires $4N^2$ real multiplications and $4N^2$ real additions

- Recall, direct computation of all N samples of {X[k]} requires N² complex multiplications and N(N-1)complex additions
- Equivalently, direct computation of all N samples of {X[k]} requires 4N² real multiplications and N(4N-2) real additions
- Thus, Goertzel's algorithm requires 2*N* more real additions than the direct DFT computation

 Algorithm can be made computationally more efficient by observing that H_k(z) can be rewritten as

$$H_{k}(z) = \frac{1}{1 - W_{N}^{-k} z^{-1}} = \frac{1 - W_{N}^{k} z^{-1}}{(1 - W_{N}^{-k} z^{-1})(1 - W_{N}^{k} z^{-1})}$$
$$= \frac{1 - W_{N}^{k} z^{-1}}{1 - 2\cos(2\pi k/N) z^{-1} + z^{-2}}$$
resulting in a second-order realization



• DFT computation equations are now $v_k[n] = x_e[n] + 2\cos(2\pi k/N)v_k[n-1]$ $-v_k[n-2], \ 0 \le n \le N$ $X[k] = y_k[N] = v_k[N] - W_N^k v_k[N-1]$

- Computation of each sample of $v_k[n]$ involves only 2 real multiplications and 4 real additions
- Complex multiplication by W_N^k needs to be performed only once at n = N
- Thus, computation of one sample of *X*[*k*] requires (2*N*+4) real multiplications and (4*N*+4) real additions
- Computation of all *N* DFT samples requires 2N(N+2) real multiplications and 4N(N+1) real additions

• In the realization of $H_{N-k}(z)$, the multiplier in the feedback path is

 $2\cos(2\pi(N-k)/N) = 2\cos(2\pi k/N)$

which is same as that in the realization of $H_k(z)$ $v_{N-k}[n] = v_k[n]$, i.e., the intermediate variables computed to determine X[k] can again be used to determine X[N-k]

• Only difference between the two structures is the feed-forward multiplier which is now W_N^{-k} , that is the complex conjugate of W_N^k

- Thus, computation of X[k] and X[N-k] require 2(N+4) real multiplications and 4(N+2) real additions
- Computation of all *N* DFT samples require approximately N^2 real multiplications and approximately $2N^2$ real additions
- Number of real multiplications is about onefourth and number of real additions is about one-half of those needed in direct DFT computation

- Consider a sequence x[n] of length $N = 2^{\mu}$
- Using a 2-band polyphase decomposition we can express its *z*-transform as $X(z) = X_0(z^2) + z^{-1}X_1(z^2)$

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where $X_{0}(z) = \sum_{\substack{n=0 \ N/2 \ -1}}^{(N/2)-1} x_{0}[n]z^{-n} = \sum_{\substack{n=0 \ N/2 \ -1}}^{(N/2)-1} x[2n]z^{-n}$ $X_{1}(z) = \sum_{\substack{n=0 \ n=0}}^{(N/2)-1} x_{1}[n]z^{-n} = \sum_{\substack{n=0 \ n=0}}^{(N/2)-1} x[2n+1]z^{-n}$ Copyright © 2001, S. K. Mitra

• Evaluating on the unit circle at *N* equally spaced points $z = W_N^{-k}$, $0 \le k \le N - 1$, we arrive at the *N*-point DFT of x[n]: $X[k] = X_0[\langle k \rangle_{N/2}] + W_N^k X_1[\langle k \rangle_{N/2}],$ $0 \le k \le N - 1$

where $X_0[k]$ and $X_1[k]$ are the (N/2)-point DFTs of the (N/2)-length sequences $x_0[n]$ and $x_1[n]$

Decimation-in-Time FFT
Algorithm
• i.e.,
$$X_0[k] = \sum_{r=0}^{(N/2)-1} x_0[r] W_{N/2}^{rk}$$

 $= \sum_{r=0}^{(N/2)-1} x[2r] W_{N/2}^{rk}, \ 0 \le k \le \frac{N}{2} - 1$
 $X_1[k] = \sum_{r=0}^{(N/2)-1} x_1[r] W_{N/2}^{rk}$
 $= \sum_{r=0}^{(N/2)-1} x[2r+1] W_{N/2}^{rk}, \ 0 \le k \le \frac{N}{2} - 1$

• Block-diagram interpretation

$$x[n] \xrightarrow{k_0[n]} X_0[n] \xrightarrow{k_0[n]} X_0[\langle k \rangle_{N/2}] \xrightarrow{k_0[n]} X[k]$$

$$x[n] \xrightarrow{k_0[n]} DFT \xrightarrow{k_0[\langle k \rangle_{N/2}]} \xrightarrow{k_0[k]} X[k]$$

$$x_1[n] \xrightarrow{k_0[n]} X_1[\langle k \rangle_{N/2}] \xrightarrow{k_0[k]} X[k]$$

• Flow-graph representation



- Direct computation of the *N*-point DFT requires N^2 complex multiplications and $N^2 - N \approx N^2$ complex additions
- Computation of the *N*-point DFT using the modified scheme requires the computation of two (N/2)-point DFTs that are then combined with N complex multiplications and N complex additions resulting in a total of $(N^2/2) + N$ complex multiplications and approximately $(N^2/2) + N$ complex additions Copyright © 2001, S. K. Mitra

Decimation-in-Time FFT Algorithm • For $N \ge 3$, $(N^2/2) + N < N^2$

- Continuing the process we can express X₀[k] and X₁[k] as a weighted combination of two (N/4)-point DFTs
- For example, we can write $X_0[k] = X_{00}[\langle k \rangle_{N/4}] + W_{N/2}^k X_{01}[\langle k \rangle_{N/4}],$ $0 \le k \le (N/2) - 1$

where $X_{00}[k]$ and $X_{01}[k]$ are the (*N*/4)point DFTs of the (*N*/4)-length sequences $x_{00}[n] = x_0[2n]$ and $x_{01}[n] = x_0[2n+1]$

• Likewise, we can express $X_{1}[k] = X_{10}[\langle k \rangle_{N/4}] + W_{N/2}^{k} X_{11}[\langle k \rangle_{N/4}],$ $0 \le k \le (N/2) - 1$

where $X_{10}[k]$ and $X_{11}[k]$ are the (*N*/4)point DFTs of the (*N*/4)-length sequences $x_{10}[n] = x_1[2n]$ and $x_{11}[n] = x_1[2n+1]$

• Block-diagram representation of the twostage algorithm $X_0[\langle k \rangle_{N/2}]$



• Flow-graph representation



- In the flow-graph shown N = 8
- Hence, the (*N*/4)-point DFT here is a 2point DFT and no further decomposition is possible
- The four 2-point DFTs, $X_{ij}[k]$, i, j = 0,1can be easily computed
- For example $X_{00}[k] = x[0] + W_2^k x[4], \ k = 0,1$

• Corresponding flow-graph of the 2-point DFT is shown below obtained using the identity $W_2^k = W_N^{(N/2)k}$



• Complete flow-graph of the 8-point DFT is shown below



- The flow-graph consists of 3 stages
- First stage computes the **four** 2-point DFTs
- Second stage computes the **two** 4-point DFTs
- Last stage computes the desired 8-point DFT
- The number of complex multiplications and additions at each stage is equal to 8, the size of the DFT

- Total number of complex multiplications and additions to compute all 8 DFT samples is equal to 8 + 8 + 8 = 24 = 8×3
- In the general case when $N = 2^{\mu}$, number of stages for the computation of the (2^{μ}) -point DFT in the fast algorithm will be $\mu = \log_2 N$
- Total number of complex multiplications and additions to compute all N DFT samples is N(log₂ N)

- In developing the count, multiplications with $W_N^0 = 1$ and $W_N^{N/2} = -1$ have been assumed to be complex
- Also the symmetry property of $W_N^{(N/2)+k} = -W_N^k$

has not been taken advantage of

• These properties can be exploited to reduce the computational complexity further

• Examination of the flow-graph



reveals that each stage of the DFT computation process employs the same basic computational module

• In the basic module two output variables are generated by a weighted combination of two input variables as indicated below

where $r = 1, 2, ..., \mu$ and $\alpha, \beta = 0, 1, ..., N - 1$



• Basic computational module is called a **butterfly computation**

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• Input-output relations of the basic module are:

$$\Psi_{r+1}[\alpha] = \Psi_r[\alpha] + W_N^{\ell} \Psi_r[\beta]$$
$$\Psi_{r+1}[\beta] = \Psi_r[\alpha] + W_N^{\ell+(N/2)} \Psi_r[\beta]$$

• Substituting $W_N^{\ell+(N/2)} = -W_N^{\ell}$ in the second equation given above we get

$$\Psi_{r+1}[\beta] = \Psi_r[\alpha] - W_N^{\ell} \Psi_r[\beta]$$

• Modified butterfly computation requires only one complex multiplication as indicated below



• Use of the above modified butterfly computation module reduces the total number of complex multiplications by 50%

• New flow-graph using the modified butterfly computational module for N = 8



- Computational complexity can be reduced further by avoiding multiplications by $W_N^0 = 1$, $W_N^{N/2} = -1$, $W_N^{N/4} = j$, and $W_N^{3N/4} = -j$
- The DFT computation algorithm described here also is efficient with regard to memory requirements
- <u>Note</u>: Each stage employs the same butterfly computation to compute $\Psi_{r+1}[\alpha]$ and $\Psi_{r+1}[\beta]$ from $\Psi_r[\alpha]$ and $\Psi_r[\beta]$

- At the end of computation at any stage, output variables $\Psi_{r+1}[m]$ can be stored in the same registers previously occupied by the corresponding input variables $\Psi_r[m]$
- This type of memory location sharing is called **in-place computation** resulting in significant savings in overall memory requirements

• In the DFT computation scheme outlined, the DFT samples *X*[*k*] appear at the output in a sequential order while the input samples *x*[*n*] appear in a different order



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- Thus, a sequentially ordered input *x*[*n*] must be reordered appropriately before the fast algorithm described by this structure can be implemented
- To understand the input reordering scheme represent the arguments of input samples x[n] and their sequentially ordered new representations Ψ₁[m] in binary forms

• The relations between the arguments *m* and *n* are as follows:

m:000001010011100101110111n:000100010110001101011111

- Thus, if $(b_2b_1b_0)$ represents the index *n* of x[n], then the sample $x[b_2b_1b_0]$ appears at the location $m = b_0b_1b_2$ as $\Psi_1[b_0b_1b_2]$ before the DFT computation is started
- i.e., location of Ψ₁[m] is in bit-reversed
 order from that of x[n]

- Alternative forms of the fast DFT algorithms can be obtained by reordering the computations such as input in normal order and output in bit-reversed order, and both input and output in normal order
- The fast algorithm described assumes that the length of *x*[*n*] is a power of 2
- If it is not, the length can be extended by zero-padding and make the length a power of 2
- Even after zero-padding, the DFT computation based on the fast algorithm may be computationally more efficient than a direct DFT computation of the original shorter sequence
- The fast DFT computation schemes described are called decimation-in-time (DIT) fast Fourier transform (FFT) algorithms as input x[n] is first decimated to form a set of subsequences before the DFT is computed

 For example, the relation between x[n] and its even and odd parts, x₀[n] and x₁[n], generated by the first stage of the DIT algorithm is given by

Likewise, the relation between x[n] and the sequences x₀₀[n], x₀₁[n], x₁₀[n], and x₁₁[n], generated by the two-stage decomposition of the DIT algorithm is given by

x[n]:	<i>x</i> [0]	<i>x</i> [1]	<i>x</i> [2]	<i>x</i> [3]	<i>x</i> [4]	<i>x</i> [5]	<i>x</i> [6]	<i>x</i> [7]
$x_{00}[n]$:	<i>x</i> [0]				<i>x</i> [4]			
$x_{01}[n]$:	<i>x</i> [2]				<i>x</i> [6]			
$x_{10}[n]$:	<i>x</i> [1]				<i>x</i> [5]			
$x_{11}[n]$:	<i>x</i> [3]				<i>x</i> [7]			

• The subsequences $x_{00}[n]$, $x_{01}[n]$, $x_{10}[n]$, and $x_{11}[n]$ can be generated directly by a factorof-4 decimation process leading to a singlestage decomposition as shown on the next slide



- Radix-*R* FFT algorithm A each stage the decimation is by a factor of *R*
- Depending on *N*, various combinations of decompositions of *X*[*k*] can be used to develop different types of DIT FFT algorithms
- If the scheme uses a mixture of decimations by different factors, it is called a **mixed** radix FFT algorithm

• For *N* which is a composite number expressible in the form of a product of integers:

$$N = r_1 \cdot r_2 \cdots r_{\nu}$$

total number of complex multiplications (additions) in a DIT FFT algorithm based on a ν -stage decomposition is given by $\left(\sum_{i=1}^{\nu} \right)_{i=1}^{N}$

$$\sum_{i=1}^{\nu} r_i - \nu \bigg) N$$

- Consider a sequence x[n] of length $N = 2^{\mu}$
- Its *z*-transform can be expressed as

$$X(z) = X_a(z) + z^{-N/2} X_b(z)$$

where

$$X_{a}(z) = \sum_{\substack{n=0\\ N/2 - 1\\ n=0}}^{(N/2)-1} x[n]z^{-n}$$
$$X_{b}(z) = \sum_{\substack{n=0\\ n=0}}^{(N/2)-1} x[\frac{N}{2} + n]z^{-n}$$

• Evaluating X(z) on the unit circle at

we get

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

$$+ W_N^{(N/2)k} \sum_{\substack{n=0 \ n=0}}^{(N/2)-1} x[\frac{N}{2} + n] W_N^{nk}$$
which can be rewritten using the identity

$$W_N^{(N/2)k} = (-1)^k \text{ as}$$

$$X[k] = \sum_{\substack{n=0 \ n=0}}^{(N/2)-1} (x[n] + (-1)^k x[\frac{N}{2} + n]) W_N^{nk}$$
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• For *k* even

n=0

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

=
$$\sum_{\substack{n=0 \ n=0}}^{(N/2)-1} (x[n] + x[\frac{N}{2} + n]) W_{N/2}^{n\ell}, \quad 0 \le \ell \le \frac{N}{2} - 1$$

• For k odd $X[2\ell+1] = \sum_{n=0}^{(N/2)-1} (x[n] - x[\frac{N}{2} + n]) W_N^{n(2\ell+1)}$ $= \sum_{n=0}^{(N/2)-1} (x[n] - x[\frac{N}{2} + n]) W_N^n W_{N/2}^{n\ell}, \quad 0 \le \ell \le \frac{N}{2} - 1$

• We can write

$$X[2\ell] = \sum_{n=0}^{(N/2)-1} x_0[n] W_N^{n(2\ell)}$$

$$X[2\ell+1] = \sum_{n=0}^{(N/2)-1} x_1[n] W_N^{n(2\ell)}, \ 0 \le \ell \le \frac{N}{2} - 1$$

where

$$x_0[n] = (x[n] + x[\frac{N}{2} + n]),$$

$$x_1[n] = (x[n] - x[\frac{N}{2} + n])W_N^n, \quad 0 \le n \le \frac{N}{2} - 1$$

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- Thus $X[2\ell]$ and $X[2\ell+1]$ are the (N/2)point DFTs of the length-(N/2) sequences $x_0[n]$ and $x_1[n]$
- Flow-graph of the first-stage of the DFT algorithm is shown below



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Here the input samples are in sequential order, while the output DFT samples appear in a decimated form with the even-indexed samples appearing as the output of one (*N*/2)-point DFT and the odd-indexed samples appearing as the output of the other (*N*/2)-point DFT

- We next express the even- and odd-indexed samples of each one of the two (*N*/2)-point DFTs as a sum of two (*N*/4)-point DFTs
- Process is continued until the smallest DFTs are 2-point DFTs

• Complete flow-graph of the decimation-infrequency FFT computation scheme for N = 8



- Computational complexity of the radix-2 DIF FFT algorithm is same as that of the DIT FFT algorithm
- Various forms of DIF FFT algorithm can similarly be developed
- The DIT and DIF FFT algorithms described here are often referred to as the Cooley-Tukey FFT algorithms

Inverse DFT Computation

- An FFT algorithm for computing the DFT samples can also be used to calculate efficiently the inverse DFT (IDFT)
- Consider a length-*N* sequence *x*[*n*] with an *N*-point DFT *X*[*k*]
- Recall

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

Inverse DFT Computation

• Multiplying both sides by *N* and taking the complex conjugate we get

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

 Right-hand side of above is the *N*-point DFT of a sequence X*[k]

Inverse DFT Computation

• Desired IDFT *x*[*n*] is then obtained as

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

• Inverse DFT computation is shown below:

