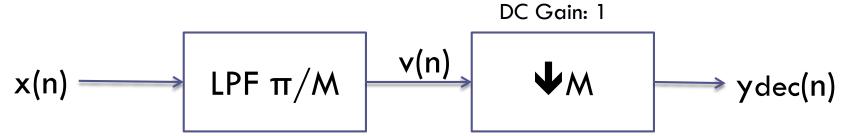
#### POLYPHASE FILTERS

- □ In order to decimate a signal, we lowpass it and then downsample it
  - We filter the whole signal, but we are going to use only a part of it (one every M samples)

MMSP 1 - 08 Polyphase Filtering



We design polyphase filters in order to save computations in the decimation phase

- A filter is decomposed into its polyphase components
- The signal is decomposed into subsequences
- The subsequences are filtered with the correspondent subfilters

MMSP 1 - 08 Polyphase Filtering

The output is reconstructed as the sum of the subresults

- □ Suppose we have x(n)=[x(0), x(1), ..., x(N-1)]
- □ We want to decimate with M=3
- □ We have a 1/M lowpass filter
  - □  $h(n) = [\clubsuit, \blacklozenge, \lor, \land, \lozenge, \nabla]$  with 6 weights
  - Both h(n) and x(n) start from n=0
- Let's compute the convolution and then the downsampling

- $\square$  First step: folding h(n)  $\rightarrow$  h(-n)
  - $\blacksquare h(-n) = [\nabla, \Diamond, \blacktriangle, \blacktriangledown, \blacktriangle, \clubsuit]$

x(0), x(1), x(2), x(3), ..., x(N)x(n)= $h(-n)=\nabla, \Diamond, \blacktriangle, \blacktriangledown, \blacklozenge, \clubsuit$ 

$$y(0) = A x(0)$$

- □ Next sample: y(1)
  - We know we are going to neglect y(1) and y(2) (M=3)
  - Let's compute directly y(3)

x(n) = x(0), x(1), x(2), x(3), ..., x(N)

$$h(-n)=\nabla$$
,  $\Diamond$ ,  $\spadesuit$ ,  $\heartsuit$ ,  $\spadesuit$ ,

$$y(3) = x(0) + x(1) + x(2) + x(3)$$

□ Next sample:  $\frac{y(4)}{y(6)}$ 

x(n) = x(0), x(1), x(2), x(3), x(4), x(5), x(6), ..., x(N)h(-n)=  $\nabla$ ,  $\Diamond$ ,  $\bigstar$ ,  $\blacktriangledown$ ,  $\bigstar$ ,

$$y(6) = \nabla x(1) + \Diamond x(2) + Ax(3) + \nabla x(4) + Ax(5) + Ax(6)$$

- □ And so on...
- □ Let's summarize...

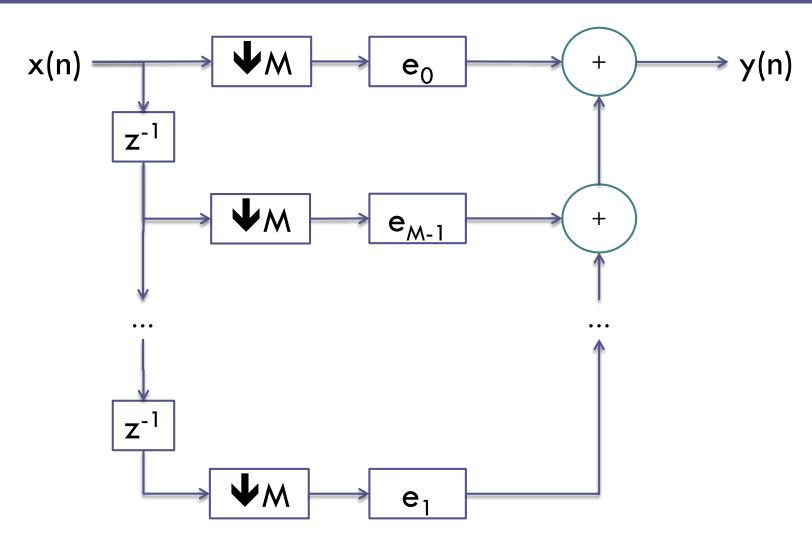
y(0) = Ax(0)y(3) = Ax(3) + Ax(2) + Ax(1) + Ax(0) $y(6) = Ax(6) + Ax(5) + Ax(4) + Ax(3) + Ax(2) + \nabla x(1)$ y(9) = Ax(9) + Ax(8) + Ax(7) + Ax(6) + Ax(5) + Ax(4)

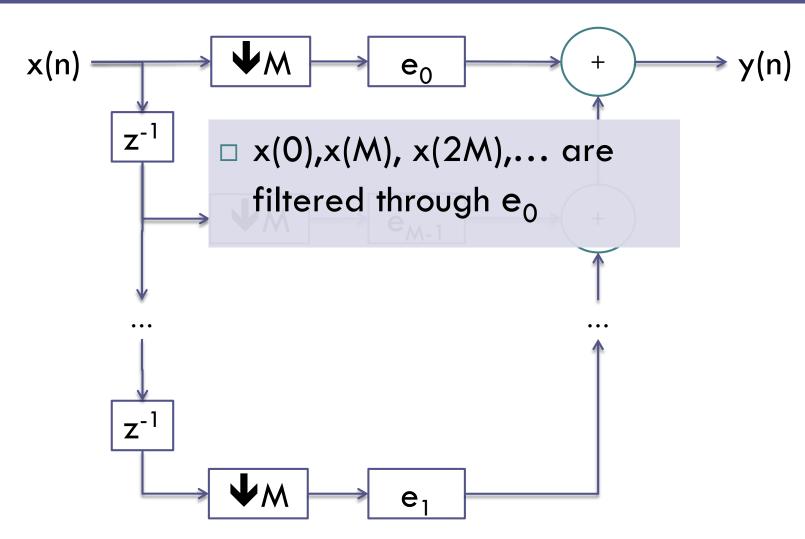
```
\lambda(0) = *x(0) + \cdots
y(3) = *x(3) + *x(2) + *x(1) + *x(0)
y(6) = *x(6) + *x(5) + *x(4) + *x(3) + 0 \times (2) + \nabla \times (1)
y(9) = *x(9) + *x(8) + *x(7) + *x(6) + $0 \times (5) + $0 \times (4)
     \square h(n)=[\clubsuit, \diamondsuit, \heartsuit, \diamondsuit, \nabla]
     \blacksquare e_0 = [\clubsuit, \blacktriangle] \rightarrow [x(0), x(3), ..., x(kM+0)]
     \blacksquare e_1 = [\blacklozenge, \lozenge] \rightarrow [\mathbf{0} \times (2), \times (5), \dots, \times (kM+2)]
     \blacksquare e_2 = [ \lor, \lor] \rightarrow [0 \times (1), \times (4), \dots, \times (kM-m)]
```

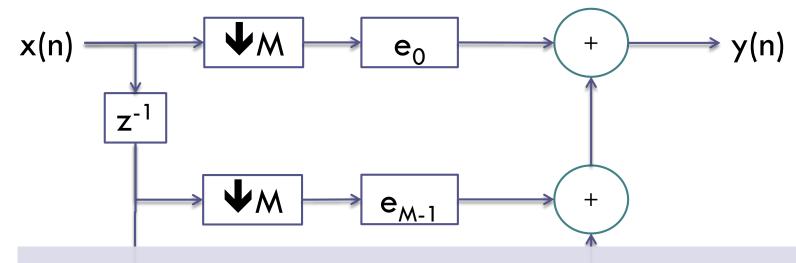
- $\Box$  Given a filter h with L weights
- $\square$  Given a signal x with M down-sampling factor
- $\square$  we have  $K = \lceil L/M \rceil$  weights for each subfilter
- $\square$  Each subfilter  $e_m$  is build as:
  - $\bullet e_m(k) = h(kM + m)$ 
    - = k = [0,1...,K-1]
    - m=[0,1,...,M-1]
- Write a function e=decompose\_filter(h, M)
  - □ e is a matrix with K rows (weights) and M columns (subfilters)

```
MMSP 1 - 08 Polyphase Filtering
function [ e ] = decompose filter(h, M )
  L=length(h); K=ceil(L/M);
  e=zeros(K,M);
  if K*M>I
     h(K*M)=0; % zero-pad up to K*M
  end
  for m=1:M
    sub=h (m:M:end);
    e(:,m) = sub(:);
  end
end
```

□ Try to implement it using the reshape function





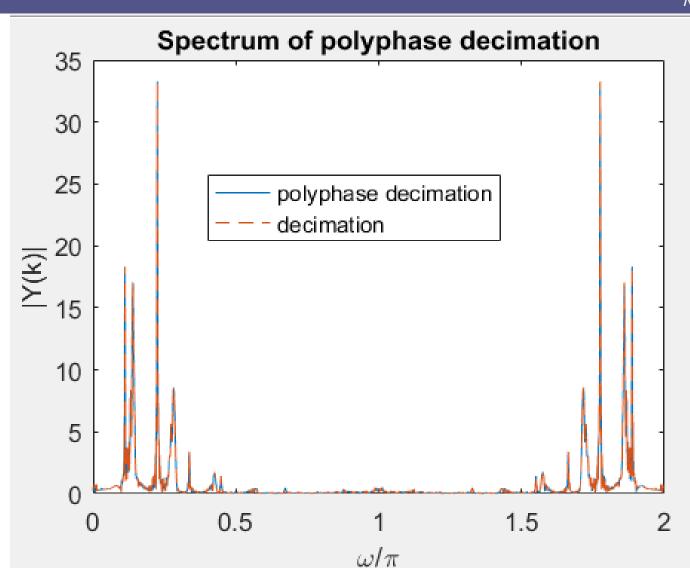


- $\square$  0, x(1),x(M+1), x(2M+1),... are filtered through  $e_{M-1}$
- In order to take old samples into account, it is required to add a 0 at the beginning of the sequence

- ☐ Given a sequence x with L weights
- Given a signal x with M down-sampling factor
- $\square$  we have K = [L/M] weights for each subfilter  $e_m$
- $\square$  We build K subsequences  $\mathcal{X}_m$ 
  - $\mathbf{x}_m(k) =$ 
    - $\blacksquare$  0 if k==0 and m!=0 (take old samples into account)
    - x(Mk m) otherwise
    - = k=[0,1...,K-1]
    - = m=[0,1,...,M-1]
- $\square$  Write a function x m=decompose signal(x, M)
  - x m is a matrix with Nm rows (samples) and M columns (subfilters)

```
function [xm] = decompose signal(x,M)
    Nm=floor(length(x)/M);
    x m=zeros(Nm,M);
    x 0=x (1:M:end);
    x m(1:length(x 0), 1) = x 0;
    for m=1:M-1
        x = x (M-(m-1) : M : end);
        x m(2:length(x)+1,m+1)=x;
    end
end
```

```
[x,Fs]=audioread('Toms diner 16.wav');
n x=[0:length(x)-1]/Fs;
M=4; h=fir1(32,1/M);
e=decompose filter(h,M);
x = decompose signal(x, M);
y=zeros(size(x m(:,1)));
for m=1:M
   y m=filter(e(:,m),1,x m(:,m));
   y=y+y m;
end
n y = [0:length(y)-1]/(Fs/M);
```

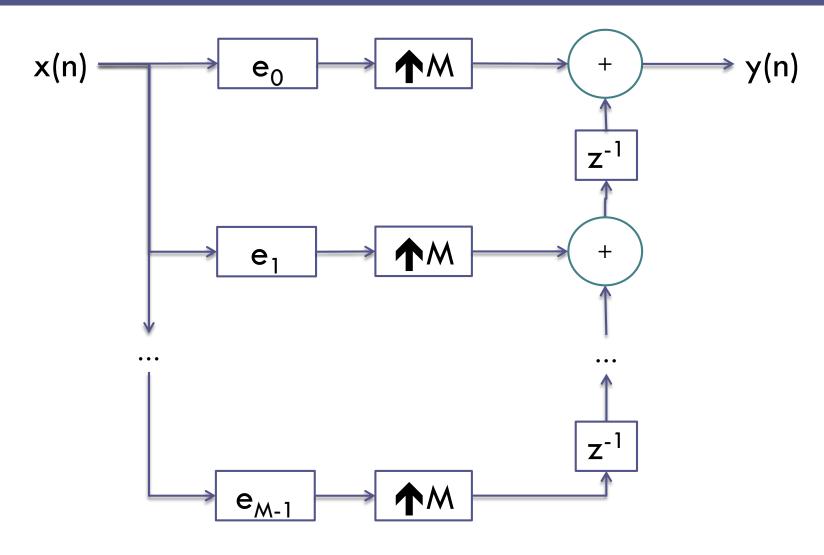


- Polyphase interpolation is the dual of polyphase decimation
  - In classic intepolation, the sequence is upsampled before the low-pass filtering

- a signal L times longer then the original is filtered
- Polyphase interpolation is used to achieve computational efficiency

- ☐ Given a filter h with L weights
- □ Given a signal with M up-sampling factor
- $\square$  we have  $K = \lceil L/M \rceil$  weights for each subfilter

- $\square$  Each subfilter  $e_m$  is build as:
  - $\blacksquare e_m(k) = h(kM + m)$ 
    - = k = [0,1...,K-1]
    - = m = [0,1,...,M-1]



```
% see above for y and e
y int=zeros(length(y)*M+M,1);
for m=1:M
 y m f=filter(M*e(:,m),1,y); % scaling
 y int(m:M:end-M) = y int(m:M:end-M) + y m f;
End
```

