Multimedia Communications

Subband Coding

McMaster University

Subband coding

- Subband coding: decompose the input signal into different frequency bands
- After the input is decomposed to its constituents, we can use the coding technique best suited to each constituent to improve the compression performance
- Each component may have different perceptual characteristics
  - Quantization errors that are objectionable in one component may be acceptable in a different component

Copyright S. Shirani

Subband Coding

- Idea: decompose a signal into components by applying frequency-selective filtering. Then select the best coding technique that best suits each component (subjectively and objectively).
- Example: slow- and fast-varying components.

$$y[n] = (x[n] + x[n-1])/2 \qquad z[n] = (x[n] - x[n-1])/2$$

The signal can be recovered: $x[n] = y[n] + z[n]$

The filters are: $h[n] = (\delta[n] + \delta[n-1])/2 \quad g[n] = (\delta[n] - \delta[n-1])/2$

Copyright S. Shirani

Subband Coding

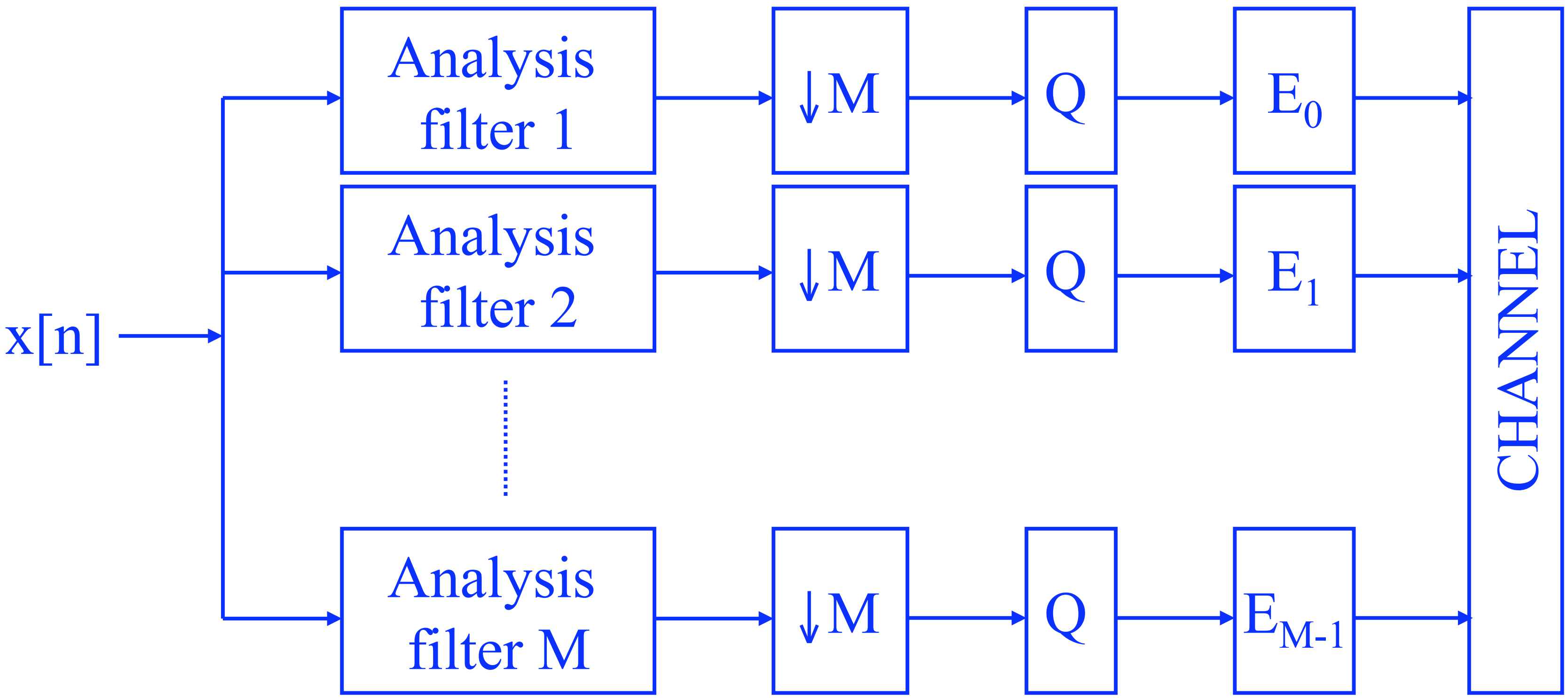
- If we use the same number of bits for each of y[n] and z[n], we are transmitting twice as many samples, doubling the bit rate.
- We can avoid this by sending every other value of y[n] and z[n] (e.g., even numbered elements)

$$y[2n] = (x[2n] + x[2n-1])/2$$
$$z[2n] = (x[2n] - x[2n-1])/2$$

$$x[2n] = y[2n] + z[2n]$$
$$x[2n-1] = y[2n] - z[2n]$$

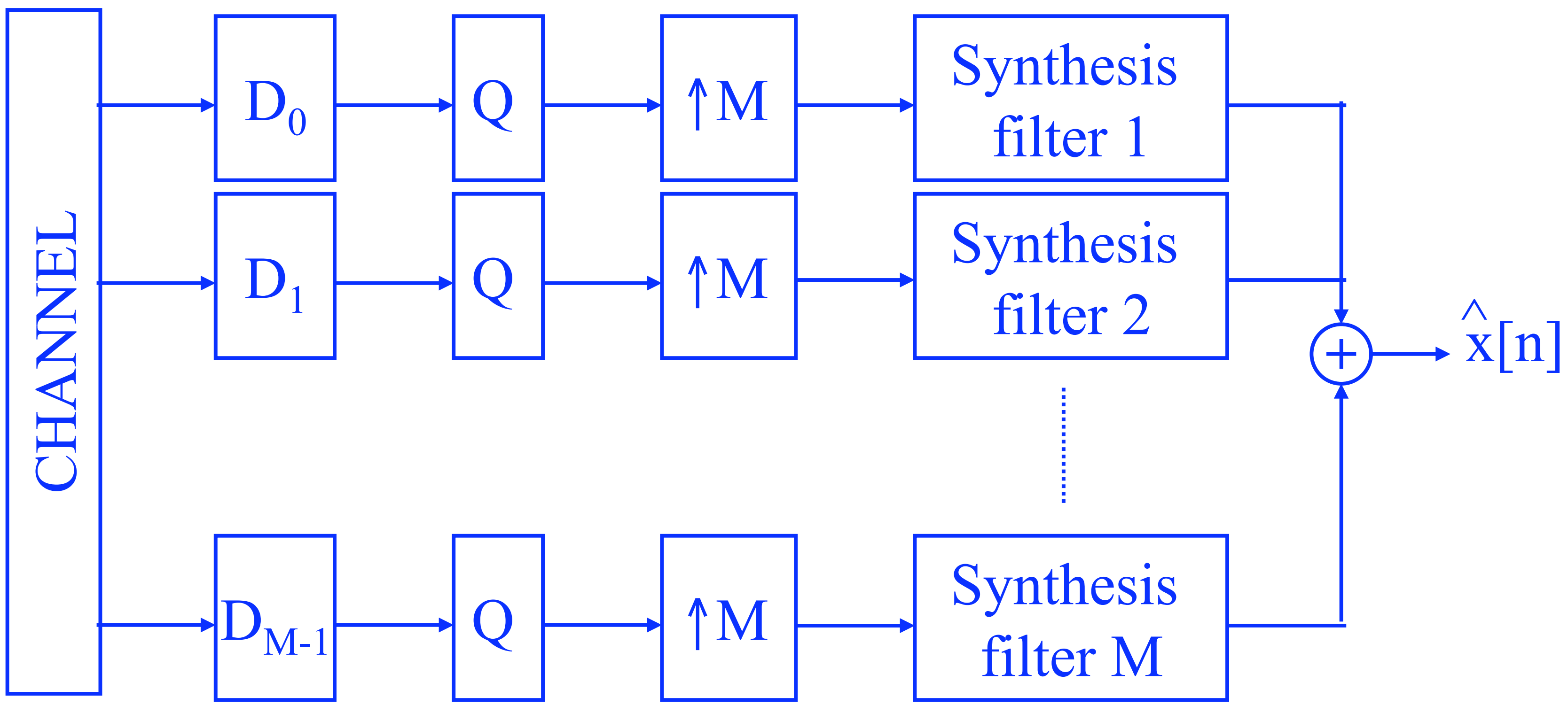
Copyright S. Shirani

Subband Encoding



Copyright S. Shirani

Subband Decoding



McMaster University

Copyright S. Shirani

Subband Coding

Analysis

- Source output is passed through a bank of filters (analysis filters)
- Analysis filters cover the range of frequencies that make up source output
- Passband of the filters can be non-overlapping or overlapping
- Output of filters are then subsampled (also called decimation or downsampling)
- Justification for subsampling: Nyquist rule (range of frequencies of output of the filter is less than input to the filter)

Copyright S. Shirani

Subband Coding

Quantization, coding and bit allocation

- Selection of compression scheme and allocation of bits between subbands is important and can have significant impact on the quality of the final reconstruction

Synthesis

- Encoded samples from each subband are decoded
- Decoded values are then upsampled by inserting an appropriate number of 0s between samples
- Upsampled signals are passed through a bank of reconstruction filters
- Output of reconstruction filters are added to give final output

Copyright S. Shirani

Subband Coding

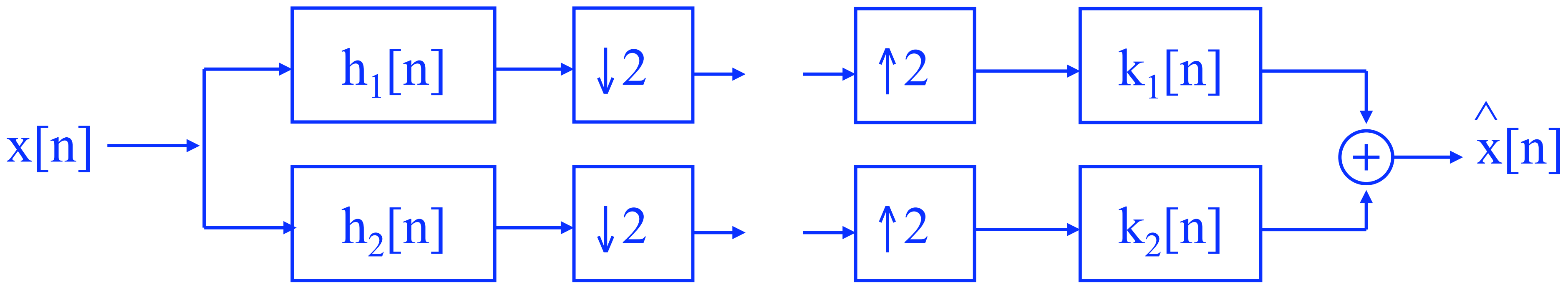
- Three major components of subband system are:

1. Analysis and synthesis filters
   - Simple to implement, good separation between frequency bands
2. Bit allocation (quantization)
   - Can have s significant affect on the quality of the reconstruction
3. Encoding scheme
   - Based on the characteristics of each of the subbands, we can use a separate compression scheme
   - Human perception is frequency dependent. We can use this fact to design our compression scheme so that the frequency bands that are most important to perception are reconstructed most accurately

McMaster University

Copyright S. Shirani

Filter Banks: Two-Band



- For M=2 the filters are easy to analyze.
- Goal:
  - good frequency-domain separation
  - no aliasing terms
  - perfect reconstruction: system is equivalent to a delay

McMaster University

Copyright S. Shirani

Filter Banks: Two-Band

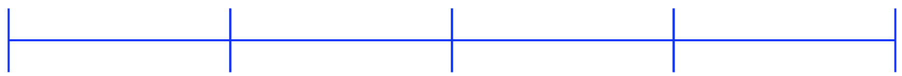
- Quadrature Mirror Filters (QMF) solution
  - no aliasing, no magnitude distortion, some phase distortion
  - the filters are symmetric and
  - set of filters have been designed by Johnston
  - the decomposition efficiency increases with the length
- Conjugate Quadrature Filters (CQF, Smith-Barnwell) solution
  - perfect reconstruction
  - better frequency characteristics for the same nr. of taps
  - closely related to wavelets

Copyright S. Shirani

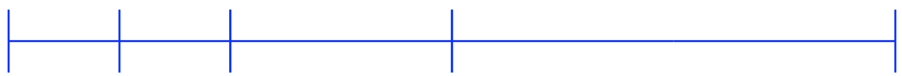
Filter Banks: Tree-Structured

- We can design an M-band filter bank by successively applying 2-band filter banks.

- Example: uniform filter bank decomposition



- Example: octave-band filter bank decomposition

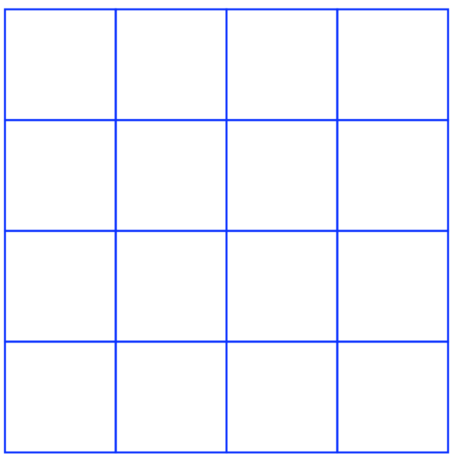


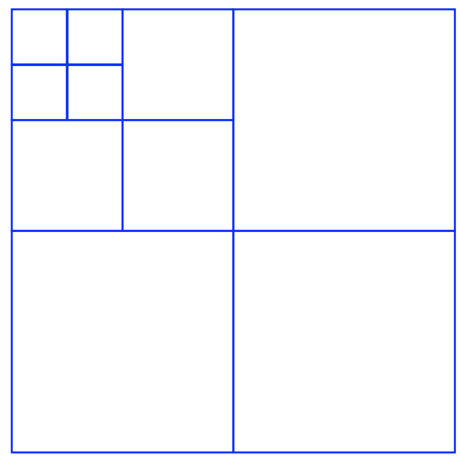
Copyright S. Shirani

2-Dimensional Filter Banks

- Most 2-D filter banks are obtained by applying 1-D decompositions separably.

Uniform decomposition   Octave-tree decomposition





McMaster University

Copyright S. Shirani

Bit allocation

- Once we have separated the source into subbands, we need to decide how much of the coding resources should be used to encode each subband
- B_T: total bits to distribute among M subbands
- R: average rate in bits per sample for the overall system
- R_k: average rate for subband k
- We assume that we have the rate-distortion function for each band
- We want to find R_k such that $$R = \frac{1}{M}\sum R_k$$

and the reconstruction error is minimized.

McMaster University

Copyright S. Shirani

Bit allocation

- Where on the rate distortion curve for each subband should we operate to minimize the average distortion?
- $J_k = D_k + \lambda R_k$
- D_k: distortion for the kth subband
- R_k: rate for the kth subband
- λ: Lagrangian parameter, specifies the tradeoff between rate and distortion
- Primary interested in minimizing the distortion: λ small
- Primary interested in minimizing the rate: λ large
- The value of D_k and R_k that minimize J_k occur where the slope of the rate-distortion curve is λ.

McMaster University

Copyright S. Shirani

Bit allocation

- What should the value of λ be and how should it change between subbands?
- Fact: we would like to allocate bits in such a way that any increase in any rates in any subbands will have the same impact on the distortion
- Why: because if the above it not true we can take the bits off the subband whose rate reduction has less effect on the distortion and assign it to other subbands
- We pick R_k in such a way that the slope of the rate distortion functions for different subbands are the same

Copyright S. Shirani

Bit allocation

- Given a set of rate-distortion functions and a value of λ, we can set the rates R_k, and compute the average rate.
- If it satisfies our constraint on the total rate we stop, otherwise we modify λ until we get a set of rates that satisfy our rate constraint
- Generally we do not have the rate-distortion function.
- We can use the operational rate-distortion curves.
- Operational: particular type of encoder operating on specific type of sources
  - Exp: pdf-optimized non-uniform quantizer with entropy coding
- If operational curve is available for a limited number of points we can estimate the other points or use curve fitting

Copyright S. Shirani

G. 722

- ITU recommendation G. 722: a technique for wideband coding of speech based on subband coding
- Objective: high-quality speech at 64 kbps.
- Recommendation has two other modes that code the input at 56 and 48 kbps (to leave some bandwidth for auxiliary channel)
- Speech is first filtered to 7kHz to prevent aliasing then sampled at 16,000 samples per second.
- Each sample is encoded using a 14-bit uniform quantizer.
- This 14-bit input is passed through a bank of two 24-coefficient FIR filter.

Copyright S. Shirani

G. 722

- Low-pass filter passes all frequency components in the range of 0 to 4 kHz.
- High-pass filter passes all remaining frequencies.
- The output of filters is downsampled by a factor of two.
- Downsampled sequences are encoded using adaptive differential PCM (ADPCM) system.
- ADPCM system that encodes the downsampled output of the low-frequency filter uses 6 bits per sample with the option of dropping 1 or 2 least significant bits (to provide room for auxiliary channels)
- Output of high-pass filter is encoded using 2 bits per sample.

Copyright S. Shirani

MPEG audio

- MPEG has proposed an audio scheme that is based on subband coding.
- MPEG has proposed three coding schemes: Layer1, Layer2, Layer 3
- Coders are upward compatible: a layer N decoder is able to decode bitstream generated by the layer N-1 encoder
- Layer1 and 2 coders use a bank of 32 filters.
- Sampling frequencies are 32,000, 44,100, and 48,000.
- Each subband is quantized with a variable number of bits.
- The number of bits assigned to each subband is determined by a psycho-acoustic model that uses the masking property of the human ears.

Copyright S. Shirani

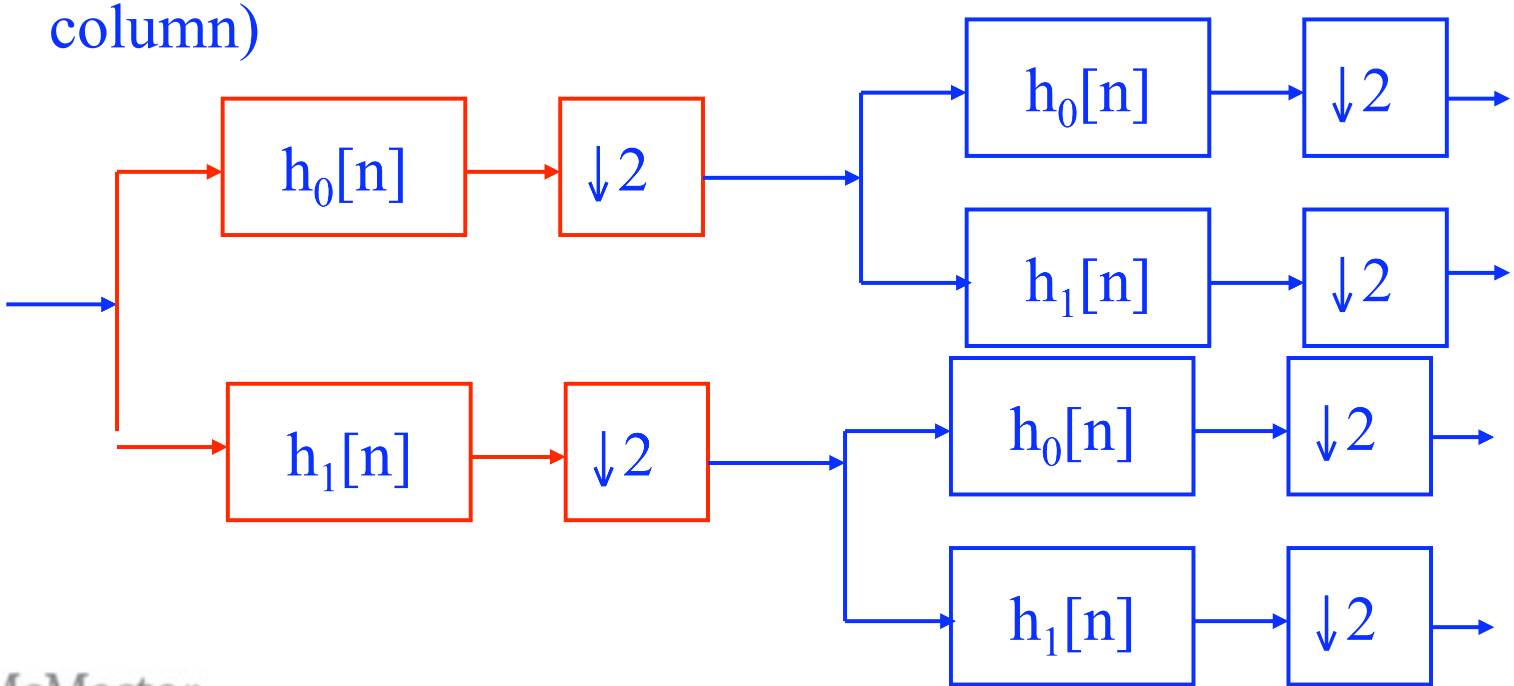
MPEG audio

- If we have a large amplitude signal at one frequency it affects the audibility of signals at other frequencies.
- A loud signal at one frequency may make quantization at other frequencies inaudible.
- If we have a large signal in one of the subbands, we can tolerate more quantization error in the neighboring subbands and use fewer bits.

Copyright S. Shirani

Image compression

- In most cases for subband coding of 2-D signals, we use separable filters.
- If the filters are separable, the 2-D filtering can be implemented as 2, 1-D filtering (filter each row and then each column)



McMaster University

Copyright S. Shirani

Image compression

- Question: when filtering the image pixels close to the borders what the past values of the signal are assumed to be?
  - Zero: not the best option
  - Reflect the values of pixels at the boundary: 6 9 5 4 7 2 is expanded to 9 6 6 9 5 4 7 2
- Once we decomposed an image into subbands, we need to find the best encoding scheme to use with each subband
- DPCM for the low-low band and scalar quantization for the other bands are common approaches.

Copyright S. Shirani