Multimedia Communications

Subband Coding

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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

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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.

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

$$\text{The signal can be recovered: } x[n] = y[n] + z[n]$$

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

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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)

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

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

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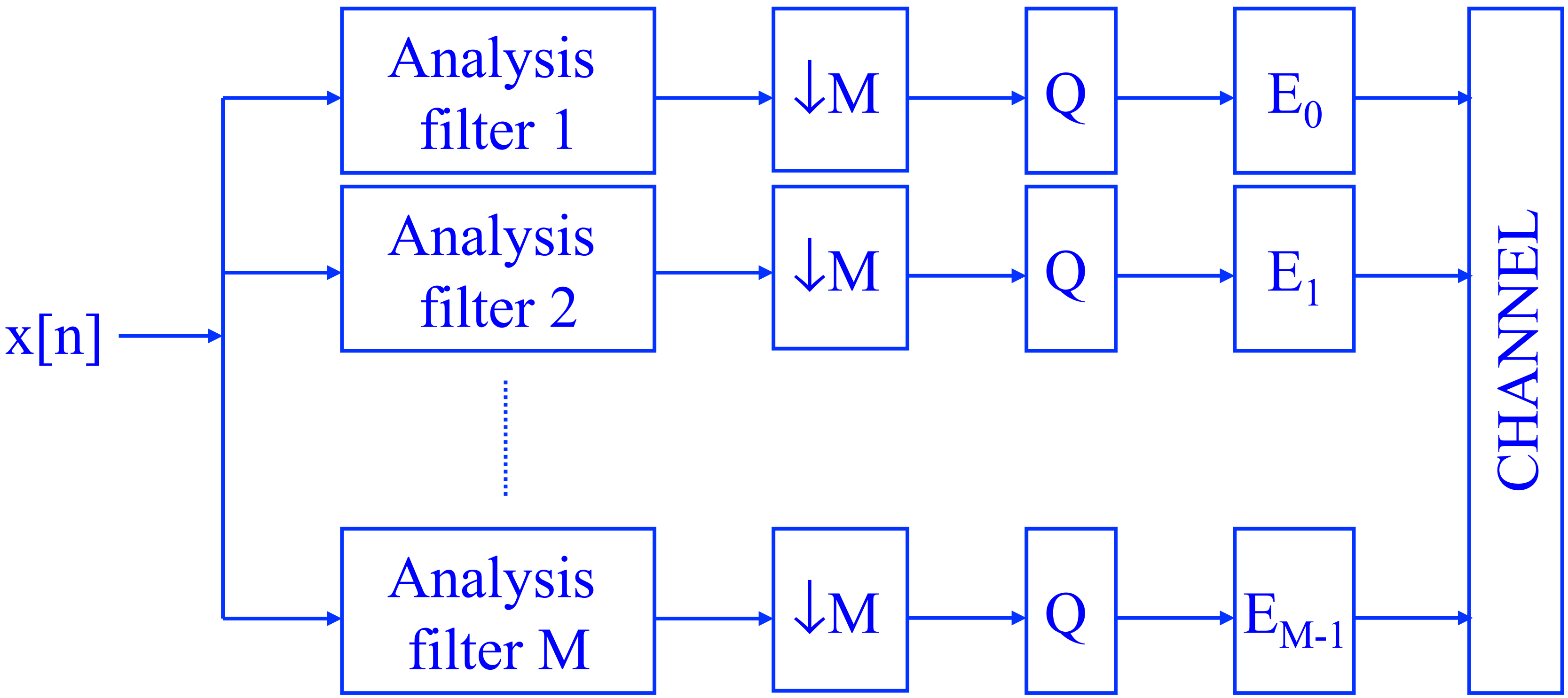
Subband Encoding



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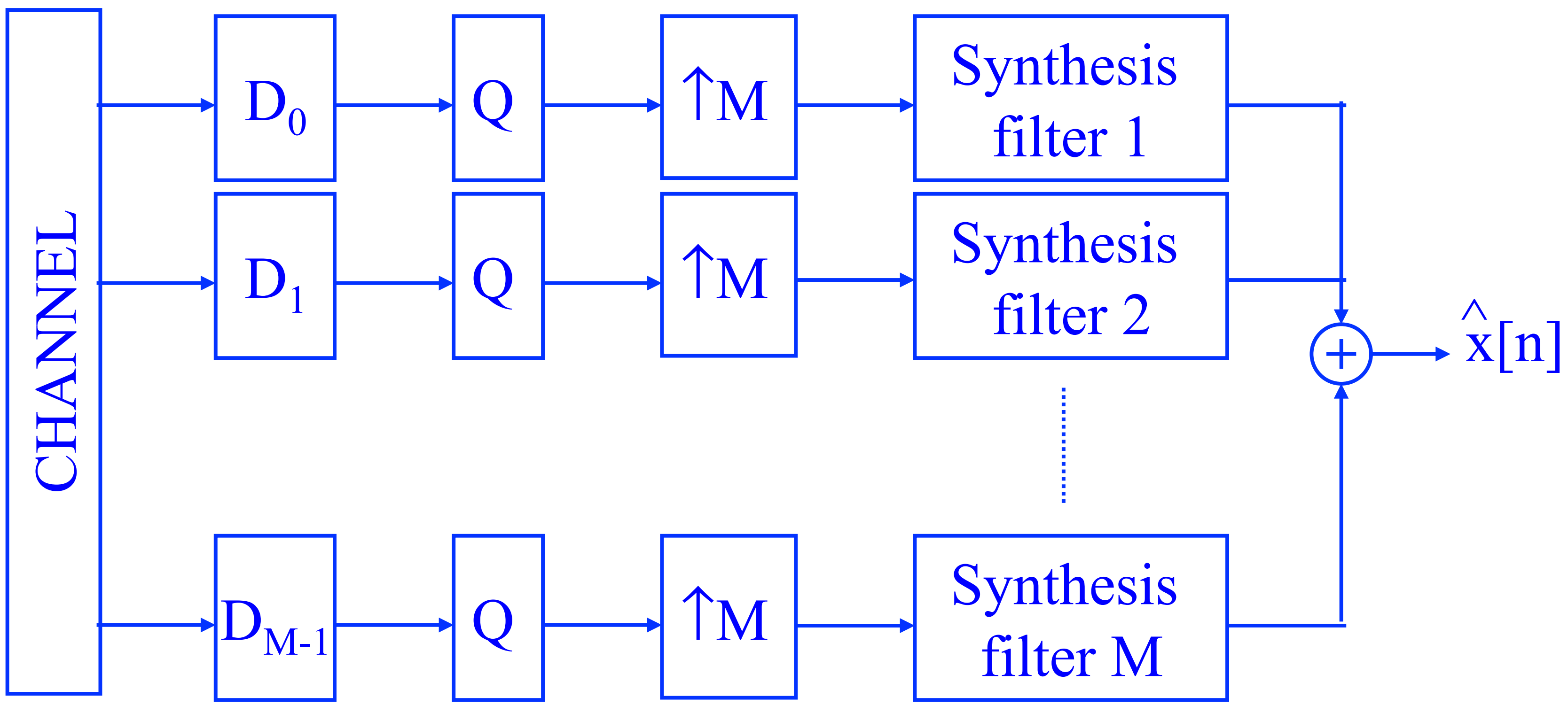
Subband Decoding



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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)

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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

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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

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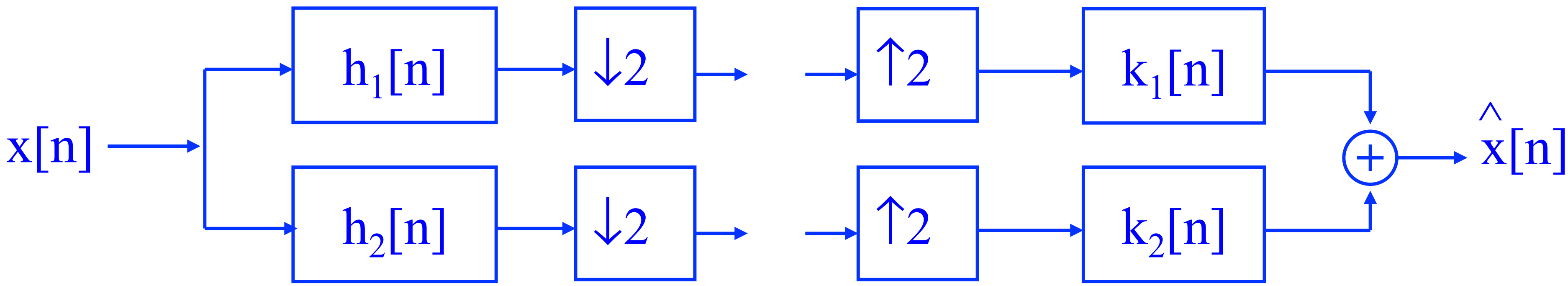
Filter Banks: Two-Band



- 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

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Filter Banks: Two-Band

- Quadrature Mirror Filters (QMF) solution
  - no aliasing, no phase distortion, some magnitude 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

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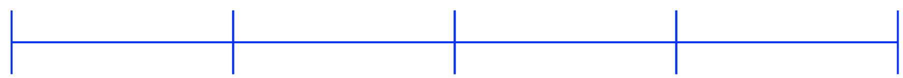
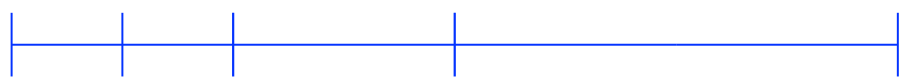
Filter Banks: Tree-Structured

- 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



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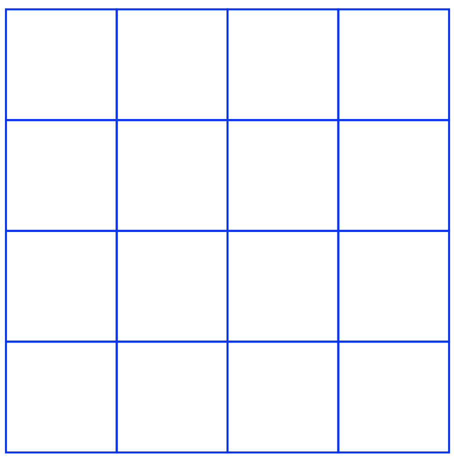
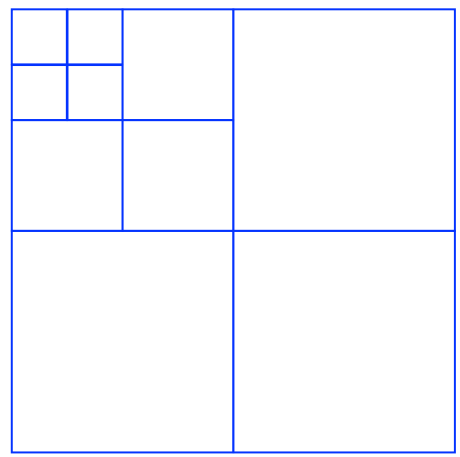
2-Dimensional Filter Banks

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

Uniform decomposition   Octave-tree decomposition





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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.

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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.

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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

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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

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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.

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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.

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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.

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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.

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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)

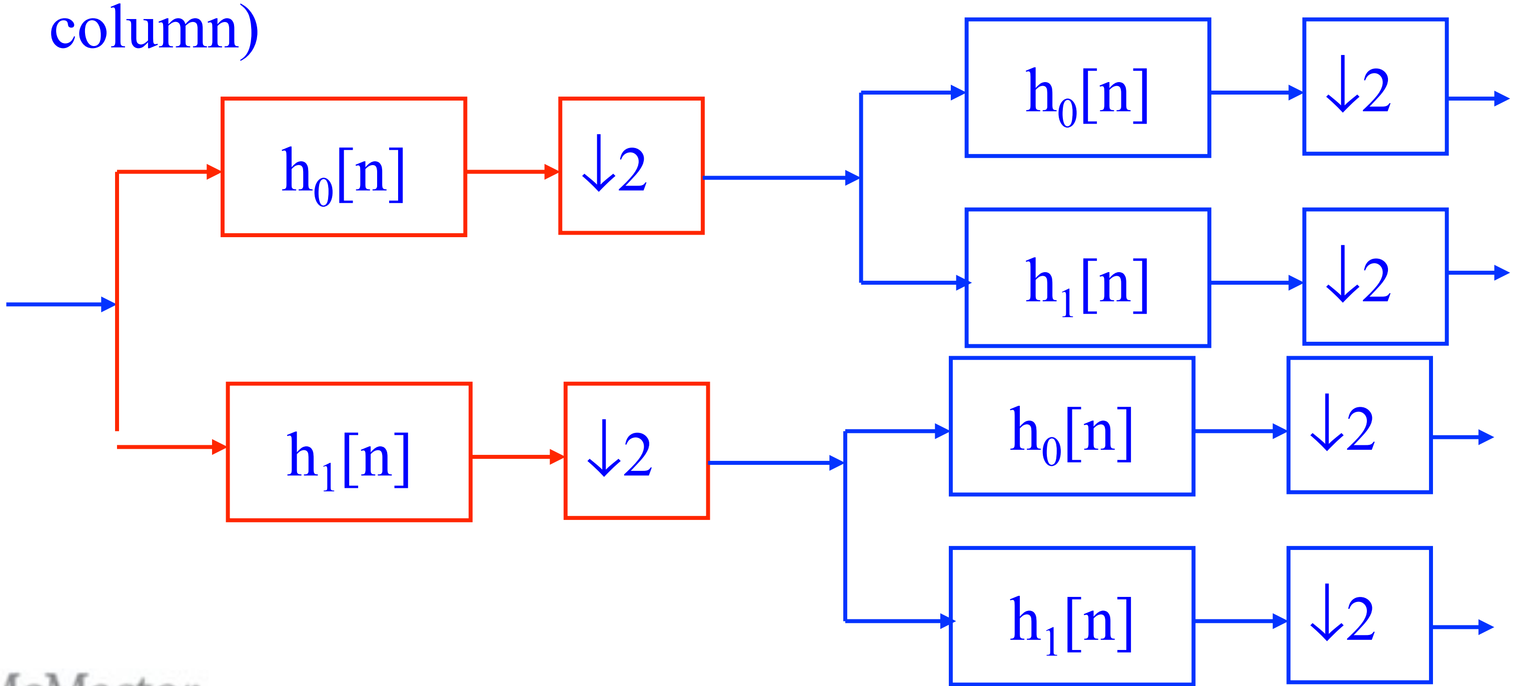

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Image compression

- 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.

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