Lecture notes for SSY150: Multimedia and video communications

Compression of Speech and Audio Signals

(Lecture 2)

Irene Y.H. Gu

Chalmers Univ. of Technology, Sweden

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Content

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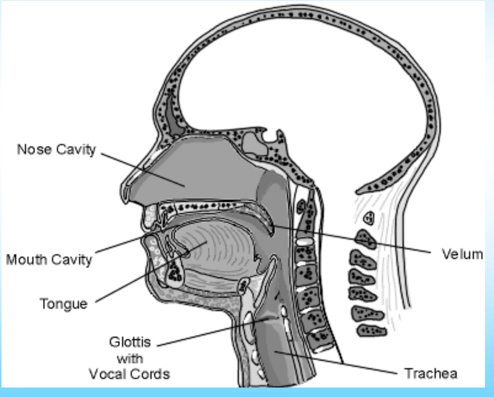
1. Speech/Audio production

Human Speech Organ:

Speech: results from the combination of the lung, glottis (vocal cords), and mouth-nose cavity



(*K.Fellbaum,* Brandenburg Tech. Univ. Cottbus, Germany)

Fig. Mouth and nose cavity acting as an articulating tract

(From: http://www.kt.tu-cottbus.de/speech-analysis)

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2. Basic methods for speech compression

Parametric (model-based)

- LPC analysis /synthesis
- Code-excited LPC (CELP) analysis / synthesis

Non-parametric (non-model based)

- Subband coding
- Transform coding
- Vector quantization

Compression is achieved by:

- using model parameters

- remove small coefficients in freq. bands
- different bit allocation in freq. bands

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Parametric methods
(model-based methods)

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a) LPC model for speech analysis/synthesis

LPC – Linear Predictive Coding: is related to a AR model/all pole model

$$s(n) = \frac{G}{1 + \sum_{i=1}^{p} a_i z^{-i}} w(n)$$

For each short time (e.g. 10-20ms) of speech (approx. stationary!), only a few parameters are required for synthesizing the speech:

- p LPC-coefficients (how many p's ? What is the principle to choose #p ?),
- Gain G,
- voiced/unvoiced indicator
- pitch period T (for voiced speech, to generate impulse sequence with T interval)

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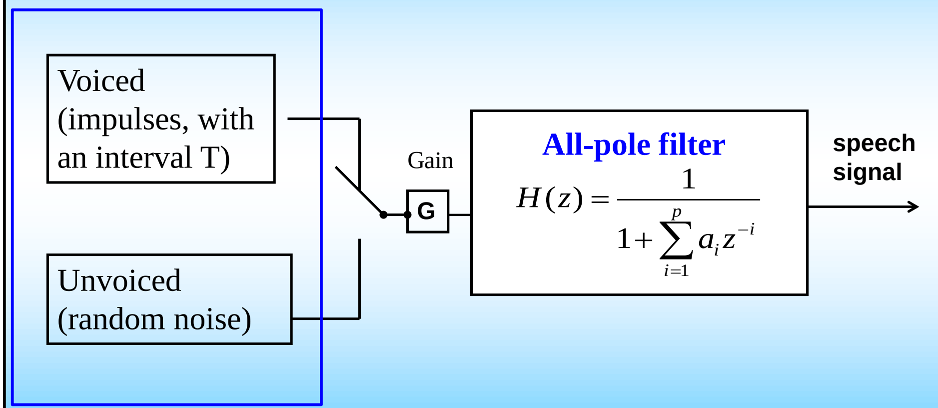
LPC Speech Synthesis



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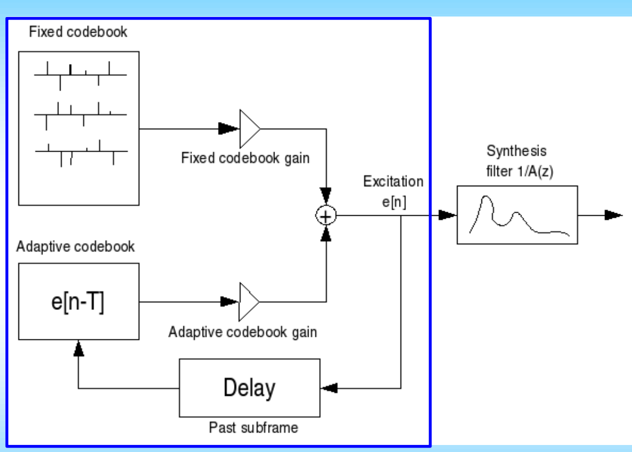
b) Code-Excited LP (CELP) for speech analysis/synthesis

Modeling the noise

Modeling the pitch



(figure from 'wikipedia, the free encyclopedia')

Vocal tract: LPC (AR model)
Excitations: an adaptive (pitch) codebook + a fixed stochastic VQ codebook

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c) Damped sinusoids in white noise for music compression

$$s(n) = \sum_{i=1}^{K} a_i e^{-\beta_i n} \cos(\omega_i n + \phi_i) + v(n)$$

For each 10-20ms of audio signal, only a few parameters are required for re-synthesizing the signal:

- damping factors β_i
- amplitudes a_i
- frequency ω_i
- initial phase ϕ_i

$i = 1, \cdots K$

These parameters can be estimated by the ESPRIT / MUSIC algorithms.

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Non-parametric methods
(non-model based methods)

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General principles of non-parametric methods

Decompose audio/speech signal by:

- subband filters (filterbank), with
  - equal bandwidths
  - octave bandwidths
  - transformation

Achieve compression through:

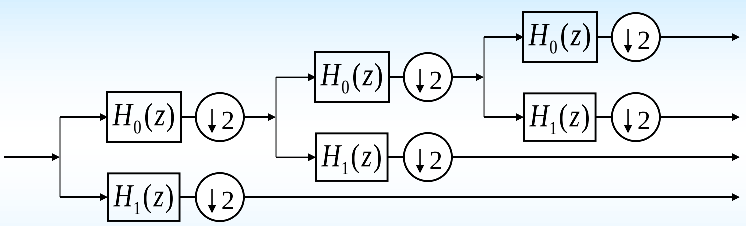
- set the bandwidth consistent to human auditory system
- bit allocation in different bands
  (set different number of quantization levels for different bands)
- variable length coding
  (set the length according the probability of quantizer outputs)

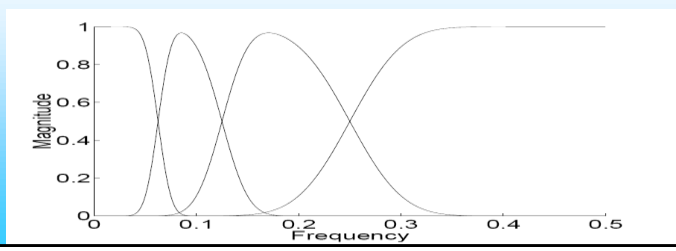
MP3 and MP4 belong to this category!

1) Subband Filters

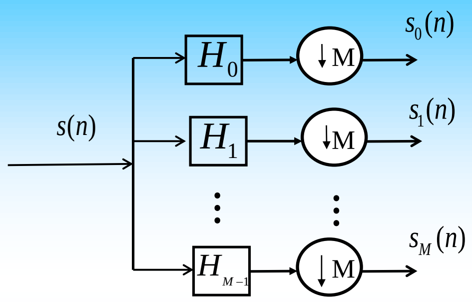
- Decompose signal into frequency bands, followed by down-sampling

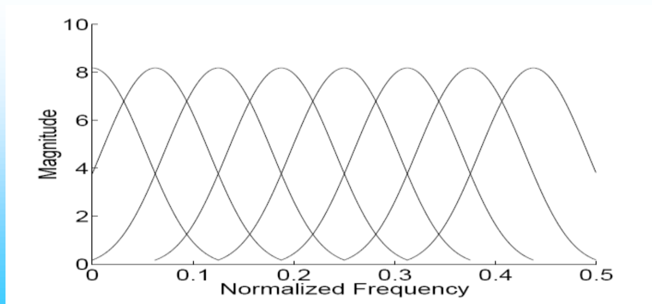
Example: Subband filters with octave bandwidths





Example: subband filters with equal bandwidths





2) DCT and Modified DCT

Conventional DCT:

Forward 1D DCT

$$f(k) = \frac{w_k}{\sqrt{N}} \sum_{n=0}^{N-1} s(n) \cos\frac{(2n+1)\pi k}{2N}, \quad k = 0, \cdots N-1, \; w_k = \begin{cases} 1 & k = 0 \\ \sqrt{2} & k > 0 \end{cases}$$

Set A as transform matrix: $\mathbf{A} = \left[c_k(n)\right]$ $\quad c_k(\mathrm{n}) = \begin{cases} \frac{1}{\sqrt{N}} & k = 0 \\ \sqrt{\frac{2}{N}} \cos\frac{(2n+1)\pi k}{2N} & 0 < k \le N-1 \end{cases}$

→ **DCT (in the vector and matrix form)** $\quad$ **f=As**

Inverse 1D DCT: $\mathbf{s} = \mathbf{A}^{-1}\mathbf{f} = \mathbf{A}^T\mathbf{f}$

↑

(Since DCT is real and orthonomal $\Rightarrow \mathbf{A}^T = \mathbf{A}^{-1}, \mathbf{A}^{\mathbf{T}} = \mathbf{A}^*$

Modified DCT (MDCT)

For each block of data (length of 2N),

$$X_k = \sum_{n=0}^{2N-1} x_n \cos\left[\frac{\pi}{N}\left(n + \frac{1}{2} + \frac{N}{2}\right)\left(k + \frac{1}{2}\right)\right]$$

Data: signal itself, or outputs of a subband filter.

MDCT: a special case of subband filters
(filter kernel length = data block size)

For compression:
remove small value DCT coefficients (set to 0 values)

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3. Quantization and encoding

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Quantization

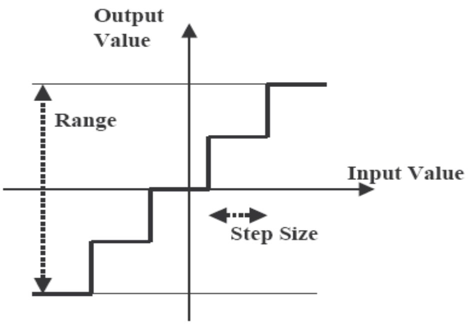
aim: continuous value magnitude → discrete values

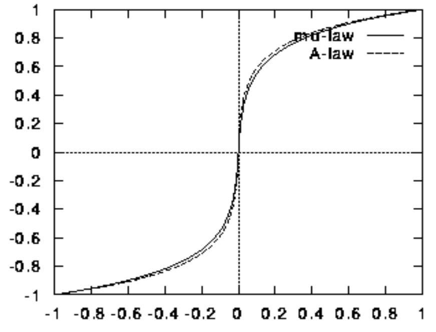
type: scalar /vector quantizer

step size: uniform, logarithm, power …

large step size: high compression,

but low quality (high quantization error)





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Source symbols encoding (lossless)

There are many,

e.g. Huffman coding;

arithmetic coding;

Ziv-Lempel (LZW) coding

…

- Huffman coding: is an entropy-based lossless coding method. Takes advantage of non-uniform distributions of symbols, where different code lengths are given according to the probabilities of symbols.

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4. Psychoacoustic model, and parameters for HAS (Human Auditory System)

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Why Psychoacoustic model ?

- ”Cheat” human ears: lossy compression, but perceptually lossless
- No bits attribute to sound that is non-audible
- No extra bits to sound components than human ears needed.

Main ”features” in HAS:

- Insensitive to phase changes
- Frequency resolution: differ in different frequencies
  HAS: ~ cochlear filters
  imply: different bit allocation to different frequency bands
- Masking effect: within a “critical band”, a stronger tone masks the remaining weaker sounds (making them inaudible)
- Critical band
- Bark scale, Bark frequency

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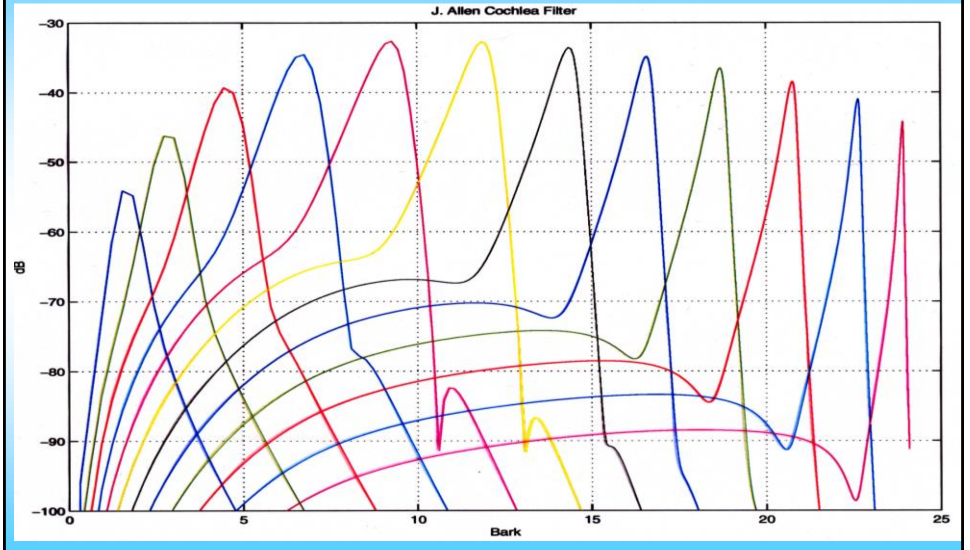
Cochlear filters: HAS modeling



(From J. Allen: Cochlear Filters)

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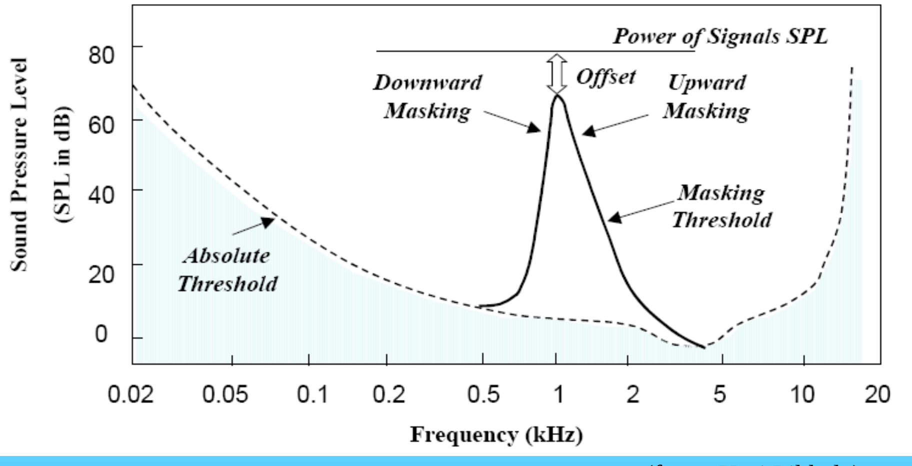
Psychoacoustic modeling: Masking effect of HAS



(from: Kari Pihkala)

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Critical band, Bark scale vs. frequency

- **Critical band**:

  A range of frequencies within which the masking SNR remains a constant.

- **Bark scale:**

  A standardized scale of frequency, where each “Bark” constitutes one critical bandwidth.

  Is approximately equal-bandwidth up to 700Hz, and 1/3 octave above 700Hz.

  A frequency scale, under which the masking phenomenon and shape of cochlear filters are approximately invariant.

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- **Bark frequency:** can be converted from the usual frequency *f* (in Hz)

$$B_f = 13\tan^{-1}\left(\frac{0.76f}{1000}\right) + 3.5\tan^{-1}\left(\left(\frac{f}{7500}\right)^2\right)$$

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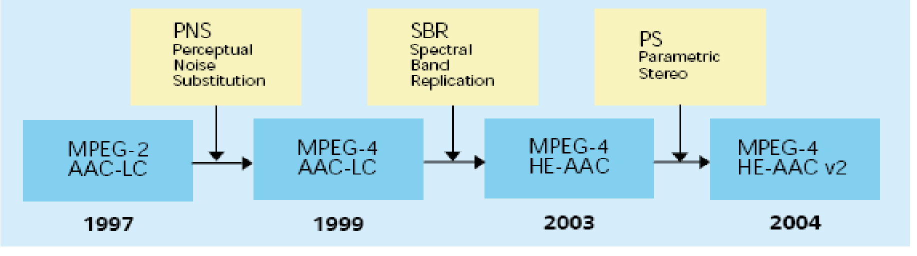
5. Speech/Audio Coding Standards: examples

Audio Coding Standards: brief Introduction

Progress in AAC (Advanced Audio Coding) standards



(From: **http://www.iis.fraunhofer.de/bf/amm/)**

LC: low complicity; HE: high efficiency

HE-AAC v2 (aacPlus v2):
is also part of the 3GPP standard for the delivery of audio content to 3G devices

Varieties in AAC codecs

Advanced Audio Coding's multiple codecs:

- Low Complexity AAC (LC-AAC)
- High-Efficiency AAC (HE-AAC)
- Scalable Sample Rate AAC (AAC-SSR)
- Bit Sliced Arithmetic Coding (BSAC)
- Long Term Predictor (LTP)
- Low Delay AAC (LD-AAC)

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MPEG/ISO audio standards

Standards	Audio sampling rate (kHz)	Compressed bit-rate (kbits/sec)	Channels	Standard Approved
MPEG-1 Layer I	32, 44.1, 48	32 – 448	1-2 channels	1992
MPEG-1 Layer II	32, 44.1, 48	32 – 384	1-2 channels	1992
MPEG-1 Layer III	32, 44.1, 48	32 – 320	1-2 channels	1993
MPEG-2 Layer I	32, 44.1, 48	32 – 448 for two BC channels	1-5.1 channels	1994
	16, 22.05, 24	32 – 256 for two BC channels		
MPEG-2 Layer II	32, 44.1, 48	32 – 384 for two BC channels	1-5.1 channels	1994
	16, 22.05, 24	8 – 160 for two BC channels		
MPEG-2 Layer III	32, 44.1, 48	32 – 384 for two BC channels	1-5.1 channels	1994
	16, 22.05, 24	8 – 160 for two BC channels		
MPEG-2 AAC	8, 11.025, 12, 16, 22.05, 24, 32, 44.1, 48, 64, 88.2, 96	Indicated by a 23-bit unsigned integer	1-48 channels	1997
MPEG-4 T/F coding	8, 11.025, 12, 16, 22.05, 24, 32, 44.1, 48, 64, 88.2, 96	Indicated by a 23-bit unsigned integer	1-48 channels	1999

BC: backward compatibility

Table is from C-M Liu and W-W Chang, '99 in http://www.mp3-tech.org/programmer/docs/AudioCoding.pdf

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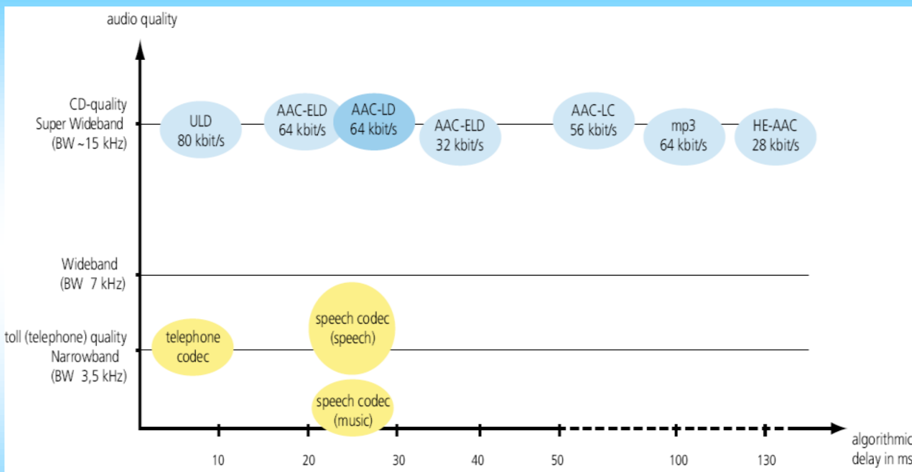
Comparison: quality, bit rate, time delay

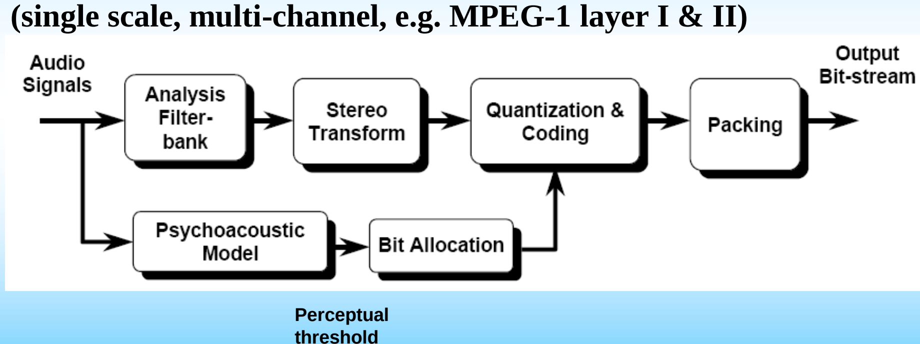


(From: http://www.iis.fraunhofer.de/bf/amm/)

Speech codecs

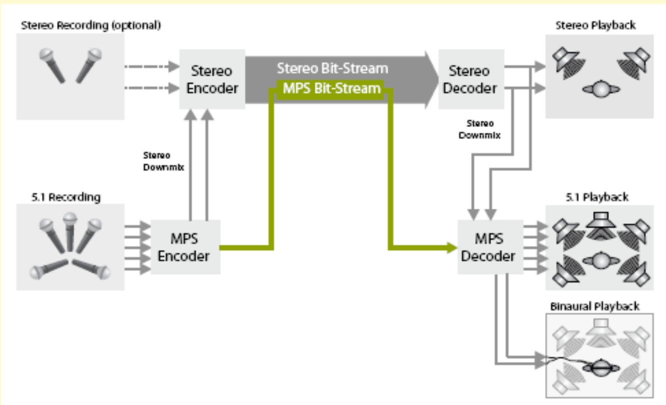
A typical perceptual audio encoder

(single scale, multi-channel, e.g. MPEG-1 layer I & II)



e.g. Use MDCT (modified DCT) as analysis filterbank
(containing 32-band polyphase quadrature filters (PQFs)).

Stereo / 5.1 channel audio:



How MPEG Surround (MPS) works

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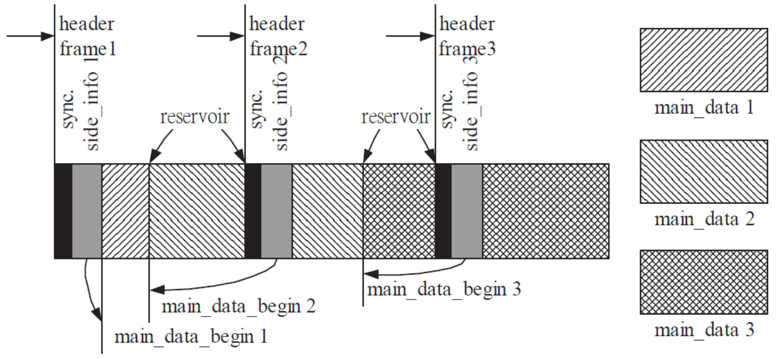
Bit Stream

Header:

bit rate, sample frequency, …

Side information:

block type, Huffman tables, subband gains, …



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Speech/audio coding standards:

Non-parametric audio coding (subband filters):
in MPEG-1, MPEG-2 standards

Parametric audio coding: (CELP-based)
in MPEG-4 standards

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MPEG-2 (part 7) and MPEG-4 (part 3):

use Advanced Audio Coding (AAC) schemes
AAC is a standardized, lossy compression and encoding scheme for digital audio.
AAC has a better quality than *MP3* at the same bite-rate, particularly under 192 kb/s.

MPEG layer 3 (or, MP3):
is the most popular audio coding standard for digital music in the computer and the Internet. MP3 is a part of the MPEG-1 and the MPEG-2 standards.

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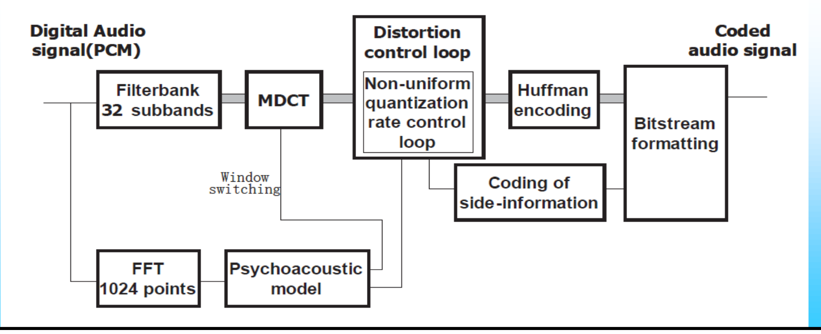
MP3 coding

- Analysis filterbank /Synthesis filterbank
- Midified DCT (MDCT) /Inverse MDCT
- Quantizer/dequantizer
- Huffman encoder/decoder
- Psycho-acoustic model (using masking effect, critical band, …)
- Bitstream

Block diagram of MP3 encoder:



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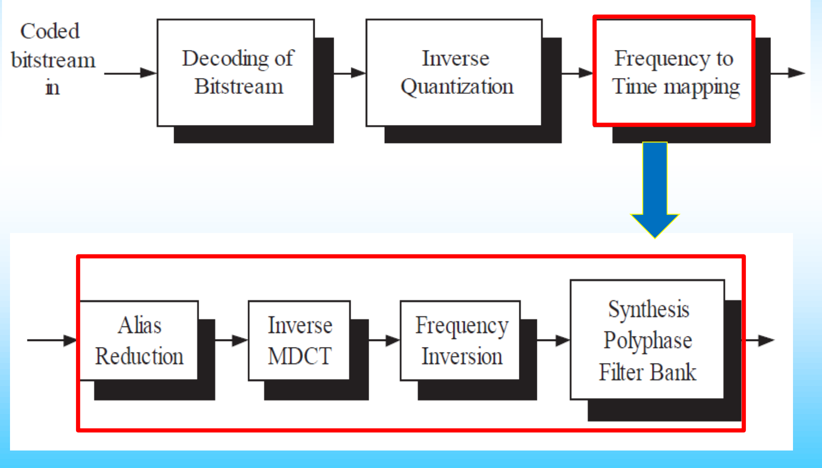
Block diagram of MP3 decoder:



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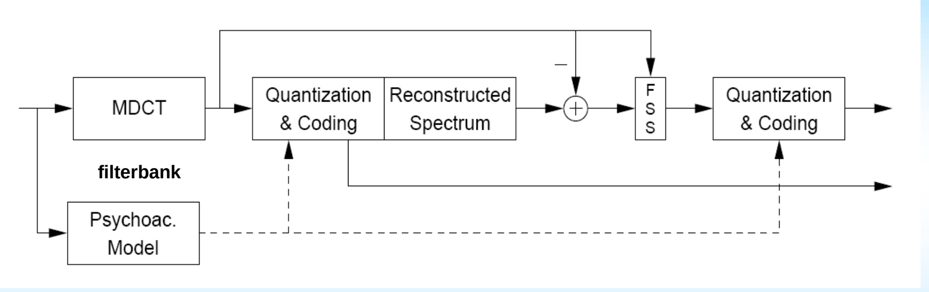
AAC coding in MPEG-2 / MPEG-4



FSS: frequency selective switch

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MDCT and hybrid filterbank

- Modified Discrete Cosine Transform (MDCT)

- Hybrid filterbank (or, subband MDCT)

  4 PQF (polyphase quadrature filter) subbands
  followed by a MDCT

  - fs: [8KHz, 96kHz]:
  - narrow/wideband: 10-40ms frame/10-20ms frame
    ➔ high/low frequency resolution
  - Channels: MPEG-4 up to 48 channels;
    MPEG-1: up to 2 channels;
    MPEG-2: up to 5.1 channels

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6. Objective quality measures for synthetic speech/audio signals

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Quality measures for synthetic audio/speech

+ Human ears are insensitive to phase changes !
  => Criteria based on speech waveform distortion is NOT suitable

+ Compute spectral distortions **in the frequency-domain**
  (e.g. magnitude spectral distortions between original and synthetic ones)

+ Or, compute distortions **in the Bark frequency-domain**:
  Bark / Modified Bark Spectral distortion (BSD/MBSD)

$$MBSD = \frac{1}{N}\sum_{j=1}^{N}\left[\sum_{i=1}^{K} M(i)\left|L_x^{(j)}(i) - L_y^{(j)}(i)\right|^{n}\right]$$

M(i): Perceptible distortion in i-th critical band

$L_x^j(i)$ **: Bark spectrum of j-th frame of coded speech**

+ Perceptual speech quality measures
  (e.g. ITU-T recommendation P.861)

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7. About laboratory project-1

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Lab.-1

Tasks: speech model: analysis, synthesis and compression (dead line: 2020-04-17, 23:55)

1. Record a (stationary) single vowel, and make Matlab programs for LPC analysis and synthesis of stationary (single-tone) speech;
2. Record a (nonstationary) speech sentence, and make Matlab programs for block-based LPC analysis and synthesis of nonstationary speech (using the residual sequence as the excitations);
3. Repeat the task 2, however, excitations to the filter are replaced by using a few prominent residuals (<20) in each block;
4. For the recorded speech sentence, determine whether a speech frame (block) is voiced or unvoiced. For those voiced frames, estimate the pitch periods either from the cepstrum.
5. Objective measures of synthetic speech quality

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What one shall do in Lab.1:

Speech modeling, analysis, synthesis and compression

- Record and save a sound or speech file to a computer, and then load the speech file in Matlab.
- Make a Matlab program on LPC (Linear predictive coding) analysis and then synthesis of single tone sound (stationary) and a speech sentence (nonstationary). Listen to the resulting sound.
  From this, you can learn how speech compression is achieved:
  a 10ms of speech signal only requires less than 20 parameters to characterize.
- Model the vocal cord excitations of speech by some impulses or white noise, to the LPC model, and listen to the synthetic speech.
- Estimate the pitch period using the cepstrum method.

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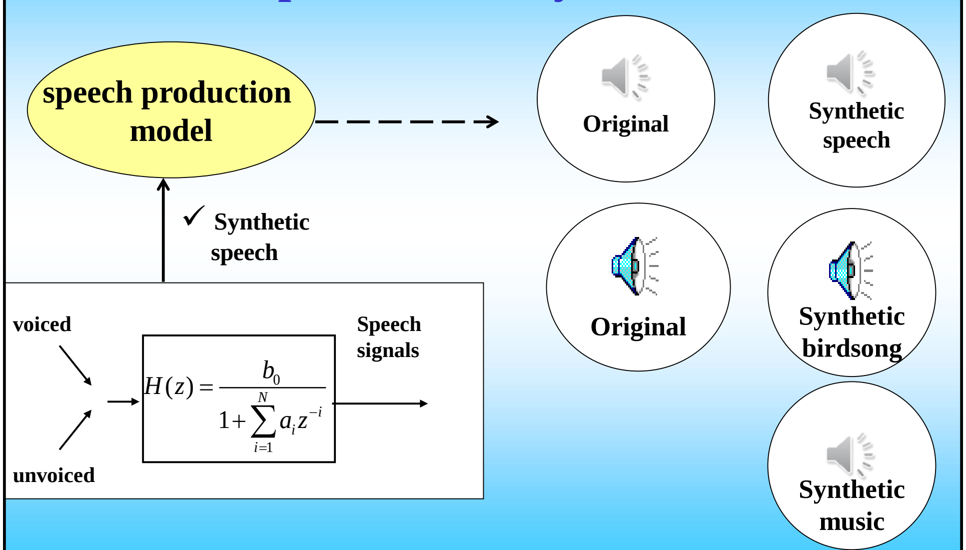
Example of what you may achieve: speech/audio synthesis



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8. References

[1] Lawrence R. Rabiner, Ronald W. Schafer, Digital Processing of speech signals, Prentice-Hall, Inc., 1978.

[2] John R., Jr. Deller, John H.L. Hansen, John G. Proakis, Discrete-time processing of speech signals, IEEE Press Classic Reissue, 1999.

[3] Alan V. Oppenheim, Ronald W. Schafer, and John R. Buck, Discrete-Time Signal Processing, 2nd edition, Prentice Hall, Inc. 1999.

[4] Wikipedia, the free encyclopedia on CELP: http://en.wikipedia.org/wiki/Code_Excited_Linear_Prediction

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