2.4 Audio Compression

2.4.1 Pulse Code Modulation

Audio signals are analog waves. The acoustic perception is determined by the **frequency** (pitch) and the **amplitude** (loudness).

For storage, processing and transmission in the computer audio signals must converted into a digital representation. The classical way to do that is called **pulse code modulation** (PCM). It consists of three steps: sampling, quantization and coding.

Sampling

The analog signal is sampled periodically. At each sampling interval the analog value of the signal (e.g., the voltage level) is recorded as a real number.

After sampling the signal is no longer continuous but discrete in the temporal dimension.

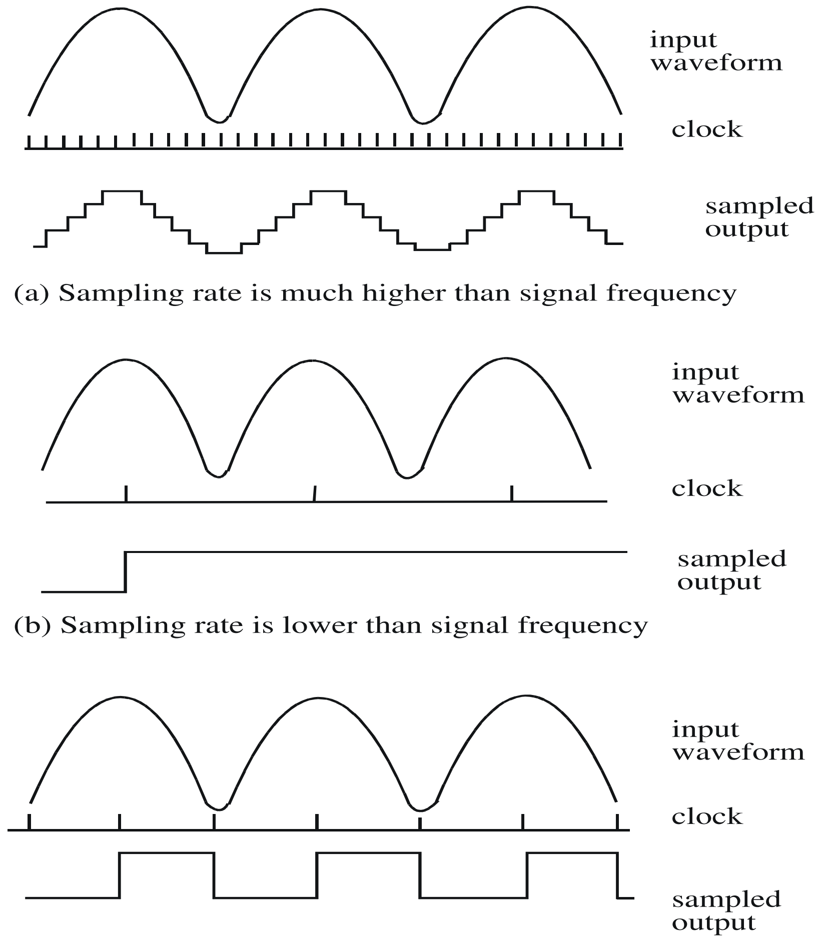
Sampling Theorem of Nyquist

In order to reconstruct the original analog signal without loss we obviously need a minimum sampling frequency. The minimum sampling frequency f_A is given by the sampling theorem of Nyquist (1924):

For noise-free channels the sampling frequency f_A must be twice as high as the highest frequency occurring in the signal:

$$f_A = 2\ f_S$$

Example: Sampling a Signal at Different Rates



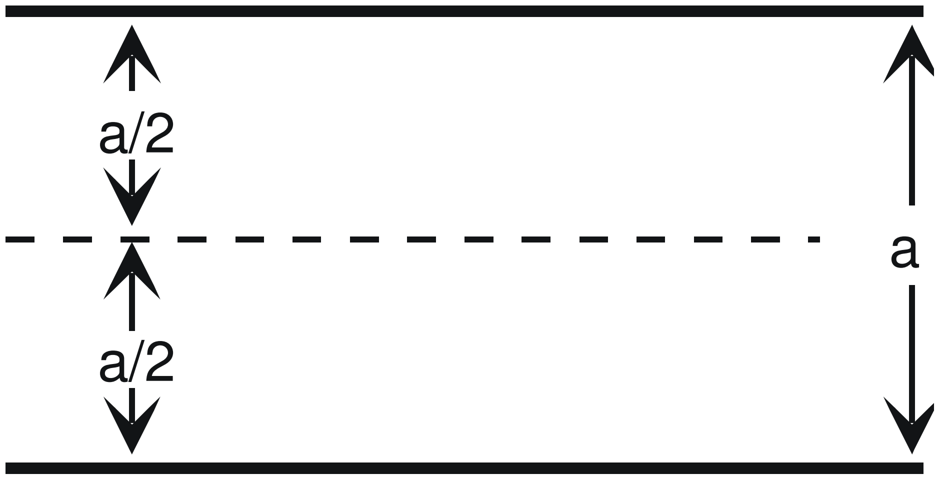
(a) Sampling rate is much higher than signal frequency

(b) Sampling rate is lower than signal frequency

(c) Sampling rate is at the Nyquist limit

Quantization

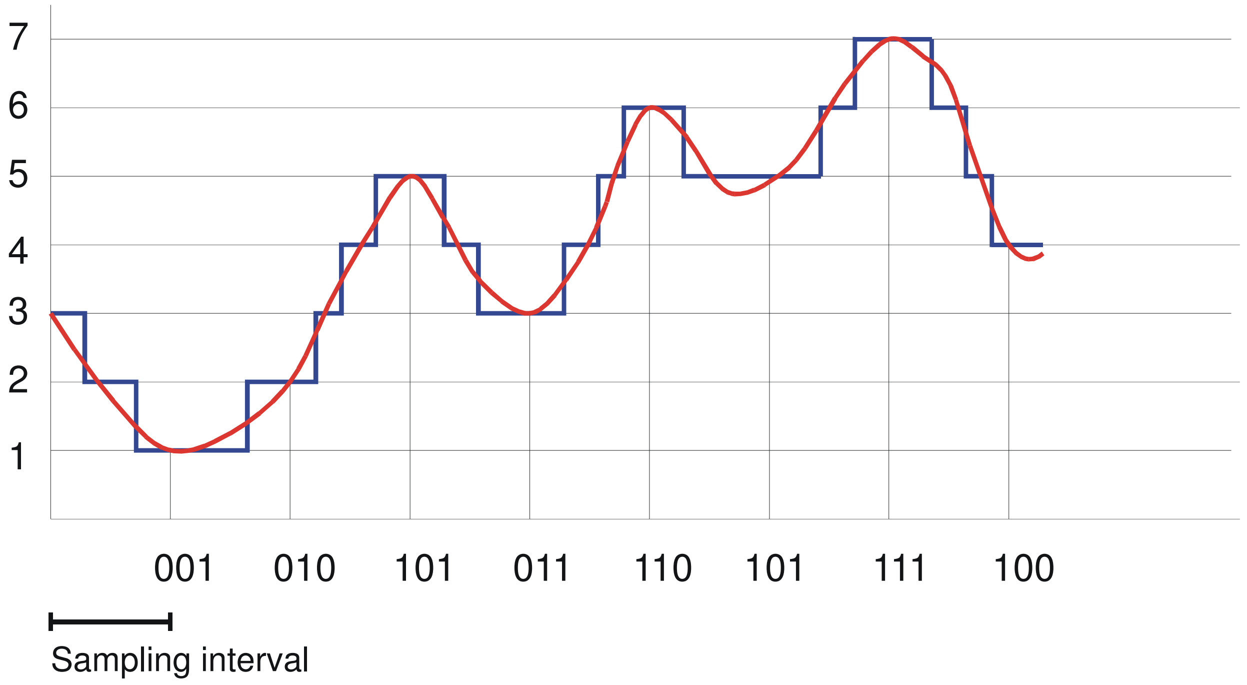
The range of values occurring in the analog signal is subdivided into a fixed number of discrete intervals. Since all analog values contained in an interval will be mapped to the same interval number we introduce a quantization error. If the size of the quantization interval is *a* then the maximum quantization error is *a*/2.



Binary Coding

We now have to determine a unique binary representation for each quantization interval. In principle any binary code does the job. The simplest code (which is in fact often used in practice) is to encode each interval with a fixed-size binary number.

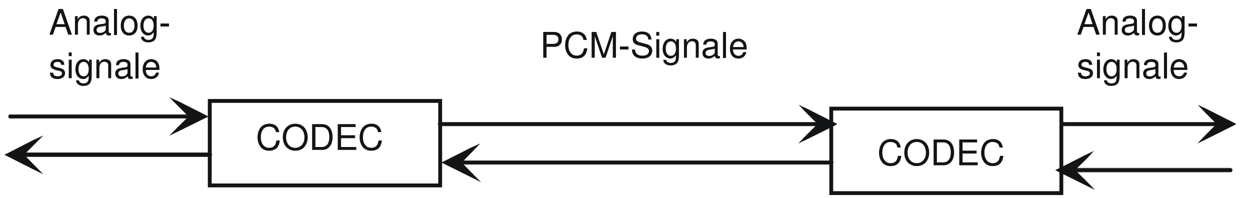
PCM: The Complete Process



The combination of the steps sampling, quantization and binary coding is called **Pulse Code Modulation (PCM).**

CODECs

The devices performing A/D conversion and D/A conversion are called CODECs (Coders/Decoders).



Note: A **modem** is used to transmit digital signals over analog links, a **codec** is used to transmit analog signals over digital links.

PCM Telephone Channel

Sampling Rate

Starting point: an analog CCITT telephone channel

Frequency range: 300 – 3400 Hz, i.e., audio bandwidth: 3100 Hz (sufficient for speech)

Sampling frequency: **$f_A = 8$ kHz**

Sampling period: **$T_A = 1/ f_A = 1/8000$ Hz**
$= 125$ µs

The sampling frequency chosen by CCITT is higher than the Nyquist limit: for a maximum frequency of 3400 Hz in the signal a sampling frequency of 6800 Hz would be sufficient. This has technical reasons (noise, influence of filters, channel separation, etc)

Quantization of the Amplitude

The minimum number of quantization intervals is determined by the understandability of speech at the receiver. Based on experimental experience CCITT has chosen 256 quantization intervals.

With standard binary coding we thus need 8 bits per sample.

Bit Rate of the PCM Channel

We conclude that the bit rate of a standard PCM channel is

8 bits * 8000/s = 64 kbit/s

Non-Linear Quantization

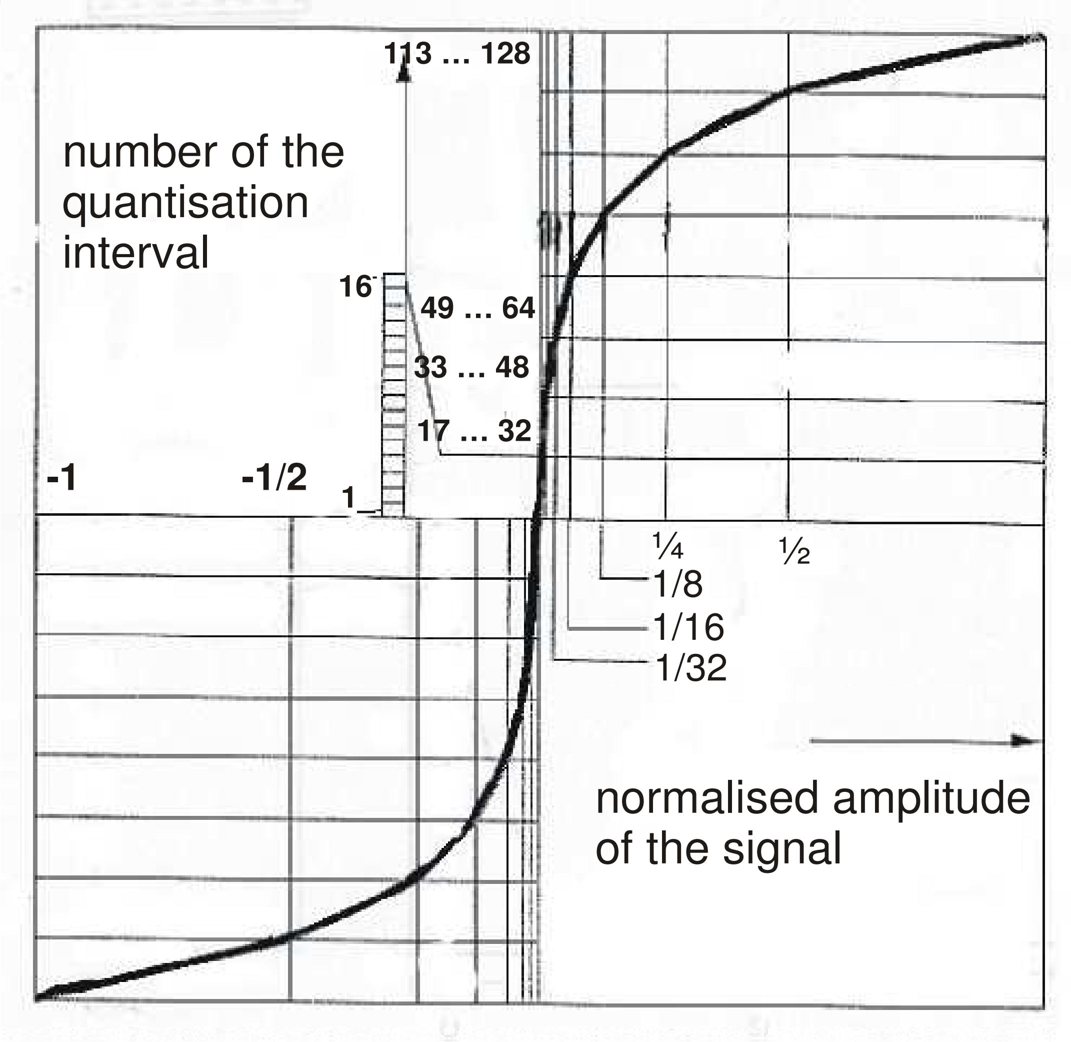
With linear quantization all intervals have the same size, they do not depend on the amplitude of the signal. However it would be desirable to have a smaller amount of quantization noise at small amplitude levels because quantization noise is more disturbing in “quiet times“.

This goal can be reached with non-linear quantization. We simply chose larger quantization intervals at higher amplitude values.

Technically this can be done by a “signal compressor“ which preceeds the coding step. At the receiver side an expander is used to reconstruct the original dynamics.

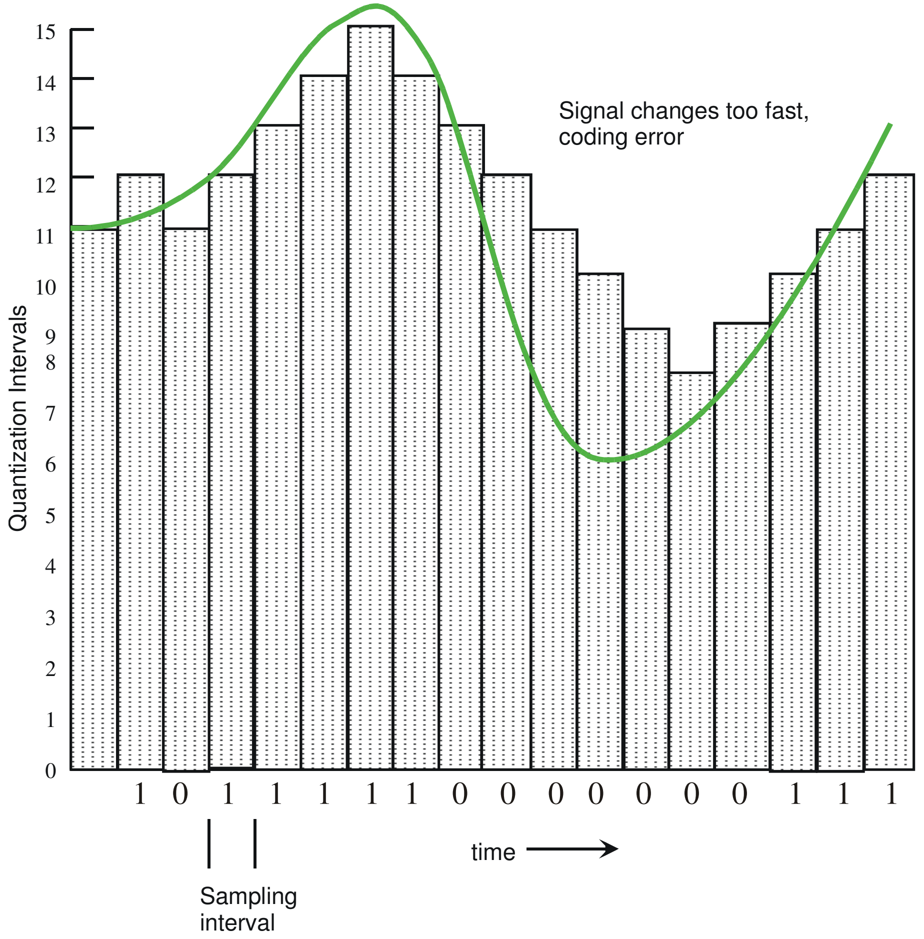
Many compressors use a logarithmic mapping. In digital electronics this is often approximated by a piecewise-linear curve. The 13-segment compressor curve is a typical example.

13-Segment Compressor Curve



Delta Modulation

Instead of coding the absolute values of the amplitude, the difference to the value in the previous interval is coded **in one bit**. Only steps of +1 or –1 are possible.



Coding: 1 = increasing signal

0 = decreasing signal

Differential PCM (DPCM)

In differential PCM we encode the actual difference between the signal values in two adjacent intervals with a small number of bits. This leads to a bit rate and precision between that of encoding the absolute values and delta modulation.

Adaptive DPCM (ADPCM)

The dynamics in real audio signals are often such that we have quiet periods and loud periods. In quiet periods (i.e., periods with low variance of the amplitude) we can encode the signal with fewer bits than in loud periods. This is called **Adaptive Pulse Code Modulation** (ADPCM).

For example, ADPCM allows us to compress a HiFi stereo audio signal from 1.4 Mbit/s to 0.2 Mbit/s without loss of quality.

Well-known ADPCM algorithms are μ-law and A-law.

Typical Sampling and Quantization Parameters

Sampling Rate

8 kHz	telephony, µ-law encoding, SUN Audio
32 kHz	Digital Radio Broadcast
44,1 kHz	Audio-CD
48 kHz	Digital Audio Tape (DAT)

Quantization

8 bits	256 amplitude levels: speech
16 bits	65536 amplitude levels: HiFi music

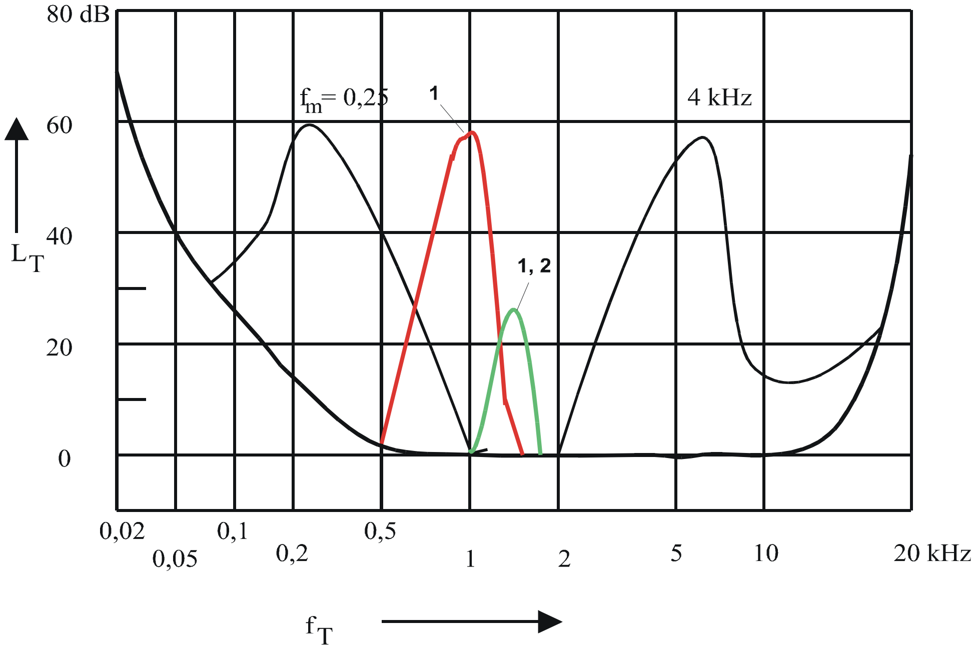
2.4.2 Audio Compression with Psycho-Acoustic Models

Compression based on semantic irrelevance

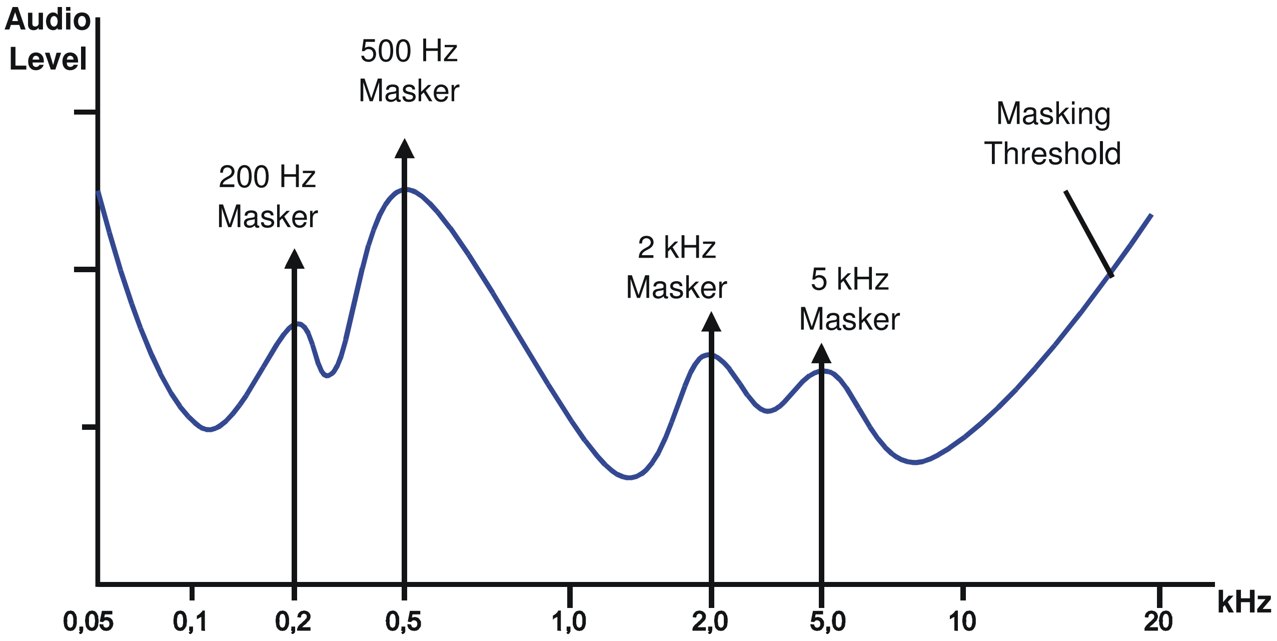
We remove those parts of the signal at the source that the receiver will not be able to hear anyway.

Example: The Masking Effect

A high-amplitude signal masks out a low-amplitude signal at an adjacent frequency



Psycho-Acoustic Models



Example: MPEG Audio

Characteristics

Compression to 32, 64, 96, 128 or 192 kbit/s

Audio channels

- Mono or
- Two independent channels or
- "Joint Stereo"

Techniques

- Sampling rates: 32 kHz, 44,1 kHz or 48 kHz
- 16 bits per sample
- Maximum encoding and decoding delay: 80 ms at 128 kbit/s

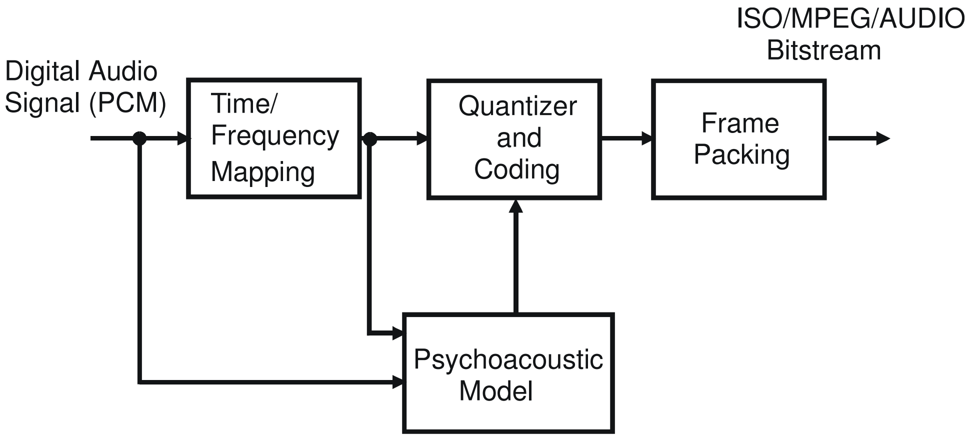
A psycho-acoustic model controls the quantization.

Two Techniques in MPEG-1 Audio

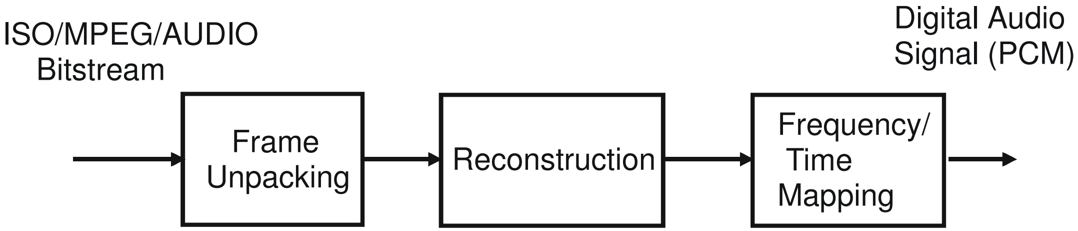
MUSICAM	ASPEC
Masking Pattern Universal Sub-band Integrated Coding and Multiplexing	Advanced Spectral Entropy Coding
Institut für Rundfunktechnik, München	FhG Erlangen
Sub-band Coding	Overlapping dynamic frequency bands, entropy coding (Huffman)
Simple and easy to implement	Very good quality at very low bit rates

MPEG Audio Encoder and Decoder

Encoder



Decoder



Three Layers in MPEG Audio

1. Sub-band coding with 32 bands with the **MUSICAM** technique
   - High data rate
   - mono, stereo, 48 kHz, 44.1 kHz, 32 kHz
2. Sub-band coding with **MUSICAM**, more complex psycho-acoustic model
   - Intermediate
   - Better sound quality at low bit rates
3. Transformation-based compression with the **ASPEC** technique
   - Lowest data rate
   - Stereo Audio in CD quality at less than 128 kbit/s!
   - Mono Audio in telephone quality at 8 kbit/s

MPEG audio layer three, encoded with ASPEC, is also called **MP3 (!)**

MP3 – History (1)

As early as 1987 the Fraunhofer Institut für Integrierte Schaltkreise (Institute of Integrated Circuits) in Erlangen (Germany) began with the development of audio compression techniques that took the specific properties of the human perceptual system into account. Their technique was included into the MPEG Audio standard of ISO (IS-11172-3 and IS 13818-3) as MPEG Audio Layer 3 (MP3). This is currently the strongest audio compression standard in use.

The original goal was a reduction of the data rate by a factor of 12 compared to an audio CD, without an audible difference.

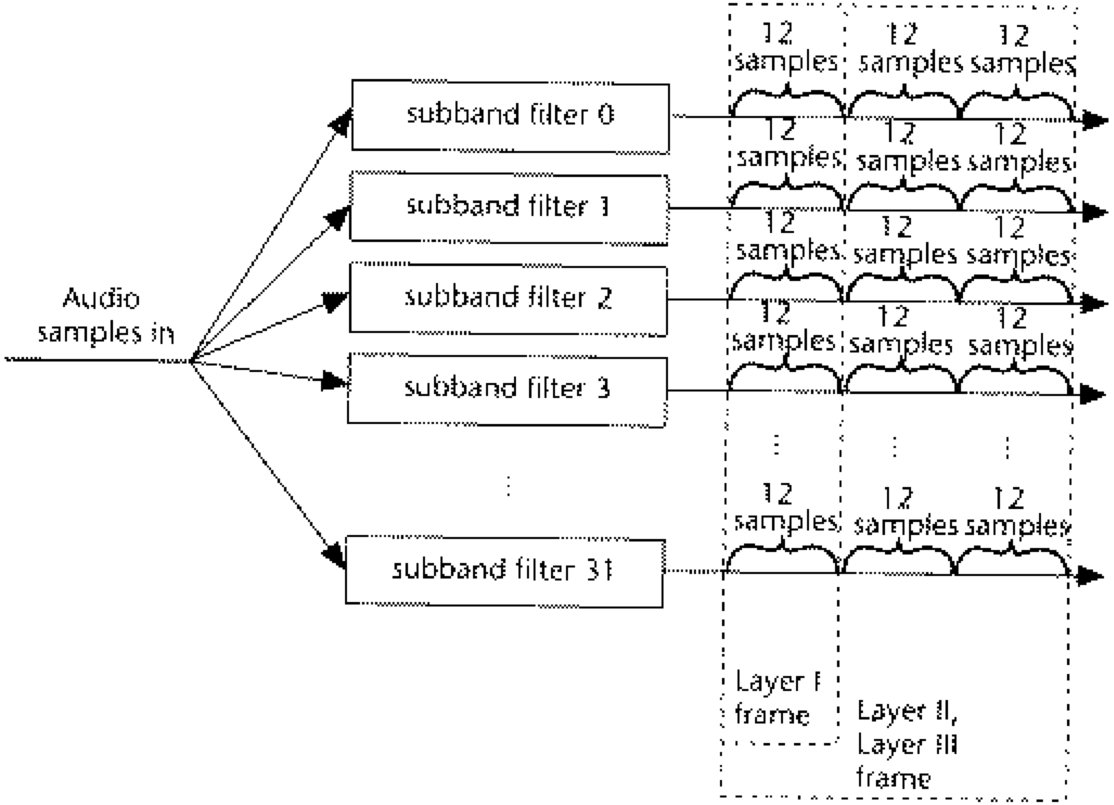
MP3 – History (2)

As usual, ISO only standardizes the technical parameters and the decoder. The inner workings of the encoder remain unspecified. This gives developers significant freedom to develop specific encoding techniques, and even get patents for their encoding algorithms.

As a consequence we know very little about the exact implementation of the MP3 encoder written by the Fraunhofer Institute. Exact details on their psycho-acoustic model are not published. The Fraunhofer Institute also holds a patent on its optimized encoding mechanism for MP3.

MPEG Audio Layers (1)

MP3 subdivides the data stream into **frames**. Each frame corresponds to the audio signal in a certain time period. It contains 384 samples. The samples represent values out of 32 frequency sub-bands. There are 12 values from each sub-band.



MPEG Audio Layers (2)

Layer 1

Frequency masking: Usage of a DCT-based filter. At any given time the algorithm only considers one frame. The frequences occurring in this frame are subdivided into the frequency bands and then filtered.

Layer 2

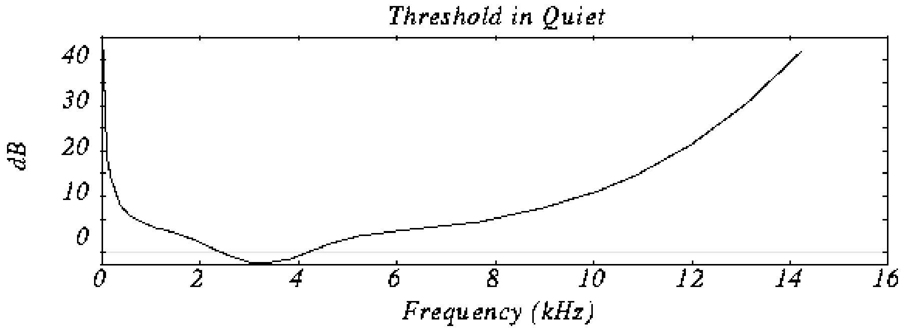
Temporal masking: At any given time the algorithm looks at three adjacent frames, the current, the previous and the next frame. This allows to take advantage of temporal masking effects as perceived by the human ear.

Layer 3

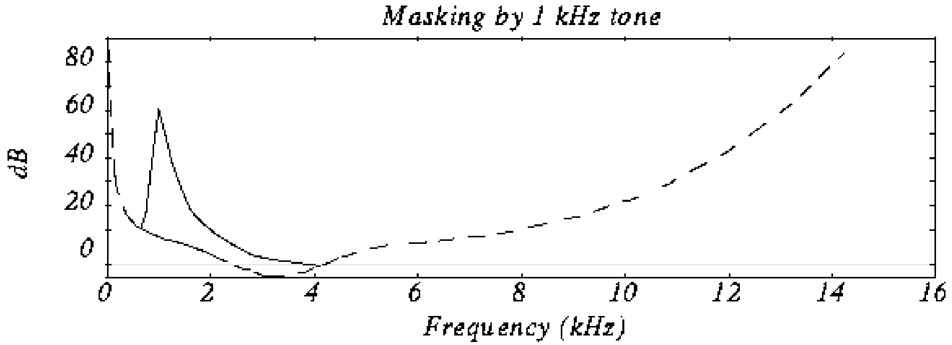
Non-linear masking: The frequencies are subdivided into bands of different widths. Also, stereo channels are encoded differentially, i.e., the difference between the two channels rather than the absolute values are encoded. The last step is a Huffman coding of the coefficients.

Layer 1: Psycho-Acoustic Effect

1. Sensitivity of the human ear



2. The frequency masking effect

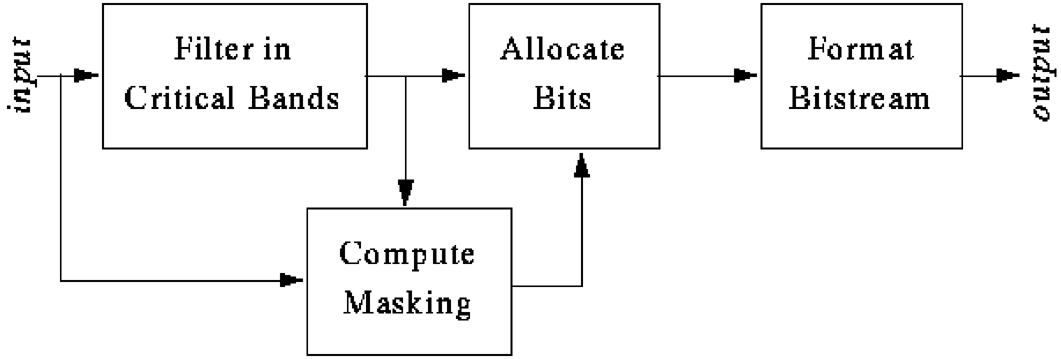


Experiment: Play a tone of 1 kHz (the masking tone) at a certain amplitude (e.g., 60 dB). Then add a test tone (e.g., of 1.1 kHz) and increase its amplitude until the the test tone is heard. This will happen at a much higher amplitude than in the quiet.

Layer 1: Compression

Apply a sub-band filter to subdivide the signal into 32 bands (“critical bands“).

For each band, define a masking curve that indicates at which level the signal will be masked by adjacent bands.



Algorithm:

- Compute the energy in each band.
- If the energy in a band is smaller than the masking threshold of a neighboring band, do not encode the band.
- Otherwise encode the band. Quantize the coefficients with a quantization factor. Choose the factor so that the quantization error is smaller than the masking factor (1 bit in the quantization corresponds to a noise of 6 dB).

Layer 1: Example

The table shows the levels of the first 16 out of the 32 bands.

Band	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16
Level	0	8	12	10	6	2	10	60	35	20	15	2	3	5	3	1

The level of band 8 is 60 dB. We assume that is has a masking threshold of 12 dB for band 7 and of 15 dB for band 9.

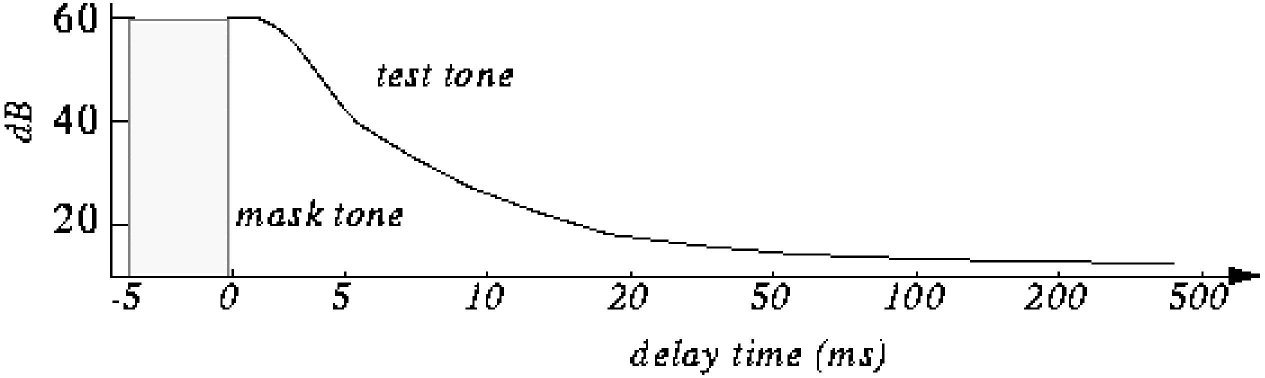
The level of band 7 is 10 dB (< 12 dB), thus we ignore it.

The level of band 9 is 35 dB (> 15 dB), thus encode it. Choose the quantization factor so that the quantization error will be less than 2 bits (12 dB).

Layer 2: Psycho-Acoustic Effect

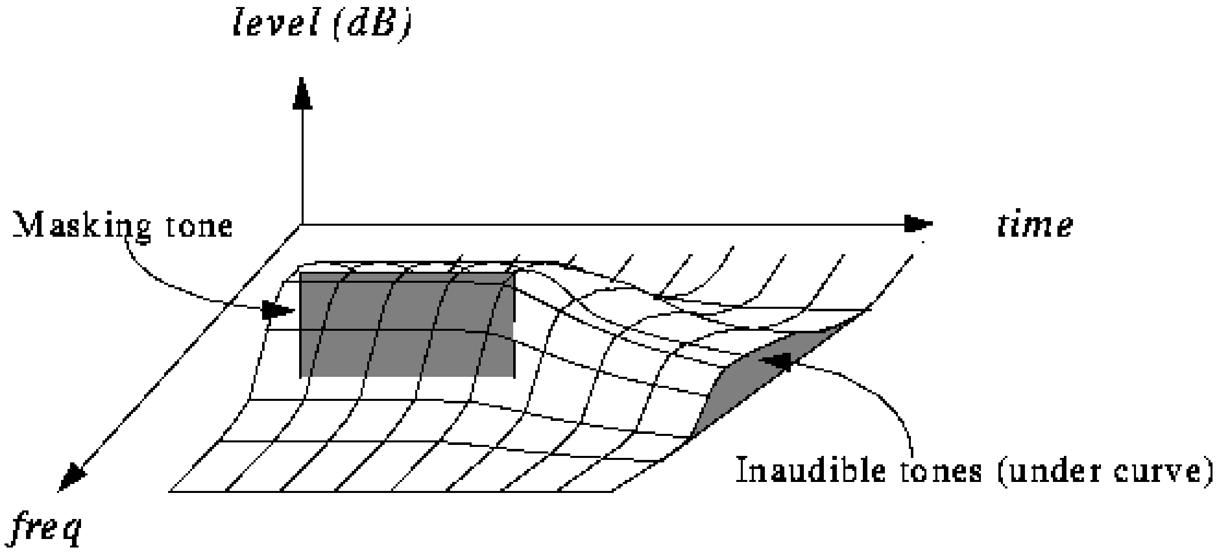
Temporal masking: When we hear a loud sound that suddenly stops, it takes a while until we can hear soft sounds again.

Experiment: Play a masking tone of 1 kHz at 60 dB and a 1.1 kHz test tone at 40 dB (the test tone is not heard, it is masked). Stop the masking tone and after a short delay also the test tone. Vary the delay to find the time threshold at which the test tone can just be heard.



Layer 2: Compression

Repeat the experiment with other test tones:



In a way similar to layer 1, we take advantage of this temporal phenomenon to mask out sub-bands, this time those of **adjacent** frames.

For simplification we assume that a sub-band can mask out its neighbors only in one preceeding and one succeeding frame.

Layer 3: Psycho-Acoustic Effect

The contrast resolution of the human ear decreases with the frequency of the signal.

In layers 1 and 2 the frequency spectrum is subdivided into 32 critical bands of identical size. In layer 3, the frequencies are distributed in a non-linear fashion, in a way so that all bands contribute equally to the perception by the ear.

The “Bark“

We introduce a new unit: the **Bark** (named after Barkhausen)

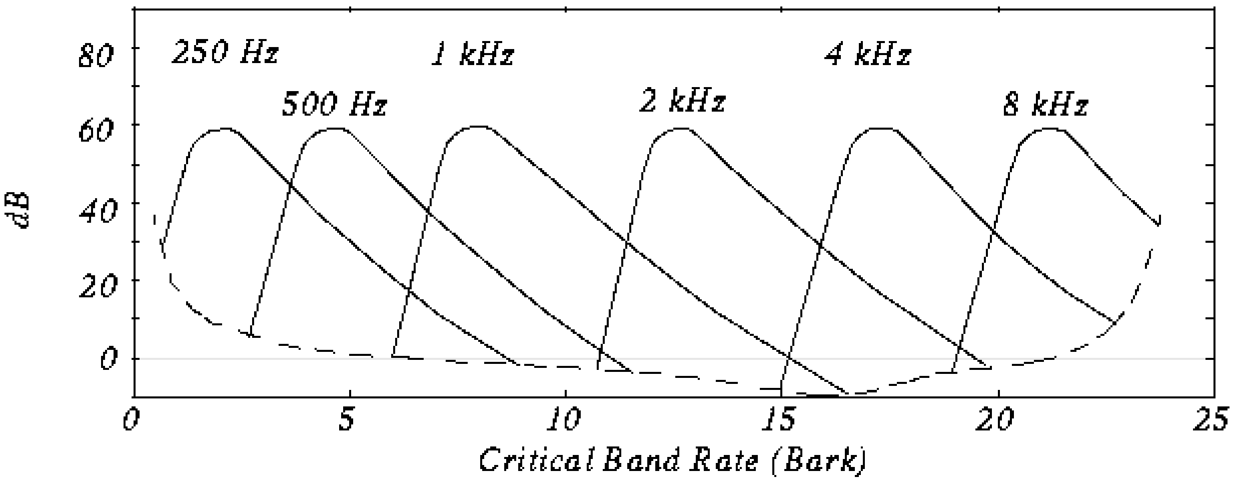
1 Bark = width of a critical band.

For frequencies < 500 Hz: 1 Bark = f/100.

For frequencies > 500 Hz: 1 Bark = 9+4 log(f/1000)

Layer 3: Compression

Masking Thresholds on critical band scale:



Layer 3 comes closer to human perception by choosing a more appropriate definition of the sub-bands, based on the Bark.

In addition to frequency masking and temporal masking, as in layers 1 and 2, layer 3 also introduces the differential coding of stereo signals, as well as an entropy encoding of the coefficients based on the Huffman code.

Performance of MP3

Layer	Target bitrate	Ratio	Quality at 64 kbit/s	Quality at 128 kbit/s	Theor. min delay
Layer 1	192 kbit/s	4:1	---	---	19 ms
Layer 2	128 kbit/s	6:1	2.1-2.6	4+	35 ms
Layer 3	64 kbit/s	12:1	3.6-3.8	4+	59 ms

Quality measure:

5 = perfect, 4 = just noticeable, 3 = slightly annoying, 2 = annoying, 1 = very annoying

Real delay is about three times the theoretical delay.

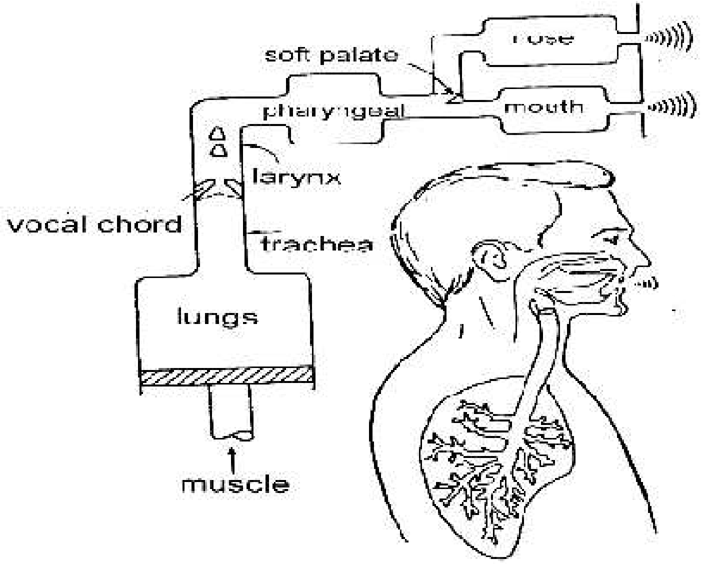
2.4.3 Speech Coding

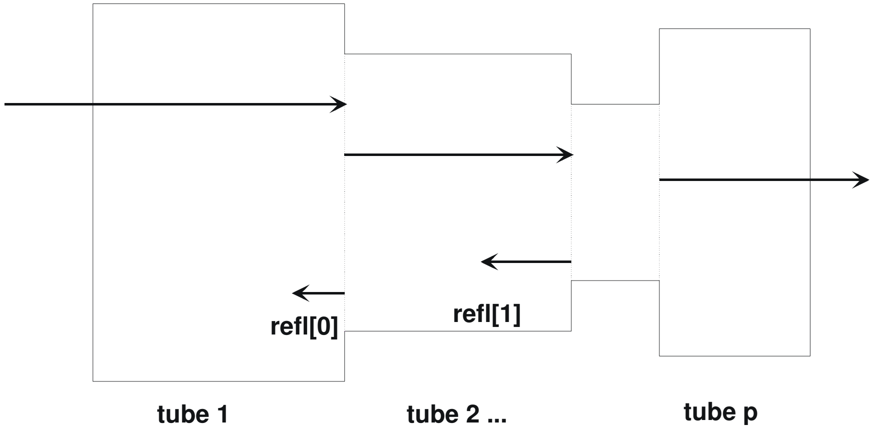
Special codecs optimized for the human voice can reach a very high speech quality at very low data rates.They operate at the normal range of the voice, i.e. at 300 – 3400 Hz.

Such special codecs are most often based on **Linear Predictive Coding (LPC).**

Linear Predictive Coding (1)

LPC models the anatomy of the human voice organs as a system of connected tubes of different diameters.





Linear Predictive Coding (2)

Acoustic waves are produced by the vocal cords, flow through a system of tubes, are partially reflected at the transitions and interfere with the following waves.

The reflection rate at each transition is modeled by the reflection coefficient refl[0], ..., refl[p-1].

We can thus characterize the (speaker-dependent) production of the voice signal with a very small number of parameters.

LPC Encoder

The LPC Algorithm

- The audio signal is decomposed into small frames of fixed length (20 – 30 ms). For each frame s[i] we compute p weights lpc[0], .. , lpc[p-1] so that s[i] is approximated by

  lpc[0] * s[i-1] + lpc[1] * s[i-2] + ... + lpc[p-1] * s[i-p]

  Popular values for p are 8 or 14.
- A synthetically generated source signal is used as input to the model. The generated source can be switched between two modes: voiced (for vowels) and noise (voiceless, for consonants).
- The differences between the synthetically generated signal and the real voice signal during the frame are detected and used to re-calculate the prediction coefficients lpc[i].
- For each frame the mode of excitation (voiced or voiceless) and the current values of the parameters are encoded and transmitted.

LPC Variations

- **CELP** (Code Excited Linear Prediction): We not only distinguish “voiced“ and “voiceless“ but many more types of excitation. These are pre-defined by the developers and stored in the form of a “codebook“. For each frame we transmit the index into the codebook and the lpc parameters.
- **ACELP:** like CELP, but with an adaptive codebook

LPC Examples

G.723.1

Adaptive CELP Encoder (Code Excited Linear Predictor).

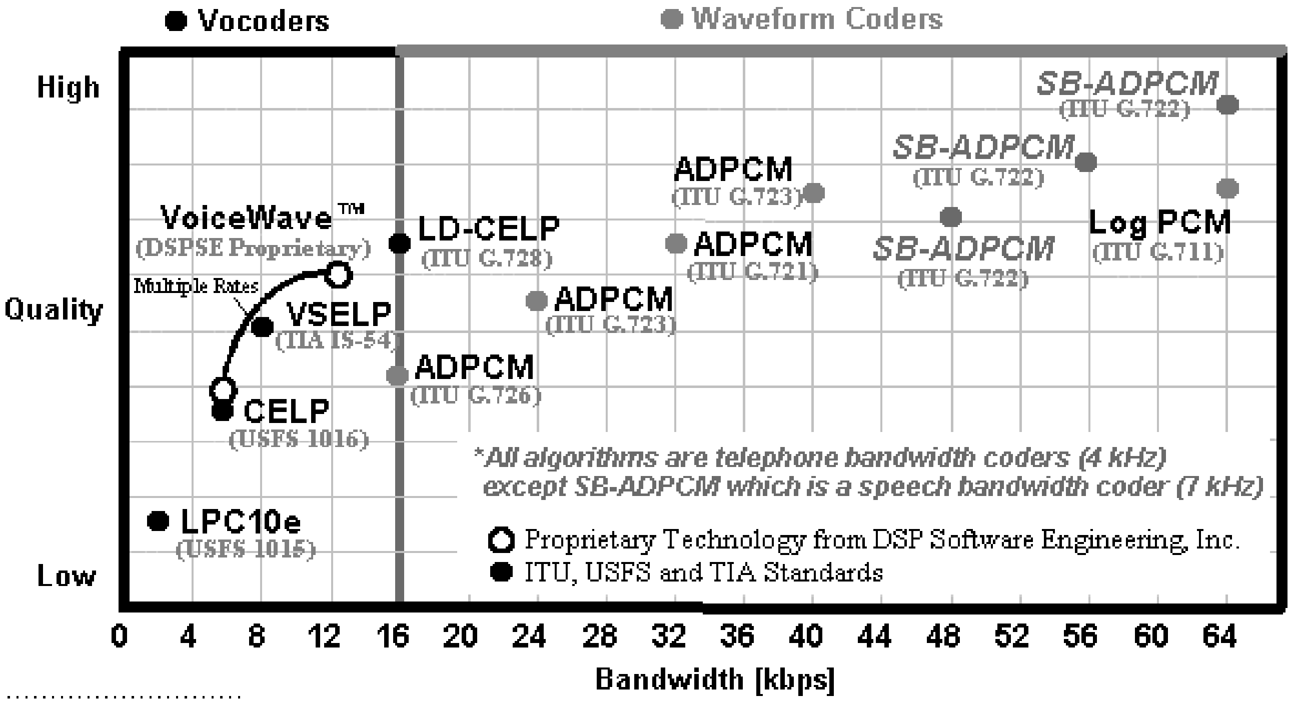
Bit rate for G.723.1: 5,3 kbit/s - 6.3 kbit/s

GSM 06.10

Regular Pulse Excitation – Long Term Prediction (RPE-LTP)

- LPC encoding
- The synthetically generated signal is based on earlier signal values.
- Bit rate for GSM 06.10: 13.2 kbit/s

Specialized Speech Coding vs. PCM Coding



ITU-T Standards for Speech Coding

A selection from the G.7xx-Standards:

- **G.711**: 64 kbit/s (GSTN telephony, H.323 and H.320 Videoconferencing)
- **G.728** LD-CELP: 16 kbit/s (GSM telephony, H.320 Videoconferencing)
- **G.729** ACELP: 8 kbit/s (GSM telephony, H.324 Vi-deo-telephony)
- **G.723.1** MPE/ACELP 5.3 kbit/s bis 6.3 kbit/s (GSTN Video-telephony, H.323 telephony)