8.5 Basic Parameters for Audio Analysis

Physical audio signal: simple

- one-dimensional
- amplitude = loudness
- frequency = pitch

Psycho-acoustic features: complex

- A real-life tone arises from a complex superposition of various frequencies.
- For human audible perception, the emerging and fading away of a tone are very important.

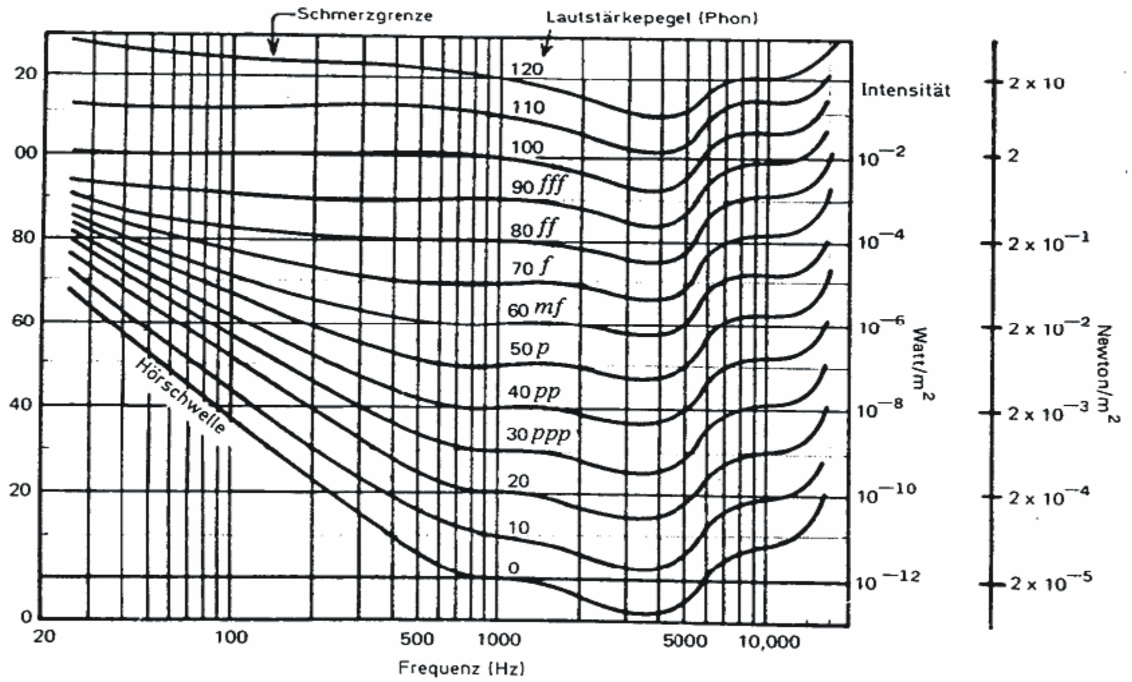
For example, both features are different for a violin and a piano.

Perception of Loudness

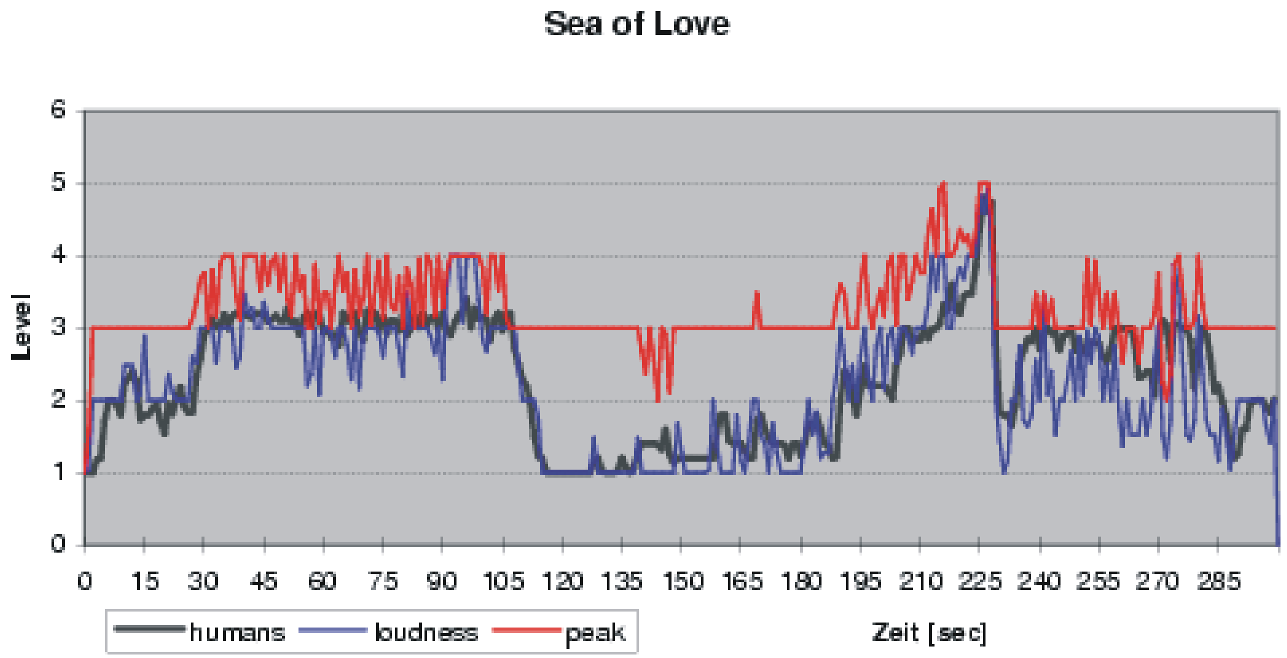
The physical measure is called **acoustic pressure**, the unit is **decibel** [dB-SPL, Sound Pressure Level].

The human audible perception is called **loudness**, the unit is **phon**.

We can empirically derive a set of curves that depicts the perceived loudness as a function of acoustic pressure and frequency. They are called **isophones**.



Experimental Results

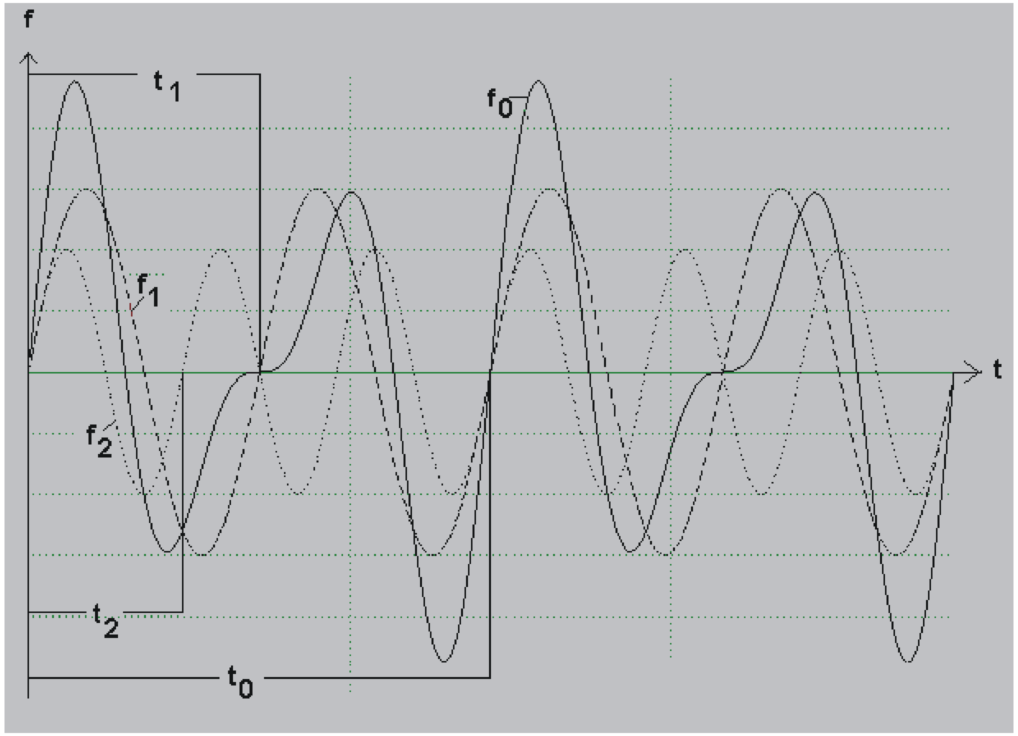


red curve: acoustic pressure

black curve: loudness as perceived by test subjects

blue curve: computationally predicted perceived loudness

Fundamental Frequencies in Harmonic Sounds



The period of the composite tone f_0 corresponds to the least common multiple of the periods of the two composing frequencies f_1 and f_2.

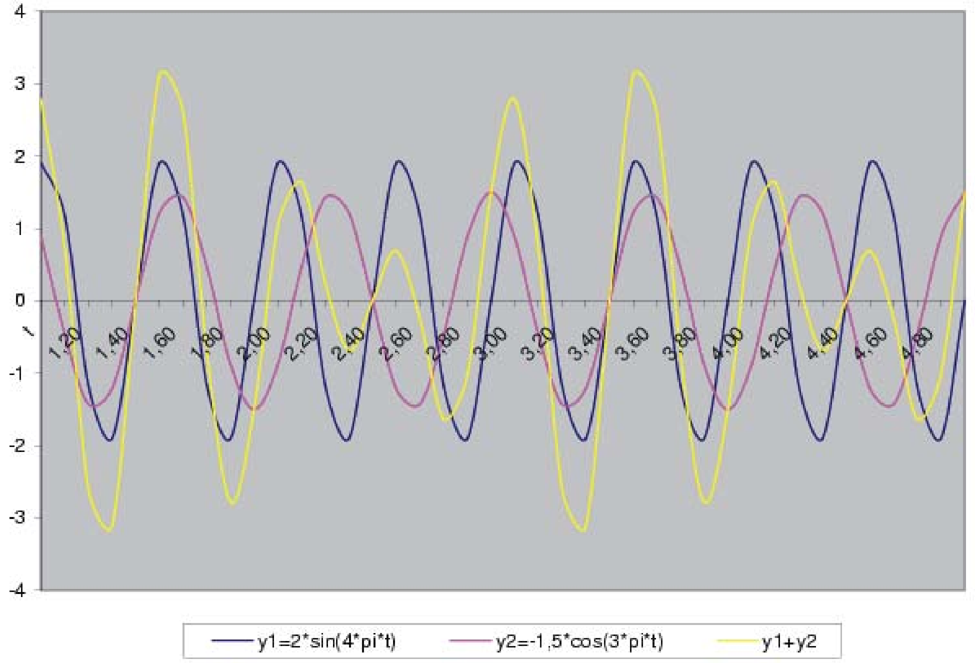
Frequency Transformations

J.B.J. Fourier (1768-1830): Each periodic oscillation can be written as the sum of harmonic frequencies:

$$s(t) = \frac{B_0}{2} + \sum_{f=1}^{N-1} [A_f \sin(\frac{2\pi ft}{N}) + B_f \cos(\frac{2\pi ft}{N})]$$

f: frequency

A_f, B_f: amplitudes

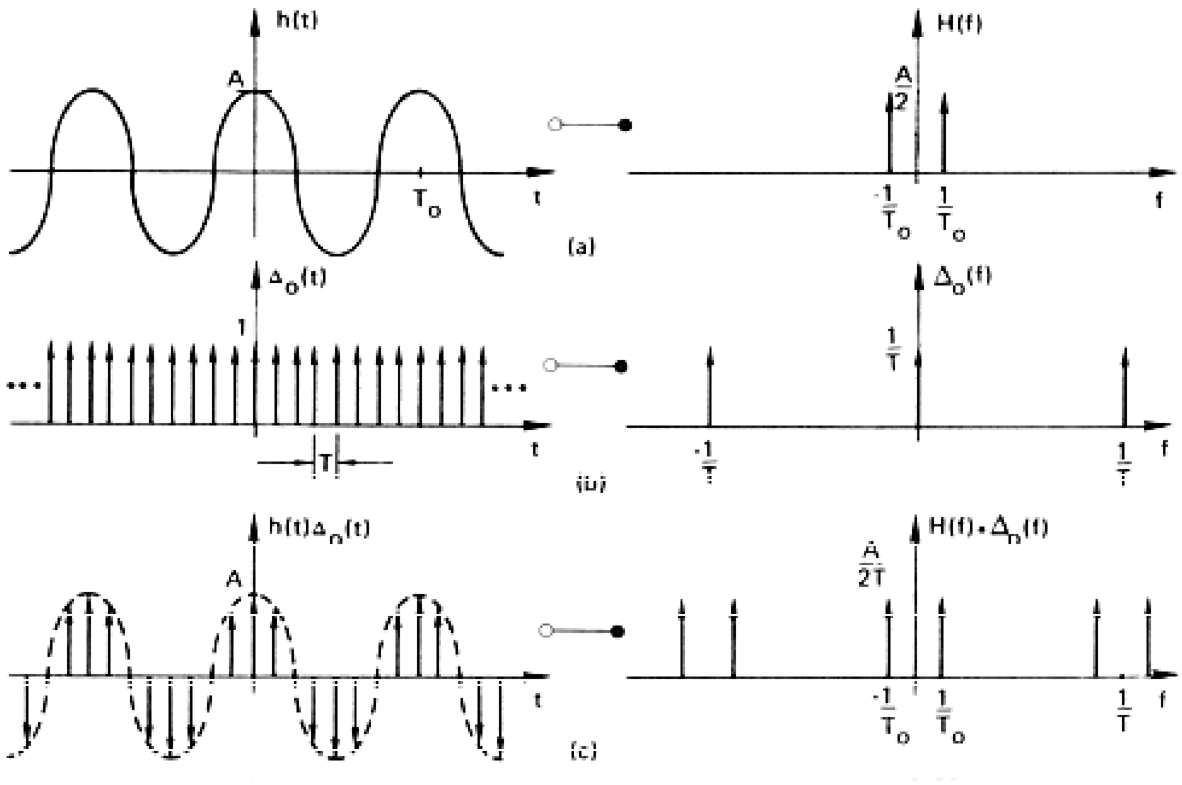


Frequency Transformation of an Audio Signal

	s(t)	continuous original signal
step 1		sampling at rate $f_s = \frac{1}{T}$
	s[t]	discrete original signal
step 2		temporal restriction to a window w(t)
	s[t]	discrete original signal containing N sampling values [0, *NT*]
step 3		N-point DFT
	S(f)	continuous Fourier transform
step 4		sampling at rate *N* per *T*
	S[f]	discrete Fourier transform

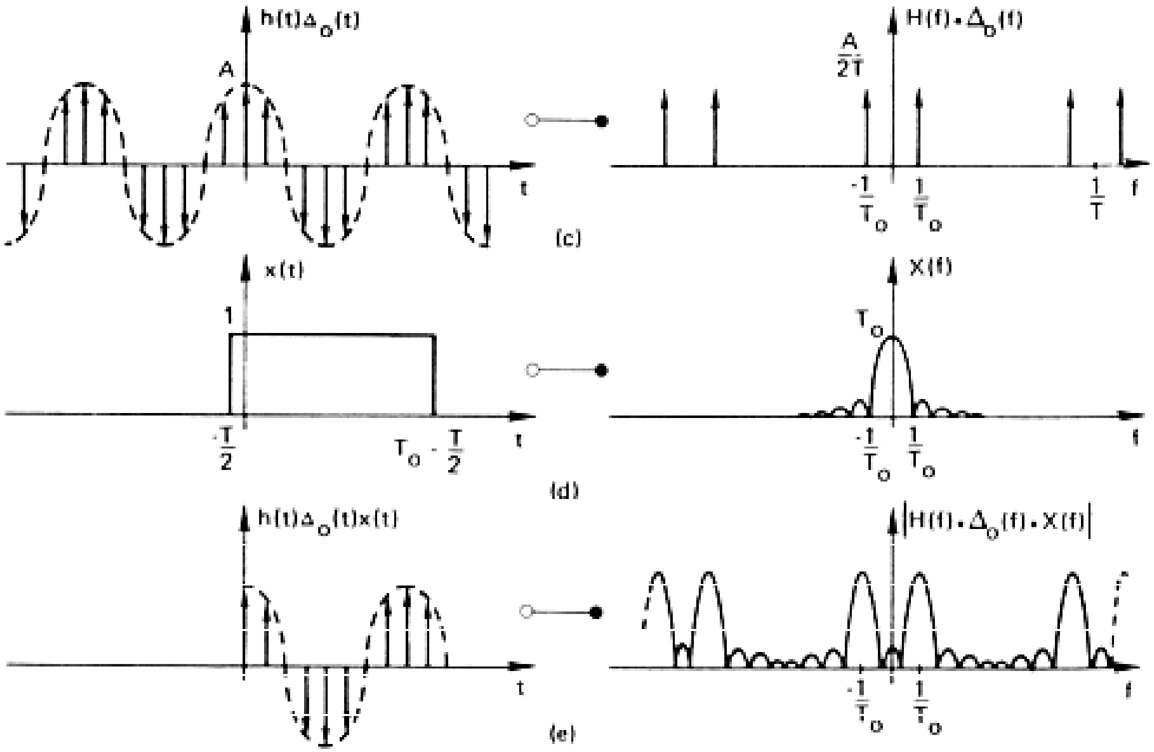
Steps 3 and 4 can be sped up considerably by means of the Fast Fourier transform (FFT). The complexity of FFT is O(n log n) compared to $O(n^2)$.

Step 1: Sampling in the Time Domain



Time domain **Frequency domain**

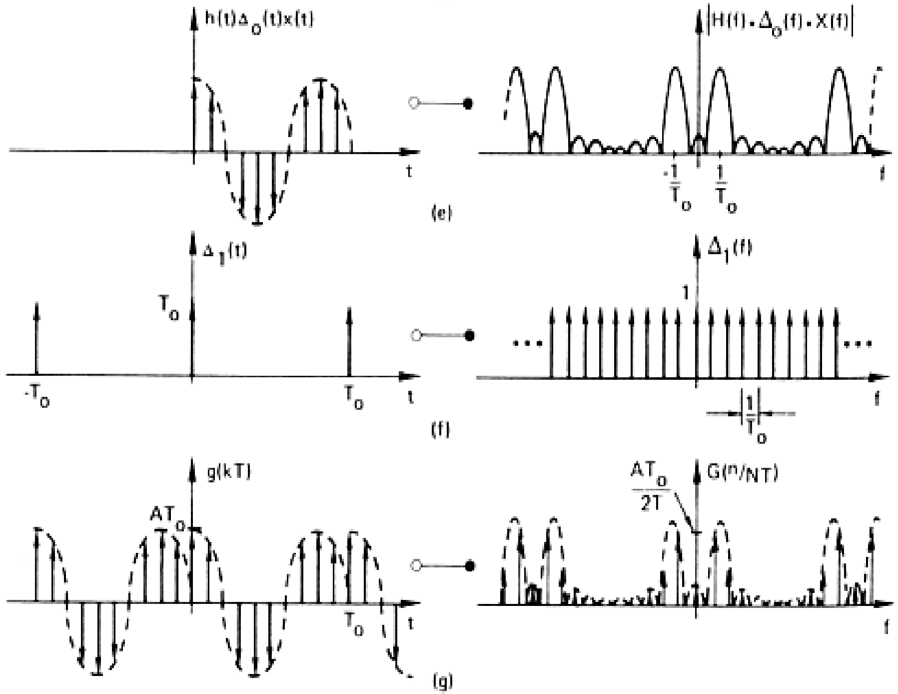
Step 2: Time Restriction to [0, *NT*]



Time domain **Frequency domain**

Step 3: Sampling in the Frequency Domain

Goal: Discretization of the data also in the frequency domain (for representation in the computer)



Time domain **Frequency domain**

Reference:

E.Oran Brigham: Fast Fourier Transform and Its Applications, Prentice Hall, 1997

Signal Analysis with the DFT

Assumption

A natural audio signal of sampling length *M* is given, e.g., *M* = 5 min of music.

Goal

Extraction of features, e.g., musical tones (pitch, loudness, onset, etc.)

Method

Definition of a window of size *N* which is moved over the audio signal. It represents a window of analysis. The DFT is computed on this window. Only with a **windowed** DFT, we can analyze the behavior of the signal over time.

Example: We can assume that musical tones are stationary for at least 10 ms. We thus choose *N* = 10 ms. When moving the window, we allow redundancy in order to also analyze the transitions between tones. Here, we chose an overlap of 2 ms. This results in

$$\frac{5x60x100}{8} = \frac{30.000}{8} = 3.750$$

frames.

Signal Analysis – Properties (1)

It is now possible to compute semantic features for the sample frames.

1. Energy

$$E_s(m) = \sum_{n=m-N+1}^{m} s^2(n)$$

m = ending time of the frame

E_s is a measure for the **acoustic energy** of the signal in the frame. It corresponds to the square of the area under the curve in the time domain.

The energy might as well be computed for the frequency-transformed signal. It then denotes a measure for its **spectral energy spread.** Computing the energy in the frequency space makes sense if one is interested in knowing frequency ranges in which the energy occurs.

Signal Analysis – Properties (2)

2. Zero-crossings

$$sign(s(n)) = \begin{cases} 1: & s(n) \geq 0 \\ -1: & s(n) \prec 0 \end{cases}$$

$$Z_s(m) = \frac{1}{N} \sum_{n=m-N+1}^{m} \frac{|sign(s(n)) - sign(s(n+1))|}{2}$$

- Counts the number of zero-crossings (i.e., sign changes) of the signal.
- High frequencies lead to a high Z_s, while low frequencies lead to a low Z_s
- This is closely related to the basic frequencies.

Many other parameters are also used in audio signal analysis.