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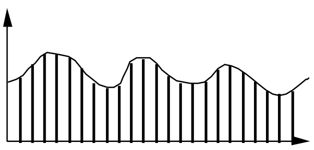
Chapter 5 - Audio

Audio Recording

- Usually speech or music, possibly arbitrary sound
- Digitization: Pulse Code Modulation - PCM
  - sampling, i.e. amplitude measurements, quantization at fixed time intervals



- Sampling theorem (Nyquist-Shannon theorem)
  - fundamental result in information theory
  - sampling of energy levels has to occur with at least double the frequency of the highest frequency occurring in the signal
  - phone: 3000 Hz, AM-radio: 4000 Hz, FM-radio: 8000 Hz
    Hifi: 22000 Hz (~ maximum frequency recognized by the human ear)
- Example Audio-CD:
  - 44100 sampling points per second and per stereo channel:
    176,4 KB per second, approx. 10 MB per minute, 635 MB per hour
- Encoding/Compression
  - Waveform Encoding (based on PCM)
  - Parameter Encoding

Waveform Encoding

- Logarithmic PCM
  - quantization intervals are not constant, but smaller for lower amplitude values
    - noise reduction in softer passages
    - e.g., μ-LAW (phone in North America and Japan), A-LAW (phone in Europe, rest of the world and international phone lines)
  - fewer bits are sufficient to cover the same amplitude
    - μ-LAW with 8 bit roughly equivalent to linear quantization with 12 bit, μ -LAW with 12 bit roughly equivalent to linear with 16 bit
- Differential PCM (DPCM)
  - subsequent sample values are often correlated
  - for each value, first compute a predicted value, then store the difference between the predicted and the actual value
  - p(xi) = a1 xi–1 + a2 xi–2 + ...
  - or simply: p(xi) = xi–1

Waveform Encoding (2)

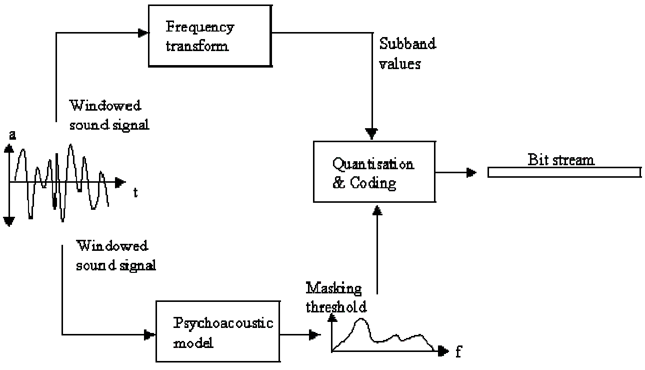
- Differential PCM (cont.)
  - with 256 quantization levels, differences between subsequent sample values of more than 32 levels rarely occur → 6 bits are sufficient

uncompressed	**112**	**114**	**117**	**115**	**111**	**109**
differences		**+2**	**+3**	**-2**	**-4**	**-2**

- Delta Modulation (DM)
  - number of quantization levels required for recording differences becomes smaller for shorter sampling intervals
  - → reduce sampling intervals until 1 bit is sufficient for recording the difference
    - for low bitrates (32 kbits/s, phone line quality) improvements over other approaches
- Adaptive DPCM (ADPCM) provides further improvement
  - prediction based on multiple preceding sample values
  - adaptive: resolution can vary
    - *high* for strong variations, *low* for weak variations

Waveform Encoding (3)

- MPEG-1 Audio
  - Layers I, II and III (MP3)
  - Bitrates [kbps]
    - 384 (1:4)
    - 256-192
    - 128-112 (CD quality)
  - psycho-acoustic model
  - redundancy of second channel
  - variable bitrate (Layer III only)



Parameter Encoding

- Parameter encoding (only for speech)
  - based on a model of the human vocal tract:
    - pitch of the voice based on oscillation frequency of vocal chords
    - position of tongue, lips, mouth, ...
    - described by parameters
  - determined using spectral analysis of short audio segments
    - window of a few miliseconds
  - e.g., using Linear Predictive Coding (LPC)
    - for the next window, only store difference to previous ("frequency x increased by y")
- both, wave form and parameter encoding realized by special hardware

Music: MIDI

- „Music Instrument Digital Interface“
  - used by music industry since 1983
  - defines an interface among (electronic) musical instruments (or computers)
- Representation based on instruments
  - type of instrument (e.g., grand piano), start/end of note, pitch, volume, etc.
  - 10 octaves, i.e., 128 notes
- requires ~200KB of MIDI-data for 10 minutes of music – a lot less than sampling
  - Keyboard → Computer: input,
    Computer → Synthesizer: output
  - Sequencer: device for storing/editing MIDI data – often a PC with sequencer software
- MIDI-Standard („General MIDI“):
  - 16 channels with one synthesizer instrument each
  - 128 instruments (e.g., 0 = "Acoustic Grand Piano")
  - 3-16 polyphonic notes per channel

Audio Media Object

- Raw data
  - sequence of energy levels (amplitude)
    or frequency components (Fourier analysis of a time window)
  - always in compressed format due to data volume
- Registration data:
  - resolution (bit depth):
    number of different energy levels
    often 256 (8 bits)
  - recording frequency (sampling rate)
  - number of channels (1 for mono, 2 for stereo, ...)
- Description data:
  - for speech: transcription as text
  - for music: transcription into musical notation or MIDI
  - structural information: pauses/silence

Audio Media Object (2)

- Operations:
  - Input - from a file in a specific format, or from a device (in real time!)
  - Output - to a file or a device (real time!)
  - Modification
    - cut/edit, similar to a recording studio;
      audio position based on time, sequence number of sampled value
    - adjust volume (difficult: non-linear)
  - Analysis, aggregation
    - statistics for sample value distributions
    - finding pauses/silence (based on amplitude threshold)
    - speech recognition
  - Comparison (search)
    - pattern matching
    - equality is too restrictive, similarity measures?!
    - based on description data (e.g., text)
- Subtypes
  - spoken language (most important), music, nature sounds, machine sounds (vehicle motor), ...

Indexing and Retrieval of Audio

- easiest method: using title, file name ...
  - most popular
  - but names may be incomplete, subjective – hard to find
  - does not support searching for audio that "sounds like" another audio
- Content-based audio retrieval
  - query by example, query by humming
  - comparison of (sub-)sequence of sample values
    - not promising, does not account for differences in sample rate, resolution
  - extraction and comparison of features
    - average amplitude
    - frequency distribution

General Approach to Audio-Retrieval

- Classification
  - most common types: speech, music, sound, ...
  - classes usually are of different importance for the application
    - class information itself may be useful for application
- Specialized treatment of each class
  - each class requires different processing and indexing techniques
    - speech is the most important class, and a number of successful speech recognition techniques/systems exist today
  - e.g., speech: speech recognition and indexing of resulting text
- Queries
  - need to be classified, processed, indexed
- Retrieval
  - based on similarity of query features and stored audio document features
  - search space can be reduced to a specific class

Audio Properties and Features

- Basis for classification and retrieval
- Two forms of representation
  - **time-domain** (amplitude over time)
  - **frequency-domain** (signal strength over frequency)

  with different properties/features
- Additional features
  - subjective, e.g. timbre

Features in the Time Domain

- Amplitude
  - represents pressure level compared to normal pressure of a medium
  - silence = amplitude is zero
- Average energy
  - charcterizes the loudness of the audio signal

$$E = \left(\sum_{n=0}^{N-1} x(n)^2\right) / N$$

with E as average energy, N as total number of sample values, x(n) as n-th sample

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Features in the Time Domain (2)

- Zero-crossing rate
  - number of signal sign changes, in some sense the average dominating frequency of the signal

$$ZC = \left(\sum_{n=1}^{N} |\operatorname{sgn} x(n) - \operatorname{sgn} x(n-1)|\right) / 2N$$

  - with sgn x(n) as the sign of x(n); 1 if x(n) positive, –1 otherwise
- Silence ratio
  - percentage of sample values that are part of a period (!) of silence
  - two threshold values:
    - amplitude threshold, below which a signal is regarded as silent
    - number of subsequent silent sample values defining a period (or interval) of silence

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Features in the Frequency Domain

- Every stable signal (i.e., periodic, without frequency changes) is the sum of sinoidal signals of different frequencies
- Fourier-transformation of the signal
  - decomposes signal into frequency components with factors (coefficients)
  - presentation: factors over frequencies (energy per frequency in decibel dB)
  - also called **spectrum** of the signal
- Bandwidth
  - interval of the frequencies appearing in the signal
    - difference of highest and lowest frequency in the spectrum
    - only frequencies with energy > 3dB are considered
  - larger for music than for speech

Features in the Frequency Domain (2)

- Energy distribution
  - is directly apparent from the spectrum
  - frequencies with high energy level: useful for classification
    - e.g., music has more frequencies with high energy levels than speech
  - computed based on **bands of frequencies** with high or low levels of energy
    - energy per band: sum of energies of all frequencies in the band
    - e.g., frequencies above/below 7kHz to classify speech (which rarely shows frequencies above the threshold)
  - centroid (or median) frequency: the mean of the spectrum
    - lower for speech than for music
    - also called *brightness*

Features in the Frequency Domain (3)

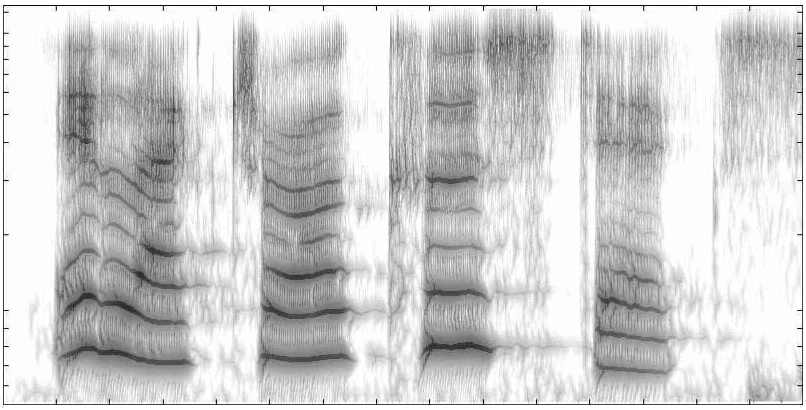
- Harmony
  - spectral compnents are often multiples of the lowest and loudest frequency ("fundamental frequency")
  - music is usually more harmonic than sounds
  - determining whether a recording is harmonic:
    is the dominant component a multiple of the fundamental frequency?
  - example: flute plays note G4;
    - peaks are at frequencies 400 Hz, 800 Hz, 1200 Hz, 1600 Hz
    - f, 2f, 3f, 4f etc. are harmonics of the note
- Pitch
  - defined for periodic (stable) signals (instruments, voice, ...)
  - not defined for percussion, noise, etc.
  - is subjective to human perception
    - usually close to, but not necessarily identical to the fundamental frequency)

Spectrogram

- Simple presentation has its limitations
  - time domain doesn't show frequency-related information
  - frequency domain doesn't show when the frequencies occur
- Combined presentation
  - raster image, matrix
  - x-axis: time
  - y-axis: frequency
  - blackness/intensity or color of pixel: energy of frequency at the specific time
- Analysis
  - regularity of occurrence of frequencies
  - music is more regular than speech

Spectrogram (2)

- Example
  - femal speaker, (english "Electroacoustics"), signal duration 1,5 s
  - source: http://www.mmk.ei.tum.de/~rue/mum/eurospeech99/demo/



Classification

- Based on features
  - here only for music and speech
  - could be further differentiated:
    - types of music, male or femal speech
- Speech
  - bandwidth relatively small, 100 – 7000 Hz
  - Centroid is lower than for music
  - frequent pauses (between words, sentences) – high silence ratio
  - characteristic structure: sequence of syllables, consisting of short periods of friction (consonants) followed by longer periods of vowels – frictions show high zero crossing rate
- Music
  - high bandwidth, 16 – 20.000 Hz
  - centroid is higher
  - low silence ratio
    - except: solo instrument, a-capella singing
  - zero crossing rate does not show string variations
  - regular beat

Classification System

- Step by step, feature by feature
  - e.g., first consider centroid – if high: music
  - then silence ratio – if low: music
  - then zero-crossing variability
    - low: solo music
    - otherwise: speec
- Order is important
  - algorithmic complexity, effectiveness of classification
- A single feature is already useful:
  - only zero crossing rate: up to 90% correctness
  - only silence ration: up to 80% correctness
- Feature vector
  - combines values of a set of features
  - training: compute average vector (reference vector) of each class
  - for new audio, compute feature vector and determine distance to all reference vectors (using, e.g., euclidean distance)

Speech Recognition

- Performed after classification
- Techniques
  - Time Warping (speed of speech)
  - Hidden Markov Models
  - neural networks
- Performance

domain	*type*	*vocabulary*	*error rate in %*
digits	read	10	< 0,3
flight reservation system	spontaneous	2500	2
Wall Street Journal	read	64000	7
radio news	read / spontaneous	64000	30
phone call	spontaneous	10000	50

Music Indexing

- Structured music (MIDI)
  - no extraction of features required
  - exact match is a valid search option
  - but maybe the instrument is different for some tracks
  - similarity is hard to define
  - one option: only consider change of pitch
    - Up, Down, Repeat – U, D, R
    - Parsons, D., *The Directory of Tunes and Musical Themes*, Spencer Brown, 1975
    - Melodyhound: http://name-this-tune.com/ (Uni Karlsruhe)

    retrieval reduced to character string comparison
- Recorded music (sample-based)
  - Query: singing, humming
  - Set of features to consider
    - e.g., volume, pitch, brightness, bandwidth, harmony
    - vector and distance computation
  - Pitch
    - extract/guess for each note ("pitch tracking")
    - representation as a series of pitches, or pitch changes (see above)
    - similarity: sequences may differ in k pitches

Summary

- Audio Digitization
  - PCM
- Audio Encoding
  - Waveform Encoding
  - Parameter Encoding
- Music: MIDI
- Audio Media Object
- Indexing and Retrieval of Audio
  - general approach: classification-based
  - features in the time domain: energy, zero-crossing rate, silence ratio
  - features in the frequency domain: bandwidth, energy distribution, brightness, fundamental frequency, harmony, pitch
  - spectrogram combines time and frequency domains
- Classification
- Music Indexing: still not mature enough