Dialogue Systems

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spring 2023

Sound Digitization

- Objectives - transformation of continuous acoustic signal into a series of a discrete values, which can be processed by computer.
  - These values can be processed by computer.
- Sound digitization steps:
  1. sampling – scanning the current value of the signal characteristics at given frequency (sampling frequency).
  2. quantization – transformation of values from sampling (real numbers) values into the computer real/integral number representation.
  3. waveform coding– the way of storing information about the input signal.

Sampling

- Scanning the present signal value – scanning is repeating at predefined rate (sampling rate).
- Sampling rate – should be double of the maximum frequency present in the signal to be able to reconstruct the original signal without loosing included information (Shannon sampling theorem)
- acquired values must be quantized and stored in suitable way.
- Commonly used sampling rates:
    - 8 kHz – phone quality
    - 16 kHz
    - 22050 Hz – FM quality
    - 44100 Hz – CD quality
    - 48 kHz – DVD quality

Quantization

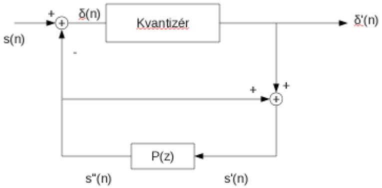
- Method how to transform continuous values into the discrete ones.
- Principle:
  - When the present sample cross the n-th multiple of quantization step the value n is send to the output.
  - Quantization step:
    - the average difference of input to change output by one.
    - quantization step = input values range/number of output values
  - quantization error – rounding error caused by value of quantization step, proportional to the quantization step.
- Commonly used integer quantizations:
  - Sound processing: 8 bits ($2^8 = 256$ levels), 16 bits ($2^{16} = 65536$ levels), 24 bits ($2^{24} = 16777216$ levels)
  - image processing adds 32 bits ($2^{32} = 4294967296$ levels)
- Besides integer quantizations there are floating point 32 bit and 64 bit quantizations.

Wave Coding Methods

- Direct storing the values from previous quantization – PCM coding (Pulse-Code Modulation).
  - sound signal changes are relatively slow – only small differences of neighbouring samples.
  - Big redundancy of data.
  - Big amplitude dispersion problem (big quantization step may cause big quantization error, small quantization step may cause sample overflow in case of amplitude increase).
- Differential PCM – stores neighbouring sample differences
- Adaptive PCM — PCM with variable quantization step value – quantization step adapts to the signal amplitude.

Differential Pulse Code Modulation

- Based on following assumptions:
  - Difference of neighbouring samples is much less than the value of a sample.
  - The following sample can be relatively precisely estimated as a previous samples linear combination.
- Differential PCM coding block schema



  - $s''(n)$ – sample assumption
  - $s'(n)$ – reconstructed signal, gained from the following sum quantized signal $\delta'(n)$ a $s''(n)$
  - $\delta(n) = s(n) - s''(n)$

Adaptive Pulse Code Modulation

- Big signal amplitude changes may cause:
  - Inaccurate week signal capture – amplitude is too small, comparable to quantization step (quantization step is too big).
  - Strong signal distortion – overflow of a range of values used to signal coding (quantization step is too small).
- Solution: adapting quantization step to a signal amplitude.