Speech Signal Processing – introduction

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DCGM FIT BUT Brno

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Agenda

- Who will be teaching
- Program of the course
- Rating
- Literature and Web

- Disciplines connected to the speech processing
- Information content of speech.
- Aplication areas + demonstrations.

Who will be teaching

- lectures: Valentina Hubeika.
- labs: Valentina Hubeika.
- maybe with help of: Stefan Kombrink.
- (some other guys when not present in Brno.) `http://www.fit.vutbr.cz/speech`

Program the course

- **Lectures** – 12 lectures, 2 hours each
- **Computer labs** – Matlab under Linux
- **Numerical exercise** – one at the end of the semester, examples to LPC, DTW, HMM, some more examples during lectures.
- **Projects** – no projects in this course

Lectures

P1 Organization of the course, applications, disciplines.

P2 Digital speech signal processing: recording of a speech signal - sampling, quantization. Speech spectrum estimation - Fourier transform - continuous time; what will be happening when the signal sampled. Discrete Fourier transform. Random signals, power spectral density. Modification of speech - filtering. Filter frequency characteristics.

P3 Speech preprocessing: mean value, preemphasis, frames, basic parameters, spectrogram.
Speech production: vocal apparatus and its model - vocal cords a vocal tract vs. excitation and modification (filter). Basic characteristics in time and spectral domain: influence of the excitation and modification. Formants. Advantages of the short-term and long-term spectrogram. How to separate excitation and modification: cepstrum + MFCC.

- P4 Linear-predictive model: estimating characteristics of the vocal tract, not the excitation ⇒ application in coding and recognition. Prediction of the sample from the previous samples - linear prediction (LP). LP error. Estimation of the LP error. Prediction of the articulation apparatus configuration using LP analysis. Spectrum by means of linear prediction. Parameters derived from LP - LAR and LSF, which are used in encoders. LPC-cepstrum.
- P5 Estimation of fundamental frequency (pitch). Terminology. Differences in pitch between males, females and children. Utilization in speech processing systems. Methods based on autocorrelation function. Normalized cross-correlation function (NCCF). Long-term predictor and cepstral analysis in pitch estimation. Accuracy and drawbacks of the pitch detectors.
- P6 Speech encoding I: Objectives. Bit rate, objective a subjective quality measure. Subdivision based on bit rate and quality. Encoding in time domain. Vocoders - LPC. Vector quantization in speech encoding.
- P7 Speech encoding II. - application of CELP, encoding in GSM: GSM, GSM-EFR, GSM-HR, Voice over IP.

P8 Introduction to recognition – objectives, classification: isolated/connected/LVCSR, speaker dep/indep. Basic functional blocks. Speech activity detection in isolated word recognition.

Recognition based on speech frame distance - alternative definition of distance. Linear correction, dynamic programming (Dynamic Time Warping DTW).

P9 Hidden Markov models (HMMs): objectives and relation to DTW, modeling, Gaussian distribution, state sequences. Utterance probability based on the state sequence, Baum Welch a Viterbi probability. Model training: Baum Welch, recognition - Viterbi. Token passing. Continuous words.

P10 HMM II. Continuous speech with a big vocabulary: recognition based on smaller tokens - phnemes... Phonetic structure of a language. Vowels and consonants, characteristics and classification of phonemes. International standards for phoneme denotation: IPA a SAMPA, TIMIT. Coarticulation.

Recognition: context-dep. tri-phones. Large vocabulary, language modeling, lattice re-scoring, forced alignment.

P11 Recognition features. Requirements - pitch discard, decorrelation. Spectral envelope. What we can do and how we can use it: LPCC, MFCC to decorrelate the features; PCA, LDA, HLDA, normalizations to reduce the influence of the channel. Other improvements - delta, delta-delta. “Hot-topics in paratemerization”: TRAPs and FeatureNet, neural nets.

Tools in speech processing.

P12 Speech synthesis: Structure of the synthesizer. Conversion of text into spoken form: text-to-speech. Text normalization. Prosody (melody, accent, timing, pausing,...) in speech synthesis. Units used in synthesis - manual and automatic selection (corpus-based). Generation of the signal in time and frequency domain. Methods of PSOLA a HNM. Applications. SW used in synthesis: EPOS, MBROLA, Festival.

N1 Numerical exercise: digital filter, LPC - calculation of the filter parameters, DTW, HMM, spectrogram interpretation.

Labs

- Speech processing in Matlab.
- LPC and vector quantization, LPT error.
- Pitch detection. Simple decoder.
- Gaussian Mixture Models. Simple classificator.
- DTW and HMM in Matlab.
- HMM using HTK.

Evaluation of the Course

category	points
6 labs, each 5 points	30
midterm - theoretical questions only	20
final exam - theory and numerical tasks	50
altogether make	100

– Materials are allowed during both exams. . .

LITERATURE

- Gold B., Morgan. N.: Speech and audio signal processing, John Wiley & Sons, 2000 [library]
- S. Young, J. Jansen, J. Odell, D. Ollason, P. Woodland: The HTK book, Entropics Cambridge Research Lab., 1996, Cambridge, UK. Good introduction to HMM, available for download at `http://htk.eng.cam.ac.uk/`

ZRE is for you!

- Something you want to change about the lectures or/and labs?
- Something is outdated or missing?
- Suggestions about the evaluation?
- All comments are welcomed during lectures or by mail.

INTRODUCTION INTO AUTOMATIC SPEECH PROCESSING

Definition

“Automatic speech processing allows voice communication between people (encoding) or between a human being and a machine.”

Disciplines in Speech Processing

Speech processing is a pluri-disciplinary field, utilizing knowledge of natural sciences, technical sciences and social sciences.

- **Physiology:** study of articulatory and auditory apparatus,
  Knowledge used to facilitate design of the model.
- **Acoustics:** studies physical mechanisms of production and perception of speech.
- **Signal processing:** multiple areas: modeling, parameterization, identification, spectral analysis, encoding, informational theory, pattern recognition, etc.
- **Social sciences**
  - *phonetics* – a subfield of linguistics that comprises the study of the sounds of human speech.
  - *phonology* – s subfield of linguistics that deals with the sound systems of languages. Subdivides speech into basic units, phonemes.
    Phoneme is the smallest posited structural unit that distinguishes meaning, though they carry no semantic content themselves.
  - *prosody* – a study of the sound of the language (melody, duration of phonemes, accent in words and sentences, ...).

- *lexicology* – is that part of linguistics which studies words, their nature and meaning, words' elements, relations between words (semantical relations), word groups and the whole lexicon.
- *grammar* – is the field of linguistics that covers the conventions governing the use of any given natural language. Grammar is the set of rules describing use of the language. Important for the synthesis.
- *syntaxis* – is the study of the principles and rules for constructing sentences in natural languages.
- *semantics* – is the study of meaning in communication. Basic item is usually a word.

INFORMATIONAL CONTENT OF SPEECH

the goal is to estimate *informational speed* (in bits per second, bit/s or bps), required to express speech in different formats. For comparison, we will express phonetic and acoustic form in digital representation.

Phonetic Form

number of phonemes in Czech is 36. To calculate *informational quantity* we will consider a source generating mutually independent elements x_i from the set $X = \{x_1, \ldots, x_S\}$, where S is a finite set of items. Each item has the probability $p(x_i)$ and the items constitute a complete system, thus:

$$\sum_{i=1}^{S} p(x_i) = 1. \tag{1}$$

Informational content of the i-th item is given by the number of bits we need to express the item:

$$I(x_i) = -\log_2 p(x_i) \quad \text{[bit]}. \tag{2}$$

Source entropy (average information value) is given by:

$$H(X) = -\sum_{i=1}^{S} p(x_i) \log_2 p(x_i). \tag{3}$$

For the Check phonemes, in case we assume the same probability of occurrence the entropy is $H(X)$=5.2 bits. Considering the true phonemes probability, the entropy becomes $H(X)$=4.6 bits. Considering the mutual *dependency* (conditional probability) of the phonemes (bi-grams: $p(x_j|x_i)$, trigrams: $p(x_k|x_i x_j)$, etc.), the value of the entropy becomes $H(X)$=3–3.5 bits.

In the average Czech conversation, a human being produces cca 80–130 words per minute, thus about 10 phones per second. Informational speech is then approximately C_{phn} = **30–40 bit/s**. Psychoacoustic tests show, that a human being is able to process incoming information with the speed of app. 50 bit/s.

Acoustic Form

In case speech is represented by a digital signal, the Nyquist–Shannon–Kotelnikov theorem must hold:

$$F_s > 2F_m, \tag{4}$$

where F_s is the sampling frequency and F_m is the highest frequency presented in the spectrum of the signal. Each sample is quantized by one of the m of allowed quantization levels, which can be expressed by $N = \log_2 m$ bits. Signal to noise ratio is proportional to $6N$ (in dB). The informational speed can be hence expressed as:

$$C_{ak} = \frac{I}{t} = \frac{\log_2 m}{T_s} = NF_s. \tag{5}$$

Examples:

1. We are given a signal in Hi-Fi quality, F_s=44.1 kHz, N=16 bit. Resulting informational speed is C_{ak} = **705 kbit/s**.
2. For a signal in telephone quality (with the band from 300 to 3400 Hz): F_s=8 kHz, N=8 bit. Resulting informational speed is C_{ak} = **64 kbit/s**.

Conclusion

We can see from the example that the acoustic form comparing to the phonetic form is greatly *redundant*. Along to the information contained in the phonetic form, the acoustic form carries information about the speaker, their mood, environment and so on. The listener then subconsciously separates different informations in their brain. Unfortunately, it is not know so far how exactly. Nevertheless, it is useful to utilize the basic knowledge about the speech production to *lower the informational speed*.

APPLICATION DOMAINS IN SPEECH PROCESSING

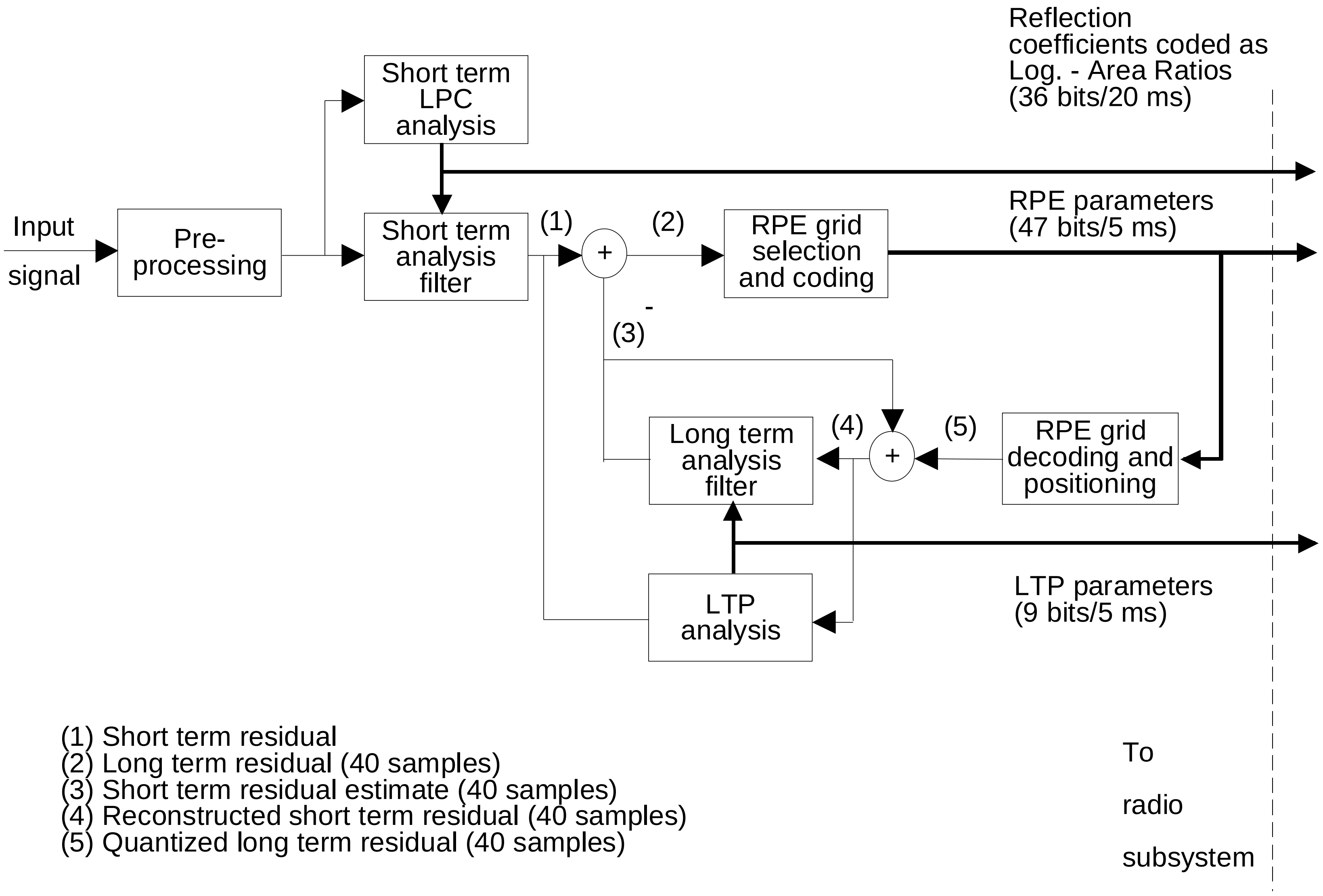
encoding: facilitates transfer and storage.

Goal: represent the speech on the smallest possible number of bits.

Requirements:

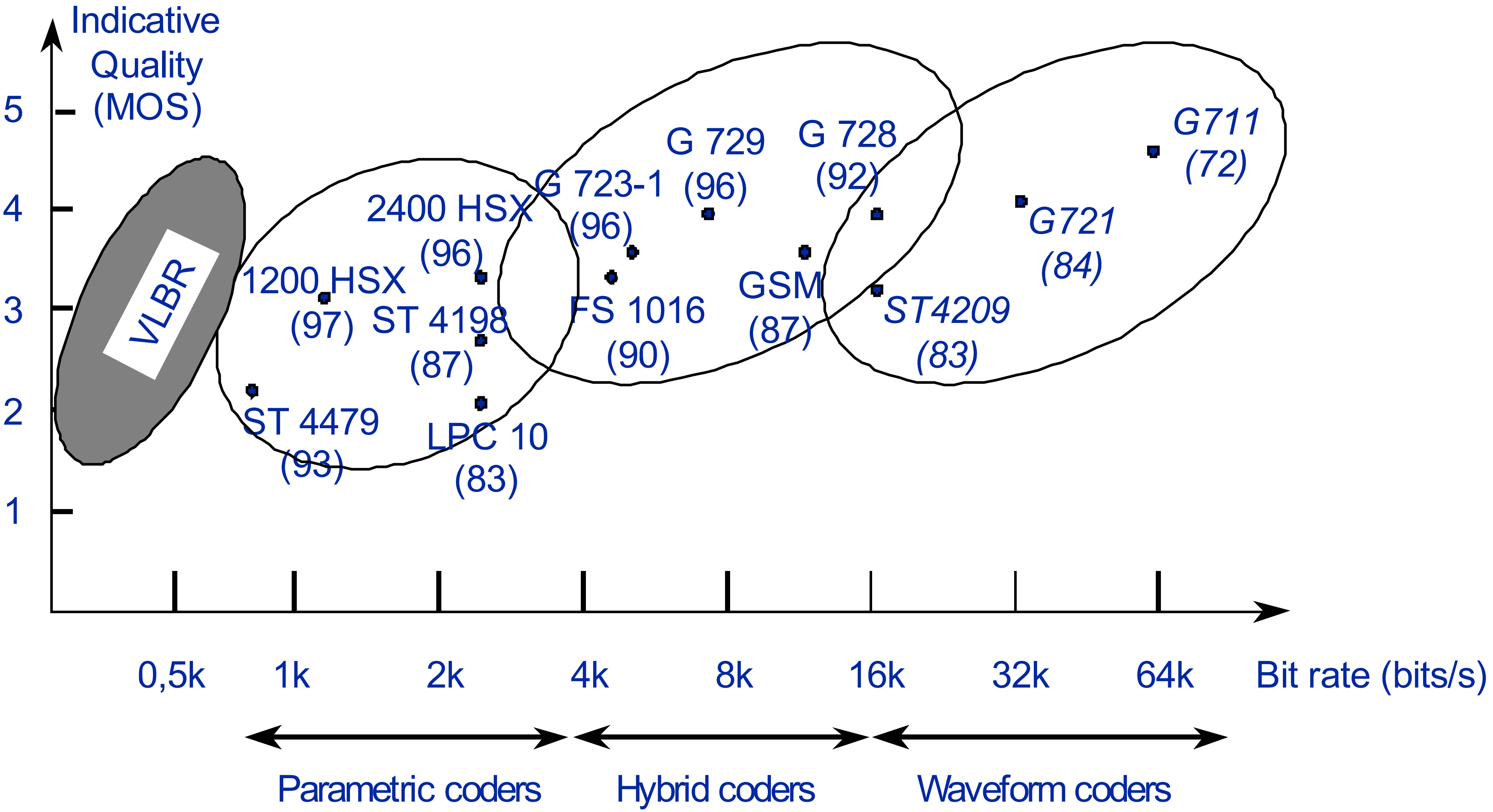
- complexity ↓,
- delay ↓,
- intelligibility ↑,
- natural sound ↑,
- robustness against errors in the channel ↑.

. . . cellphones



(1) Short term residual
(2) Long term residual (40 samples)
(3) Short term residual estimate (40 samples)
(4) Reconstructed short term residual (40 samples)
(5) Quantized long term residual (40 samples)

Characteristics over time ?



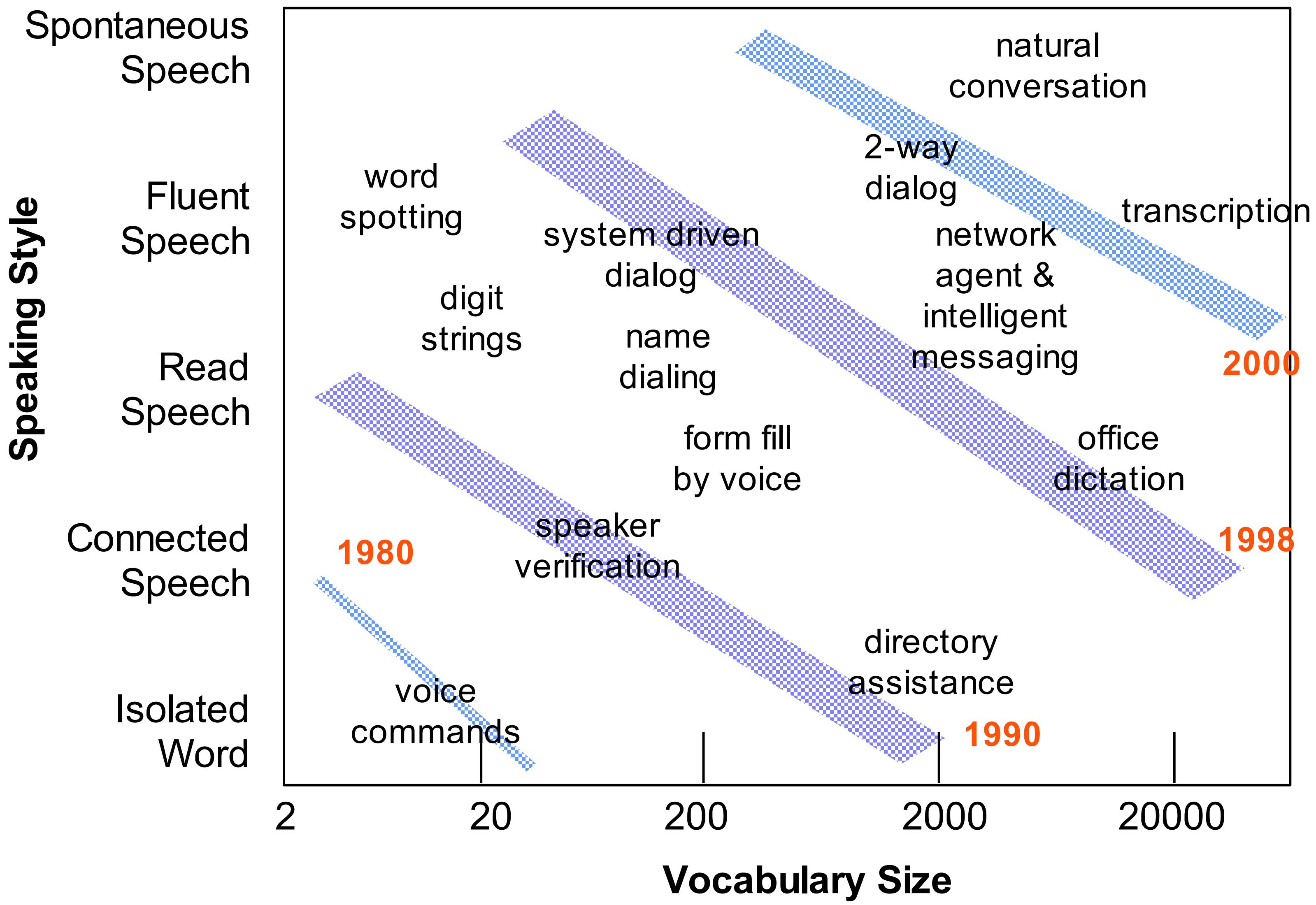
Demo — ©Andreas Spanias (Arizona University)

`http://www.eas.asu.edu/~speech/table.html`

Recognition:

- **of speech** (Speech Recognition)
  - isolated words
  - continuous words (for instance, figures in telephone number credit cards).
  - continuous speech – the hardest task, still not working in all cases, especially in cases with large vocabulary (LVCSR - Large Vocabulary Continuous Speech Recognition).
- **of the speaker** (Speaker Recognition)
  - identification – who is the speaker given a set of reference speakers ?
  - verification – are two speech segments coming from the same speaker ?

What we have?


Speaking Style
Spontaneous Speech
Fluent Speech
Read Speech
Connected Speech
Isolated Word
Vocabulary Size
2
20
200
2000
20000
natural conversation
2-way dialog
transcription
word spotting
system driven dialog
network agent & intelligent messaging
digit strings
name dialing
2000
form fill by voice
office dictation
speaker verification
1998
1980
directory assistance
1990
voice commands

synthesis:

computer has to generate speech it never “heard” before (e.g. the speech wasn’t recorded from a human speaker). The most difficult is synthesis from text. (TTS – text-to-speech).

demos: `http://www.cs.indiana.edu/rhythmsp/ASA/Contents.html`

Other applications. . .

- medicine: examination of the abnormality and illnesses of the vocal tract.
- psychology a criminalistics: lie detector, estimation of the level of alcohol in blood, stress detection. . . .
- aiding of handicapped (helping improving pronunciation to the deaf etc.)
- language identification
- keyword spotting