

LP Synth Filter: Linear Prediction Synthesis Filter



# Infocommunication Speech Processing

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# **SPEECH PROCESSING, SPEECH TECHNOLOGY**

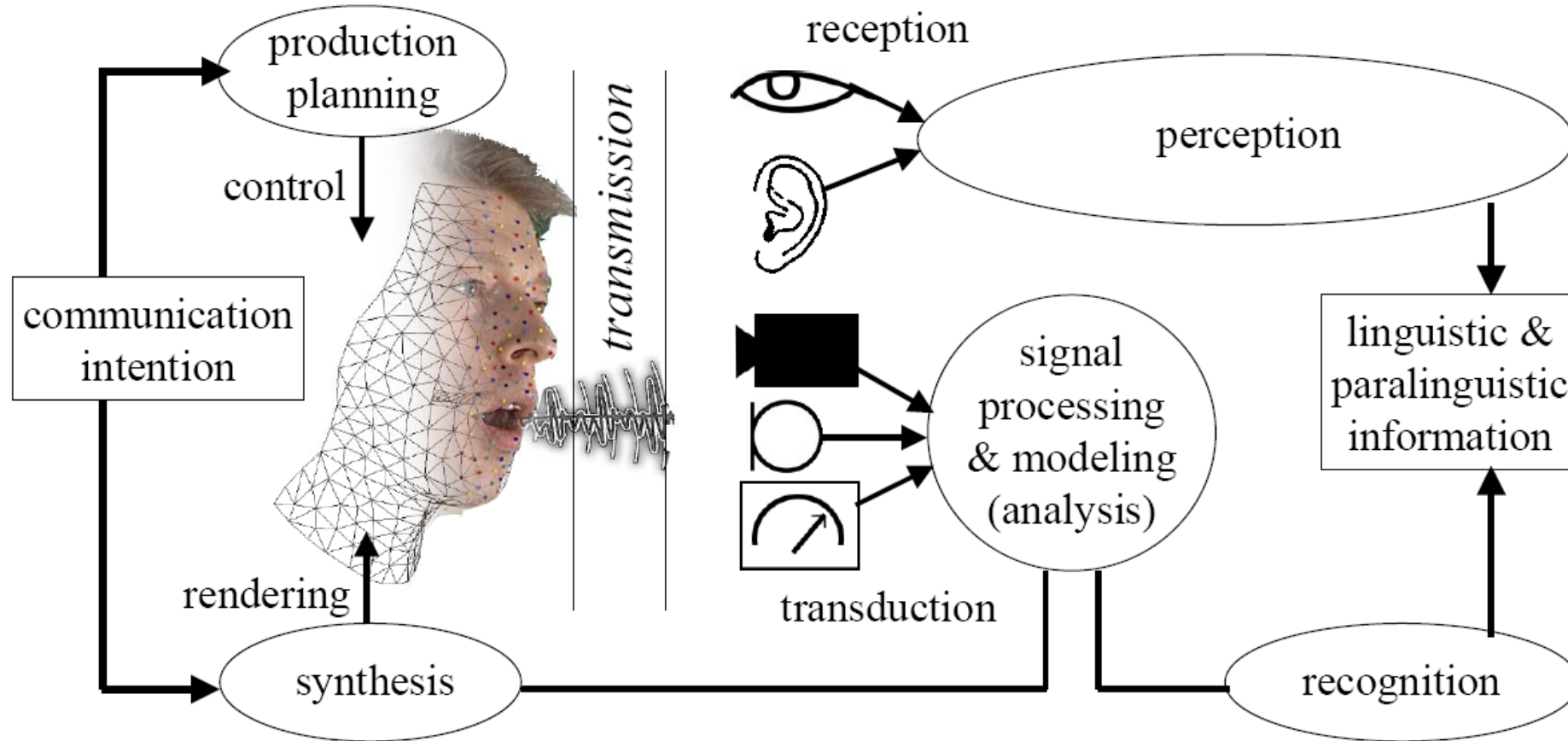
# Speech

- the most natural form of human-human communications
- related to language; linguistics is a branch of social science
- related to human physiological capability; physiology is a branch of medical science
- also related to sound and acoustics, a branch of physical science
- one of the most intriguing signals that humans work with every day

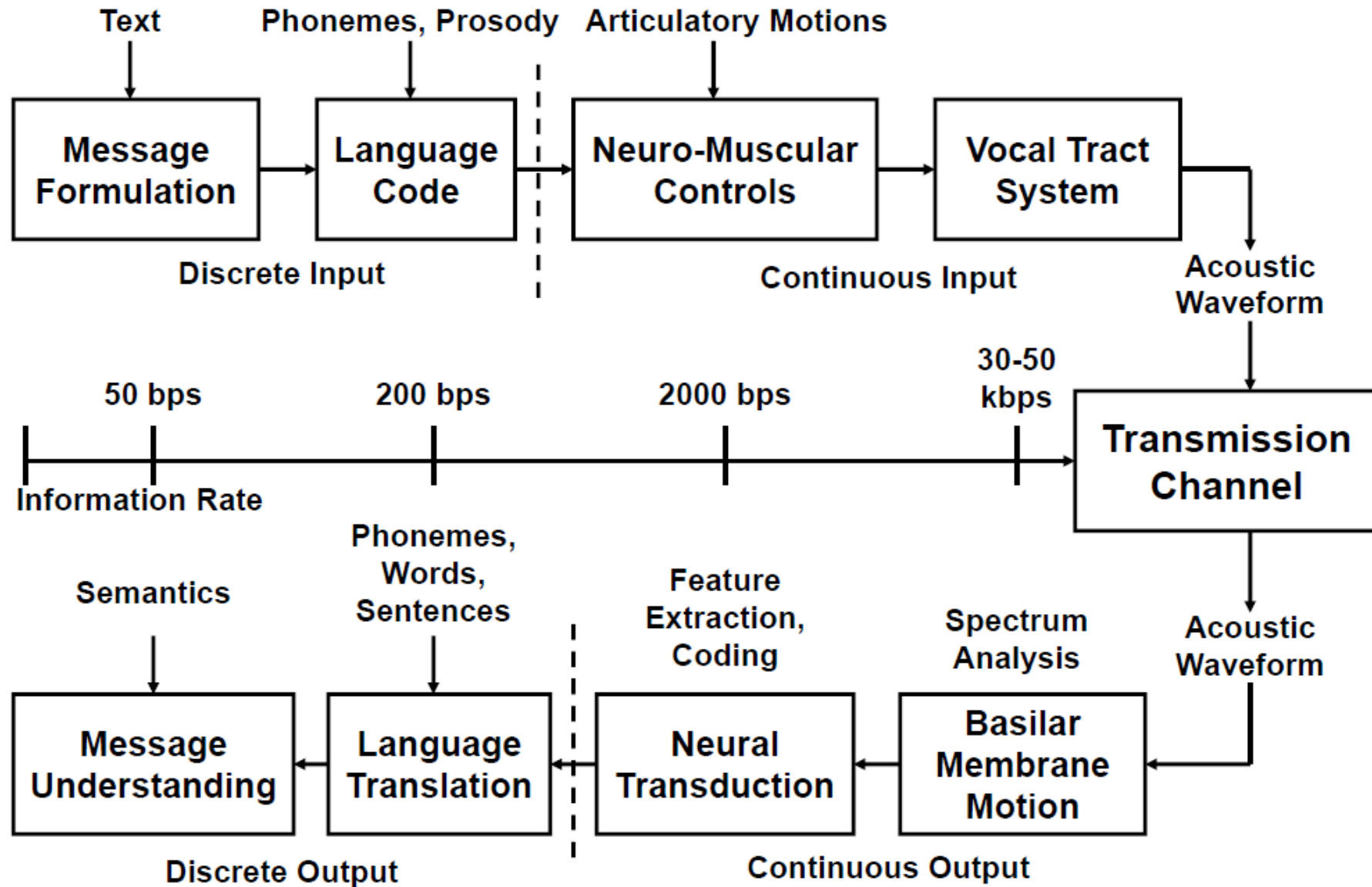
# Speech processing

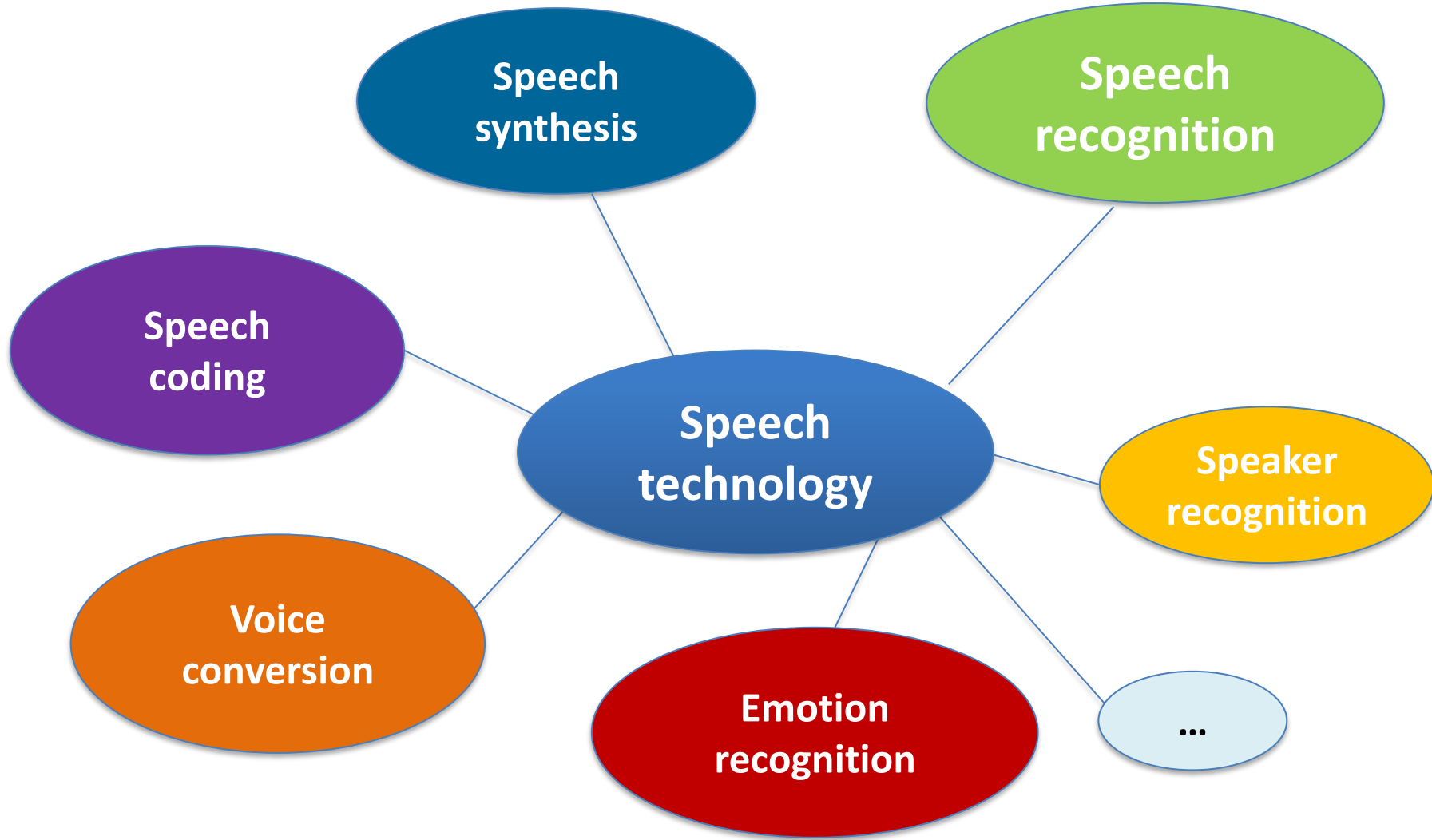
- Purposes:
  - to understand speech as a means of communication
  - to represent speech for transmission and reproduction
  - to analyze speech for automatic recognition and extraction of information
  - to discover some physiological characteristics of the talker

# Speech technology

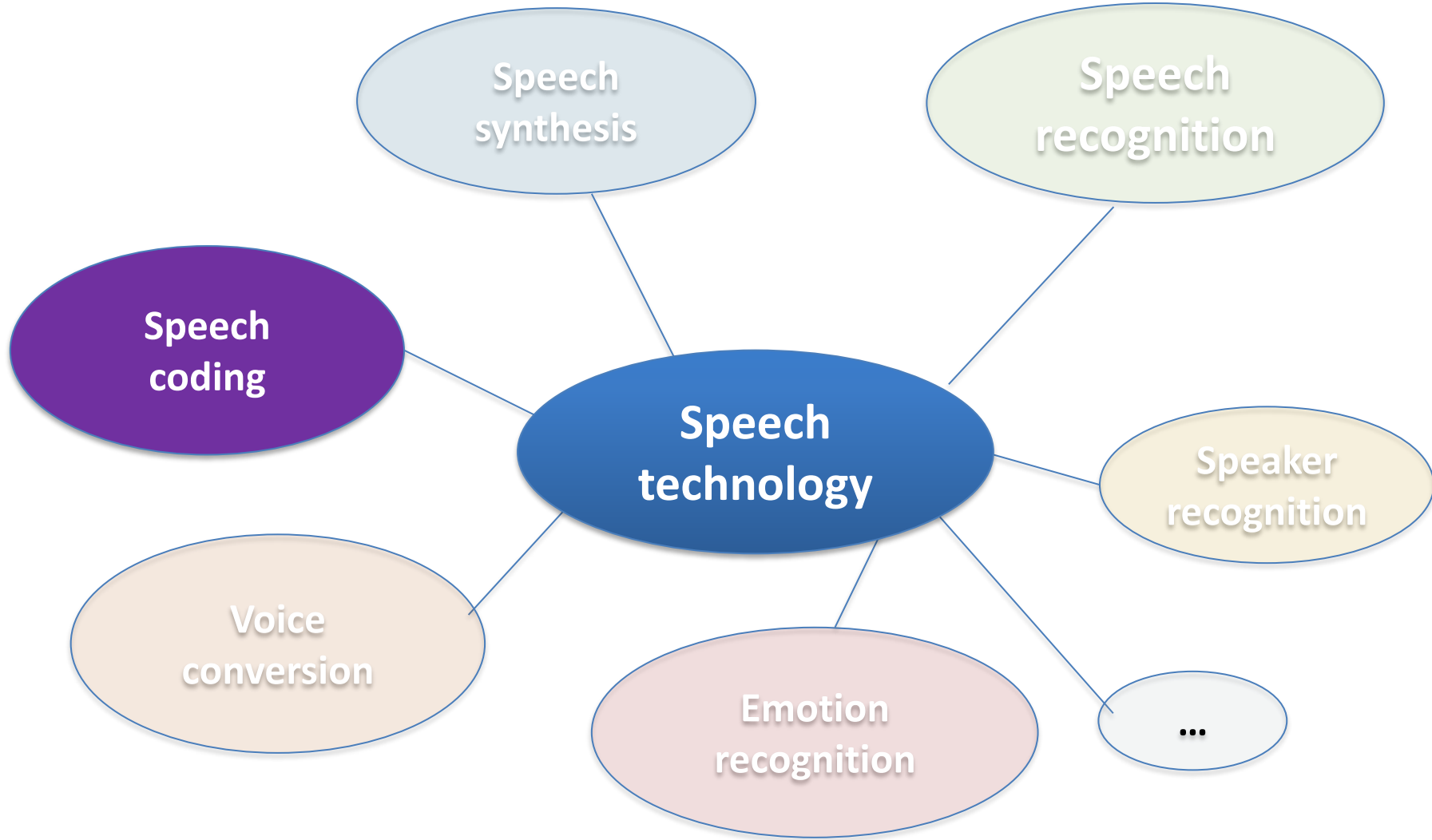


# The Speech Chain



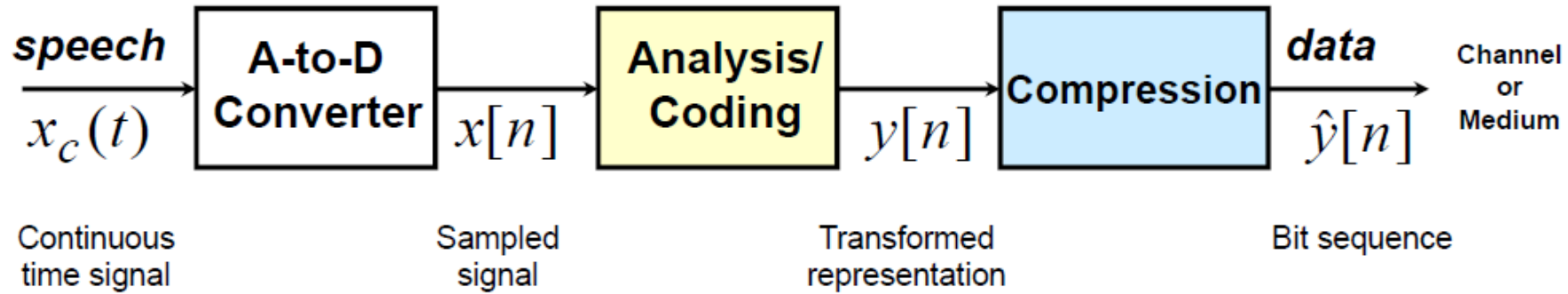




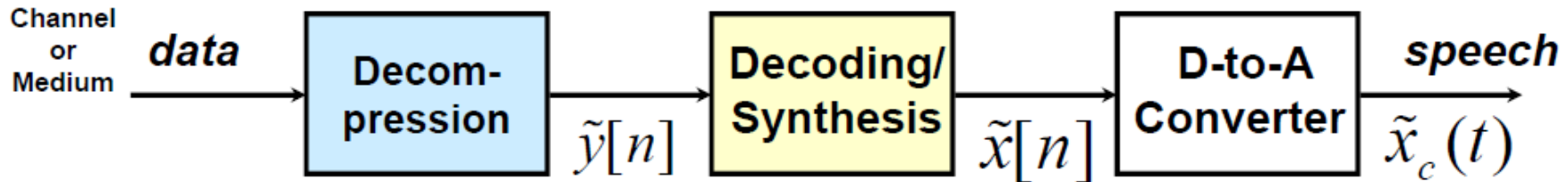


# Speech coding

## Encoding



## Decoding



# Speech coding

- ***Speech Coding*** is the process of transforming a speech signal into a representation for efficient transmission and storage of speech
  - narrowband and broadband wired telephony
  - cellular communications (e.g. GSM, UMTS)
  - Voice over IP (VoIP) to utilize the Internet as a real-time communications medium
  - extremely narrowband communications channels, e.g., battlefield applications using HF radio
  - storage of speech for telephone answering machines, IVR systems, prerecorded messages

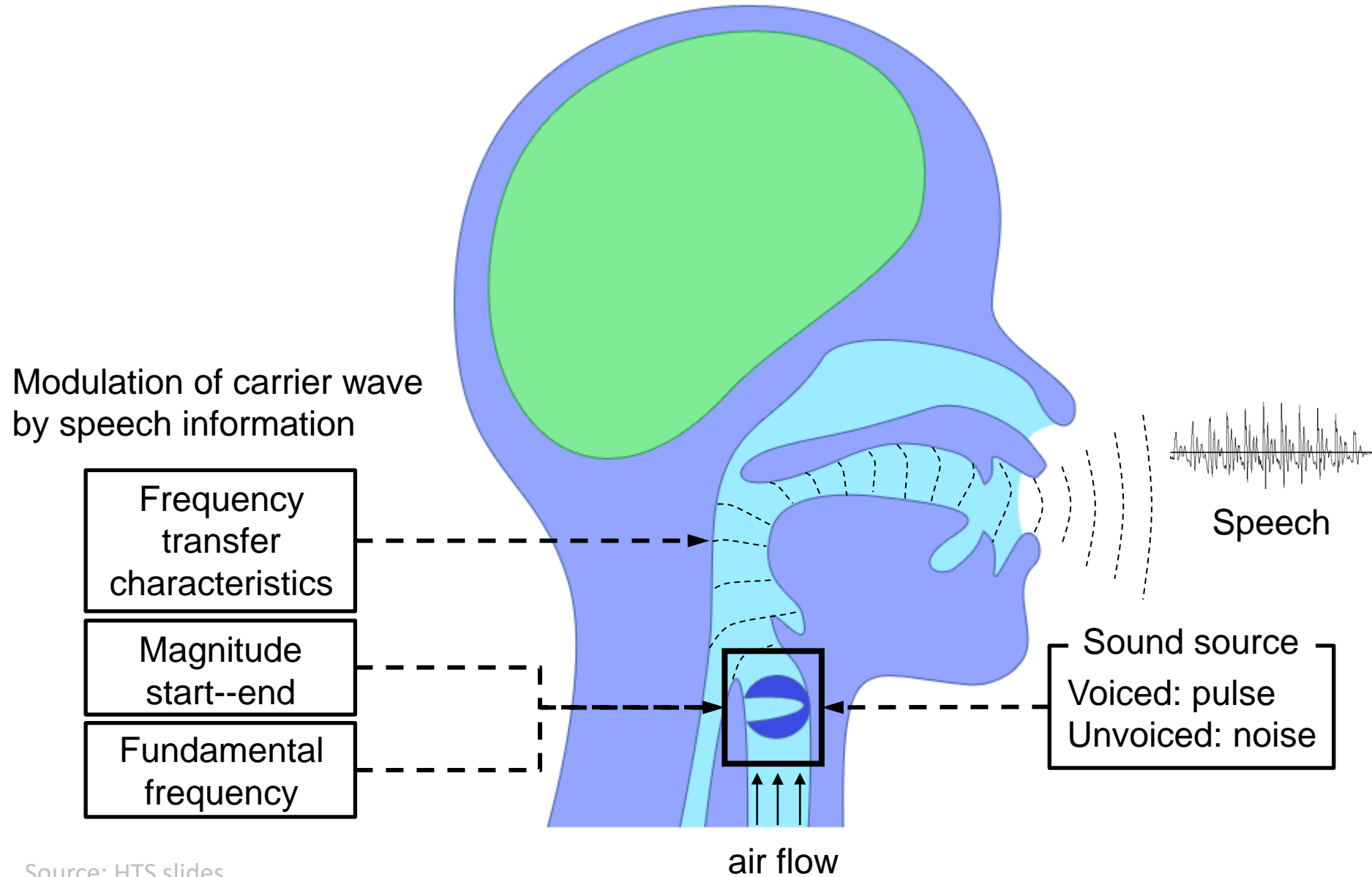
# Information Rate of Speech

- from a Shannon view of information
  - message content/information--2\*\*6 symbols (phonemes) in the language; 10 symbols/sec for normal speaking rate => **60 bps** is the equivalent information rate for speech
- from a communications point of view
  - speech bandwidth is between 4 (telephone quality) and 8 kHz (wideband hi-fi speech)—need to sample speech at between 8 and 16 kHz, and need about 8 (log encoded) bits per sample for high quality encoding => 8000x8=64000 bps (telephone) to 16000x8=128000 bps (wideband)

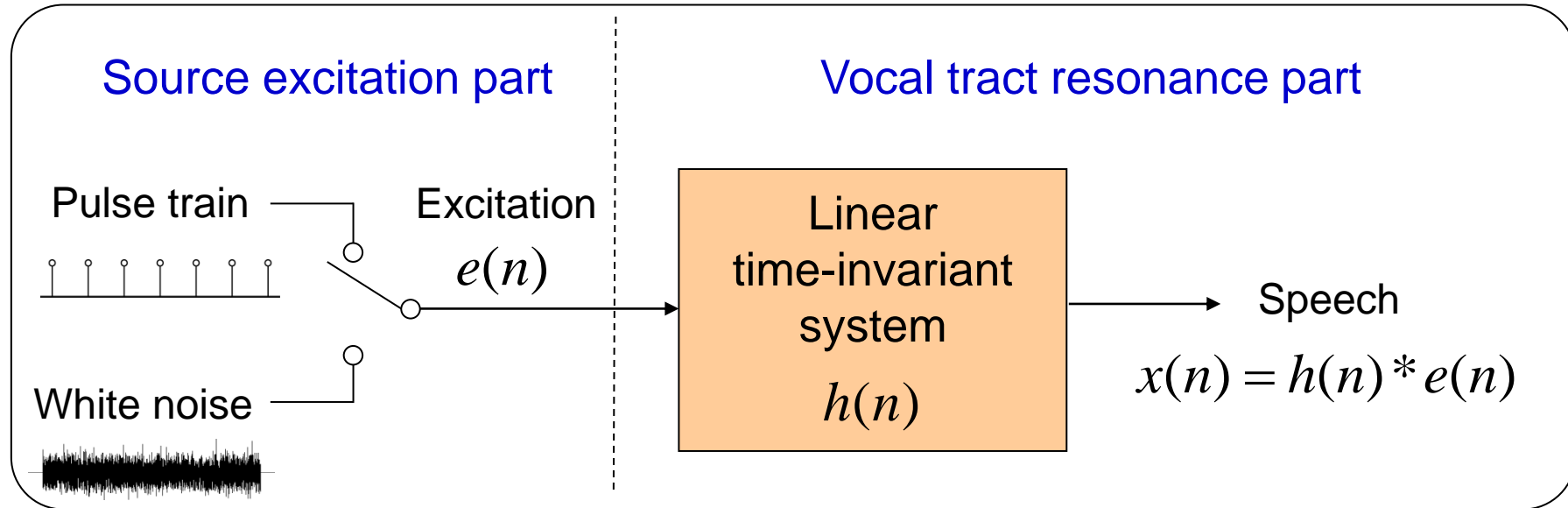
# Information Rate of Speech

- from a Shannon view of information
  - message content/information--2\*\*6 symbols (phonemes) in the language: 10 symbols/sec for normal speaking rate => **60 bps** is the equiv
  - **1000-2000 times change in information rate from discrete message symbols to waveform encoding => can we achieve this three orders of magnitude reduction in information rate on real speech waveforms?**
- from a (wideband hi-fi speech)—need to sample speech at between 8 and 16 kHz, and need about 8 (log encoded) bits per sample for high quality encoding => 8000x8=64000 bps (telephone) to 16000x8=128000 bps (wideband)

# Speech production mechanism



# Source-filter model



$$x(n) = h(n) * e(n)$$

↓ Fourier transform

$$X(e^{j\omega}) = H(e^{j\omega})E(e^{j\omega})$$

# Linear Predictive Coding (LPC)

- LPC methods provide extremely accurate estimates of speech parameters, and does it extremely efficiently
- Basic idea of Linear Prediction: current speech sample can be closely approximated as a linear combination of past samples, i.e.,

$$s(n) = \sum_{k=1}^p \alpha_k s(n-k) \text{ for some value of } p, \alpha_k \text{'s}$$



# LPC methods /1

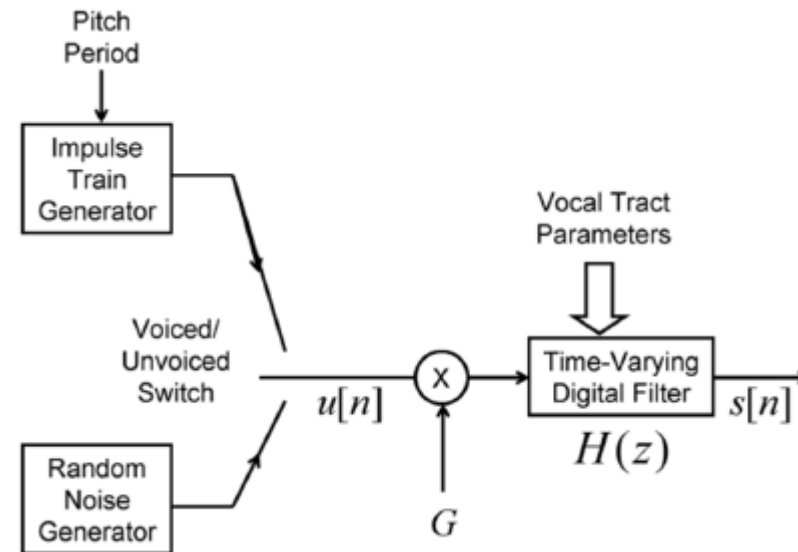
- for periodic signals with period  $N_p$ , it is obvious that

$$s(n) \approx s(n - N_p)$$

- but that is not what LP is doing; it is estimating  $s(n)$  from the  $p$  ( $p \ll N_p$ ) most recent values of  $s(n)$  by linearly predicting its value
- for LP, the predictor coefficients (the  $\alpha_k$ 's) are determined (computed) by **minimizing the sum of squared differences** (over a finite interval) **between the actual speech samples and the linearly predicted ones**

# LPC methods /2

- LP is based on speech production and synthesis models
  - speech can be modeled as the output of a linear, time-varying system, excited by either quasi-periodic pulses or noise;
  - assume that the model parameters remain constant over speech analysis interval



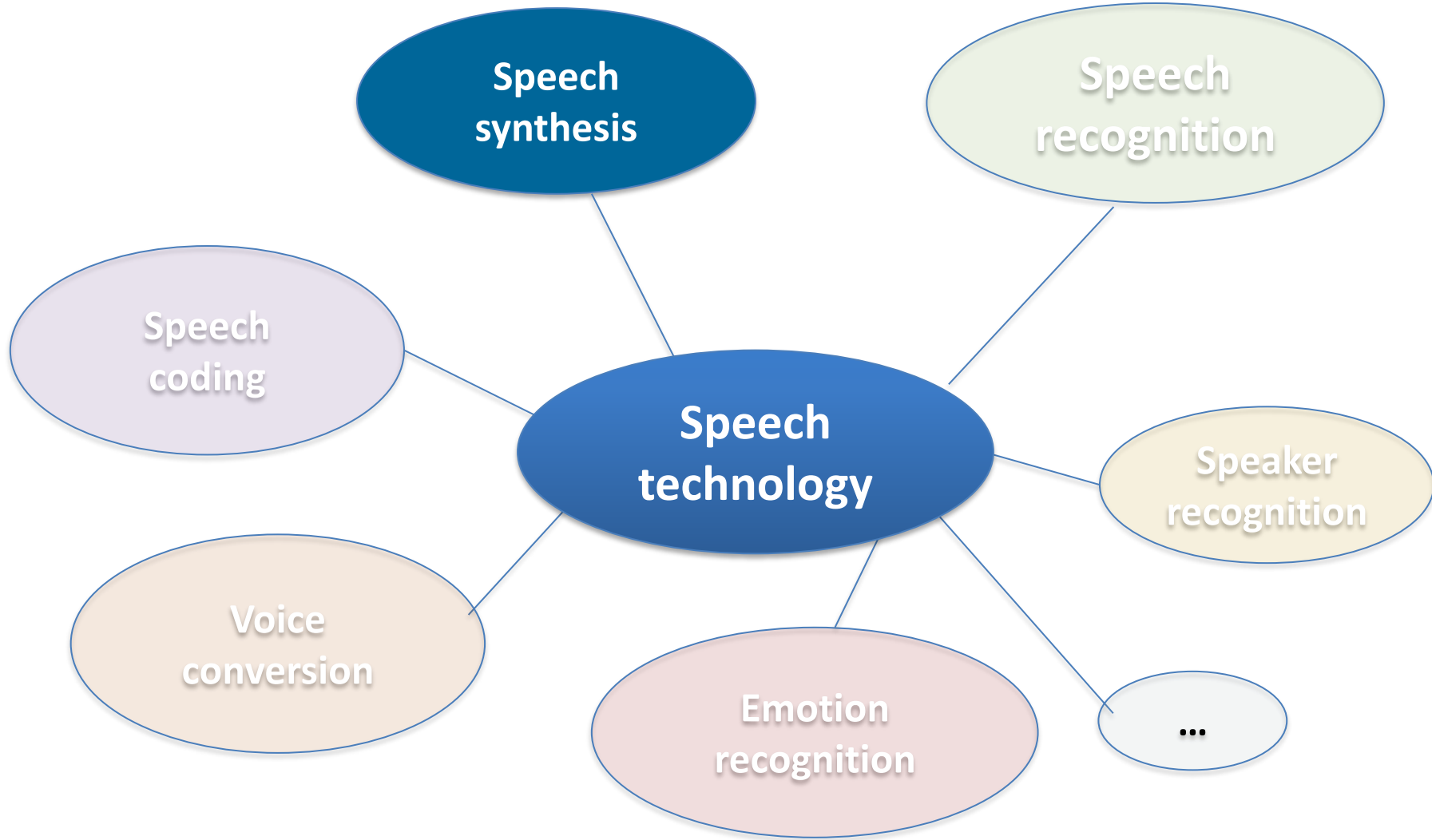
# LPC examples

- Waveform coding
  - Original (64 kbps)
  - ADPCM (32 kbps)



- Linear Predictive Coding
  - CELP (4800 bps)
  - LPC-10 (2400 bps)

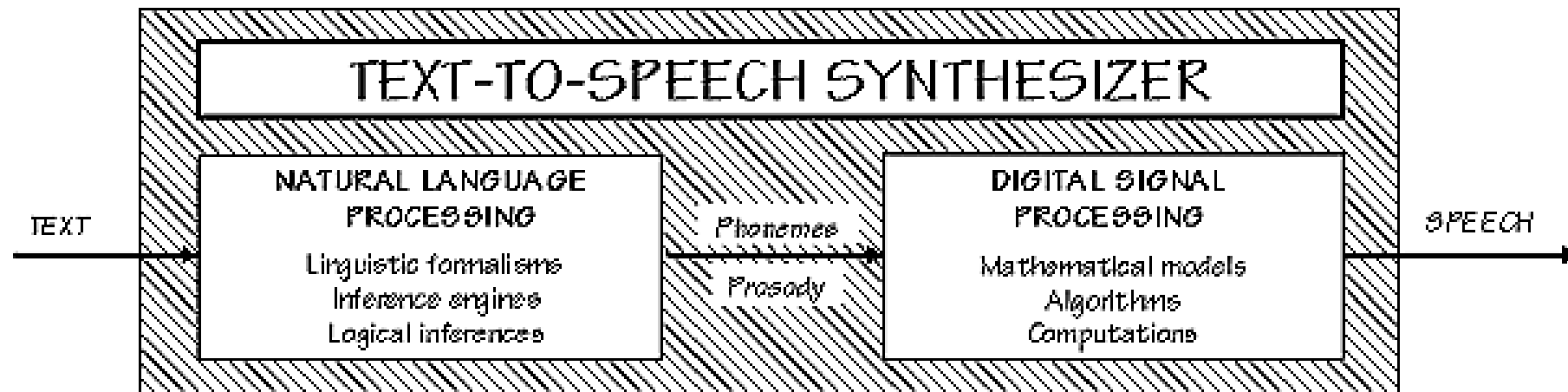




# Text-to-speech synthesis

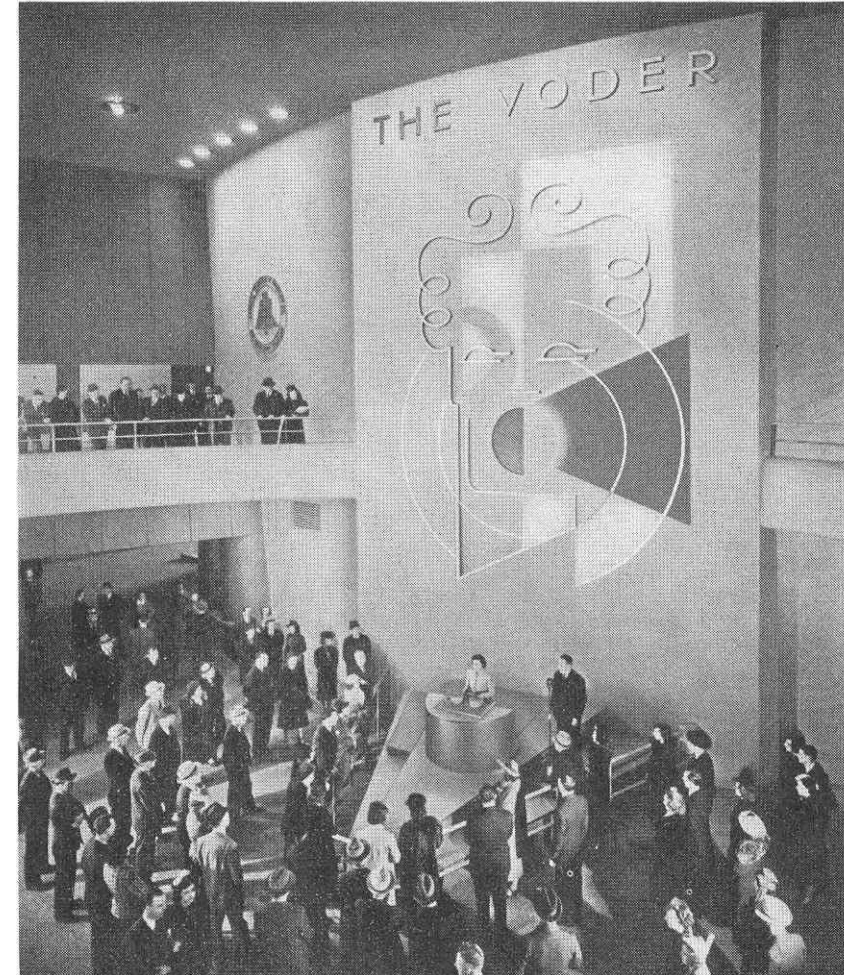
- ***Synthesis of Speech*** is the process of generating a speech signal using computational means for effective human-machine interactions
  - machine reading of text or email messages
  - telematics feedback in automobiles
  - talking agents for automatic transactions
  - announcement machines that provide information such as stock quotes, airlines schedules, weather reports, etc.
  - screen reader for the blind
  - speech communication help for the speaking impaired

# Text-to-speech (TTS)



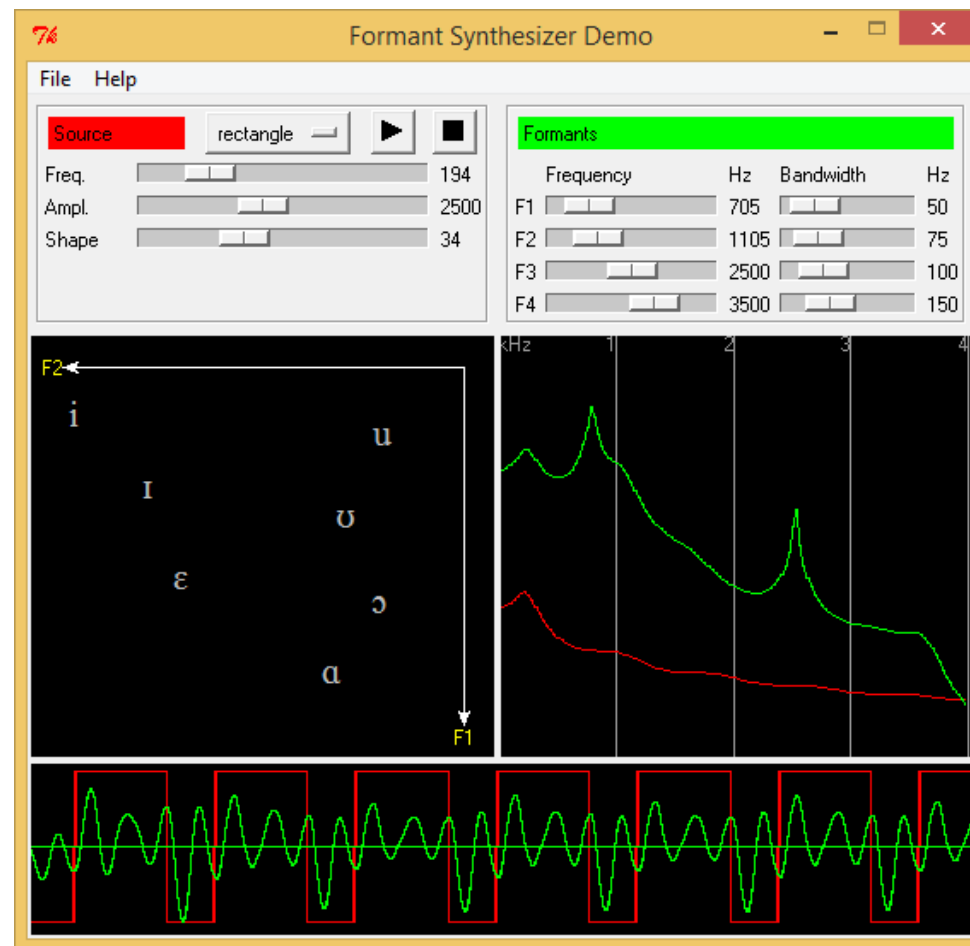
# Speech synthesis - history

- 1939, „Voder”  
electromechanical  
system
- <https://www.youtube.com/watch?v=0rAyrmm7vv0>



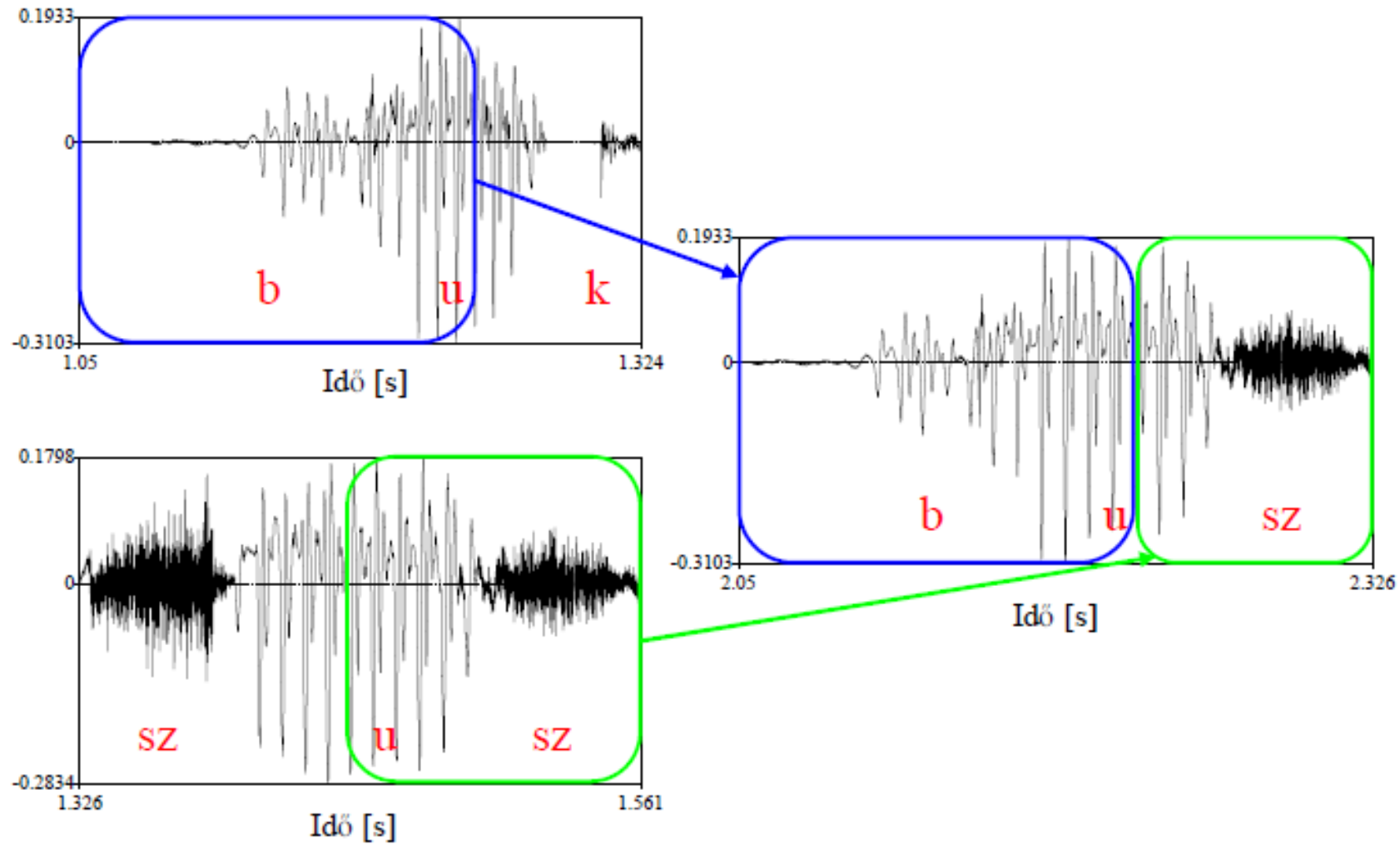
# Formant synthesis

- <http://www.speech.kth.se/wavesurfer/formant/>





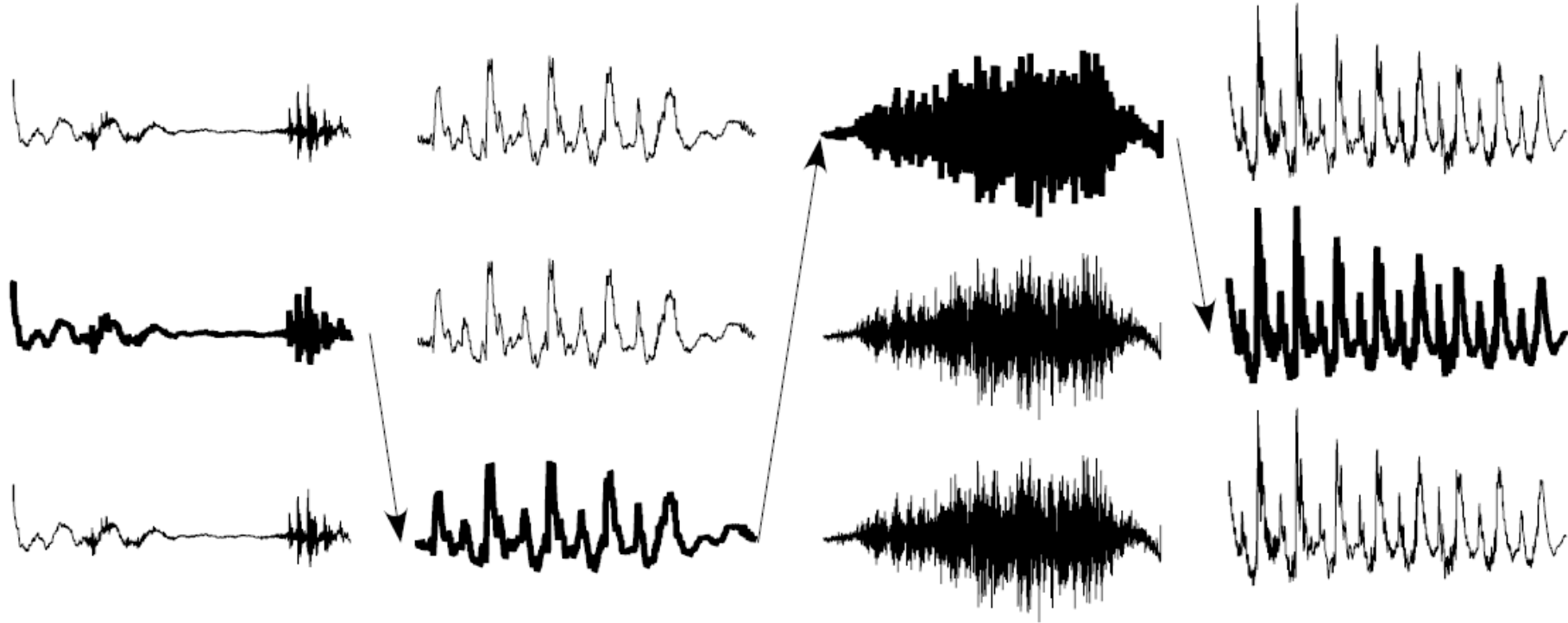
# Diphone concatenation



# Unit selection /1



# Unit selection /2



# Speech synthesis samples

- Formant synthesis ('70s)



- Diphone concatenation ('80s)

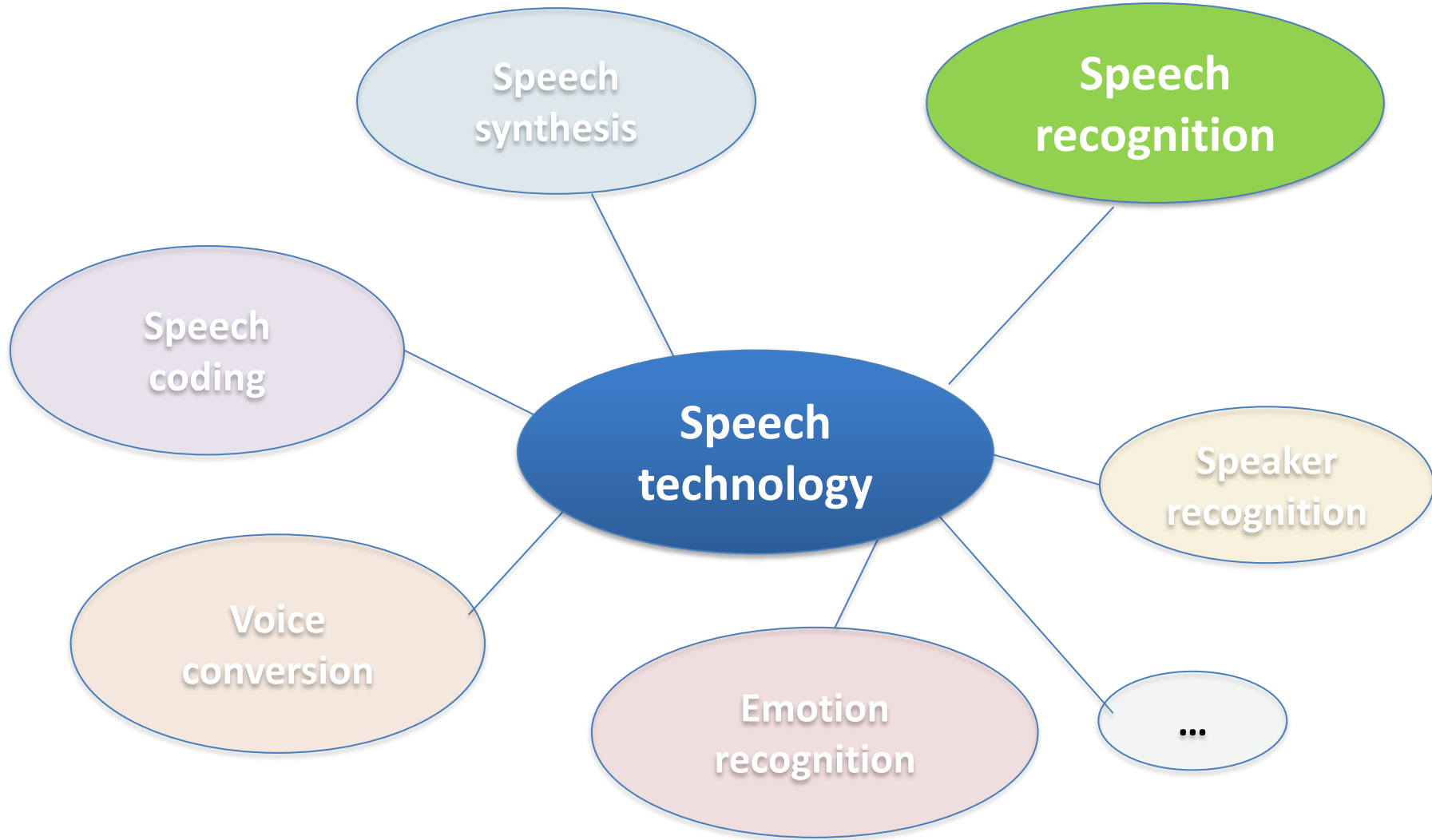


- Unit selection ('90s)

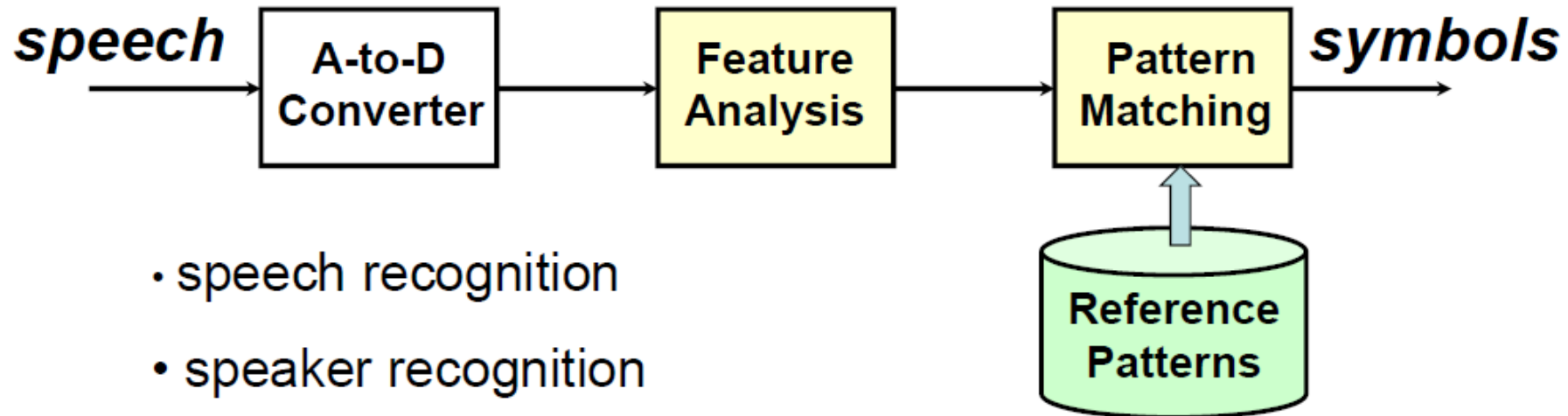


- Statistical speech synthesis (2005-)





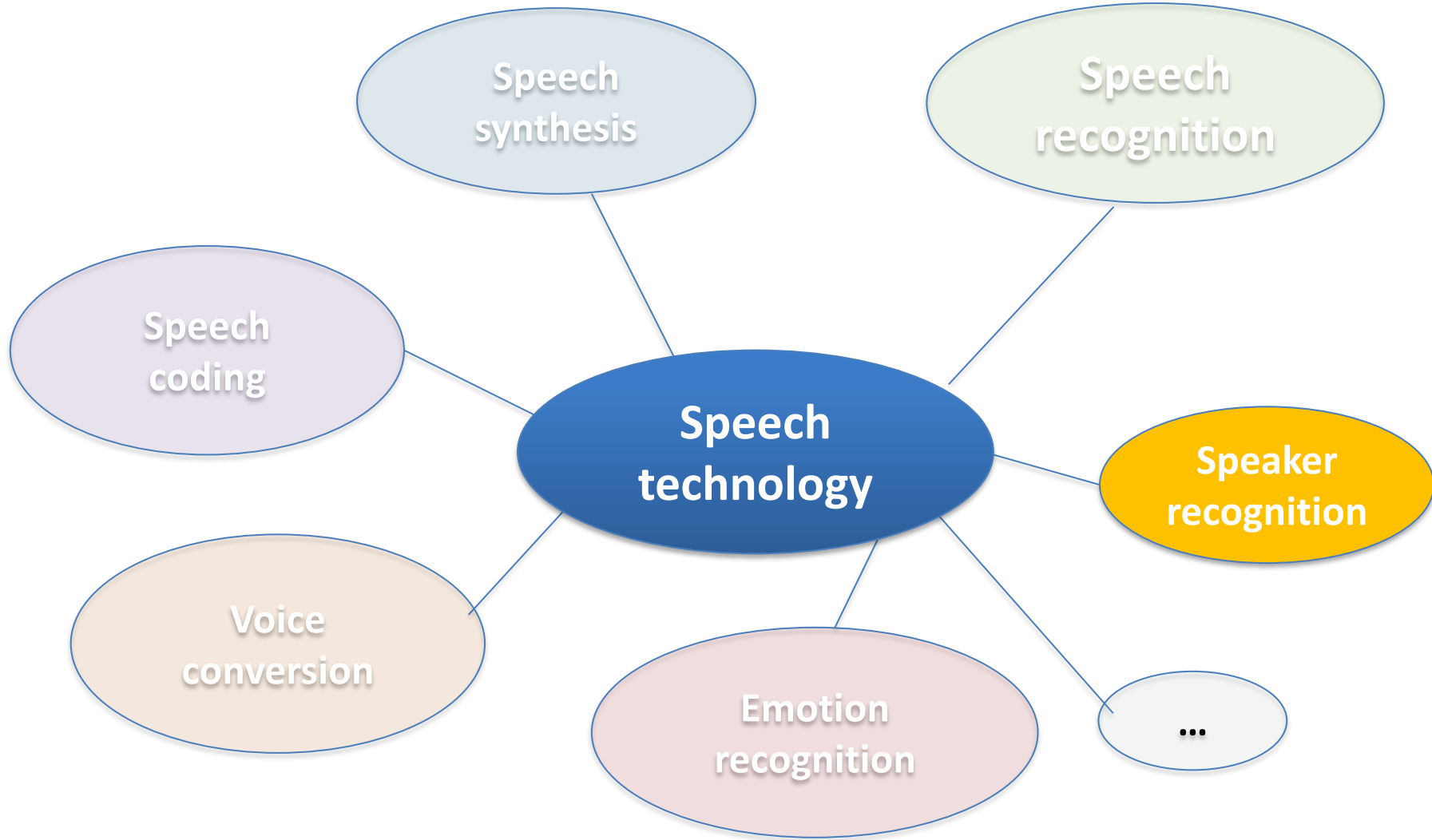
# Pattern Matching Problems



- speech recognition
- speaker recognition
- speaker verification
- word spotting
- automatic indexing of speech recordings

# Automatic Speech Recognition

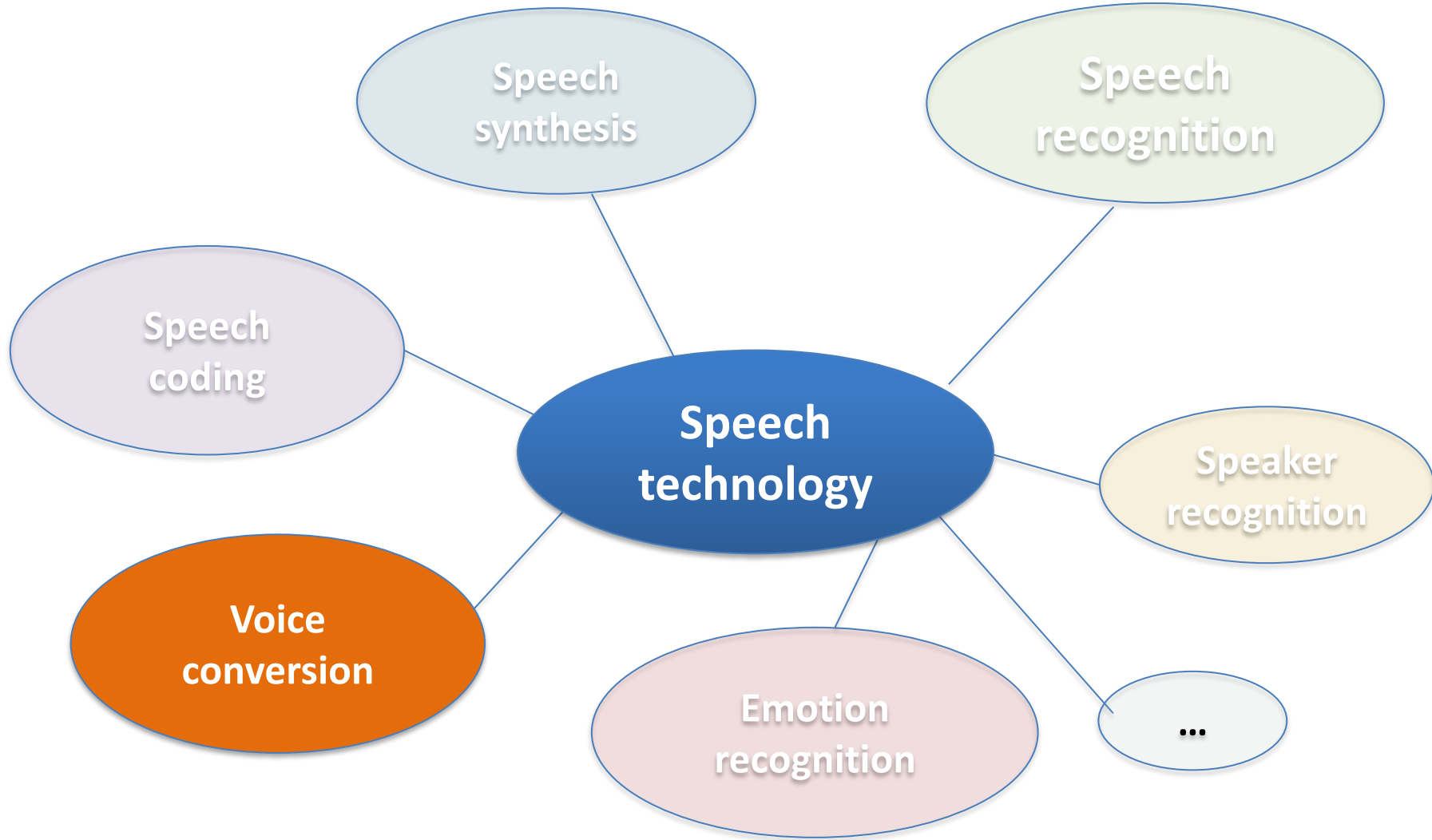
- ***Recognition of Speech*** is the process of extracting usable linguistic information from a speech signal in support of human-machine communication by voice
  - command and control (C&C) applications, e.g., simple commands for spreadsheets, presentation graphics, appliances
  - voice dictation to create letters, memos, and other documents
  - natural language voice dialogues with machines to enable Help desks, Call Centers
  - voice dialing for cellphones and from PDA's and other small devices





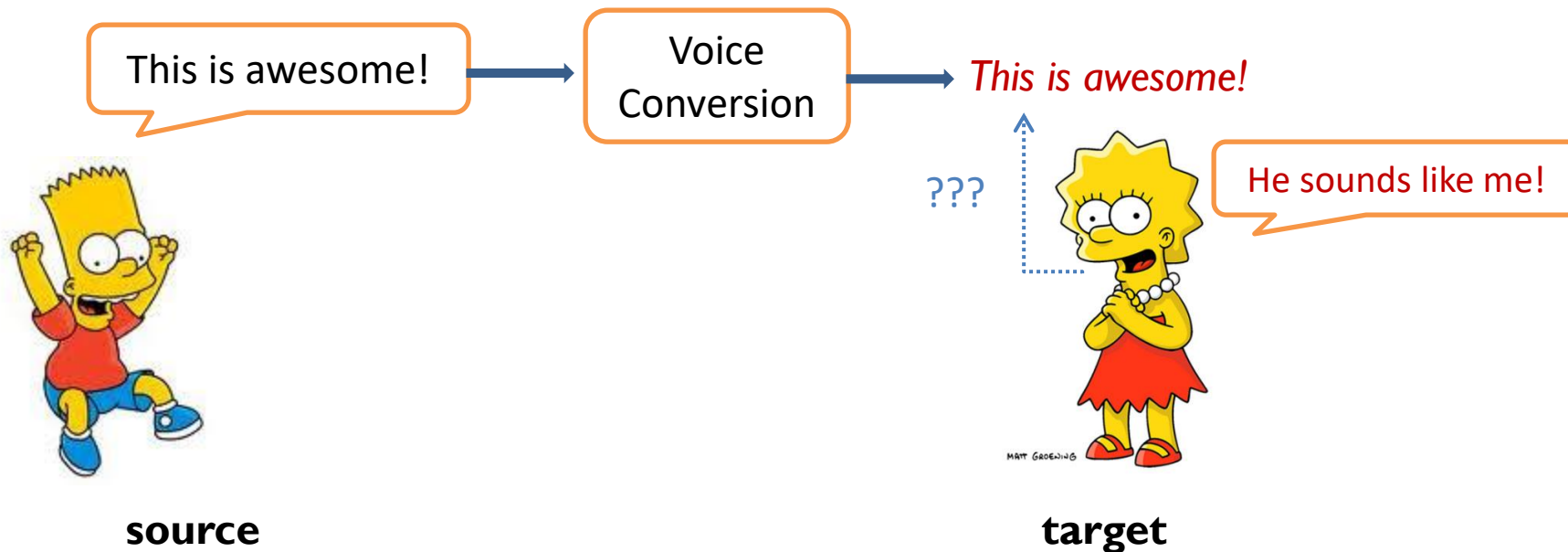
# Speaker verification, recognition

- ***Speaker Verification***
  - secure access to premises, information, virtual spaces
- ***Speaker Recognition***
  - legal and forensic purposes - national security; also for personalized services



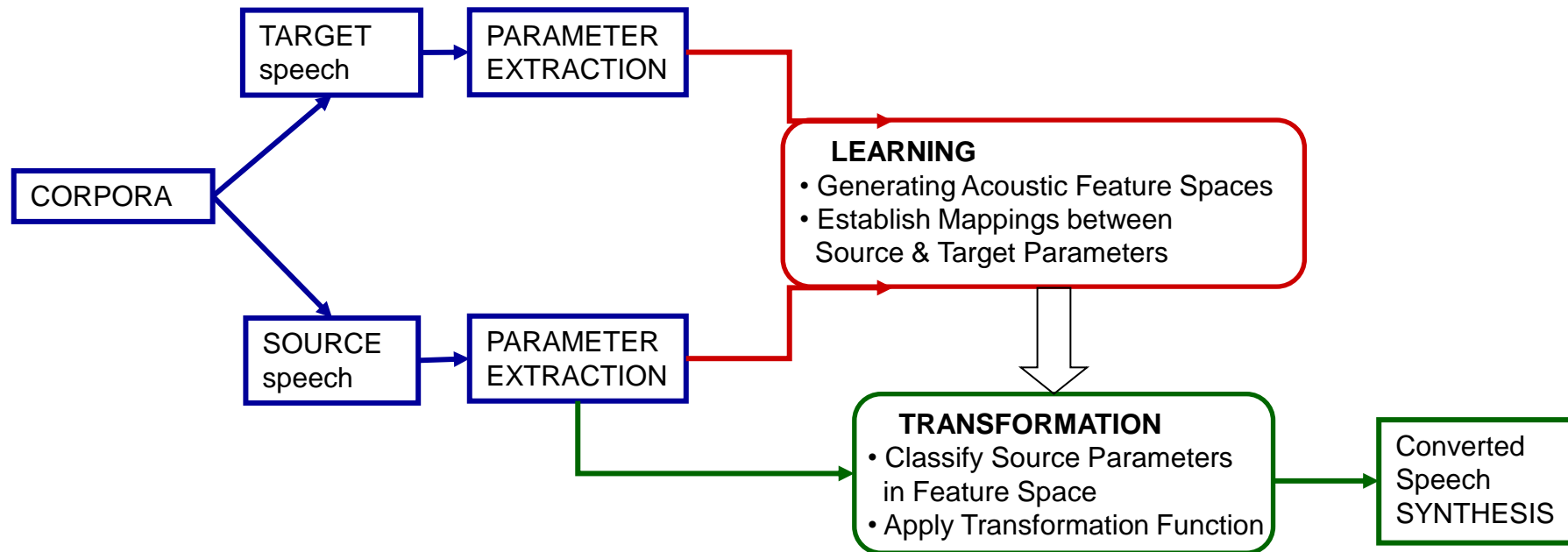
# Voice conversion

- Transform the speech of a (source) speaker so that it sounds like the speech of a different (target) speaker.



# Stages of Voice Conversion

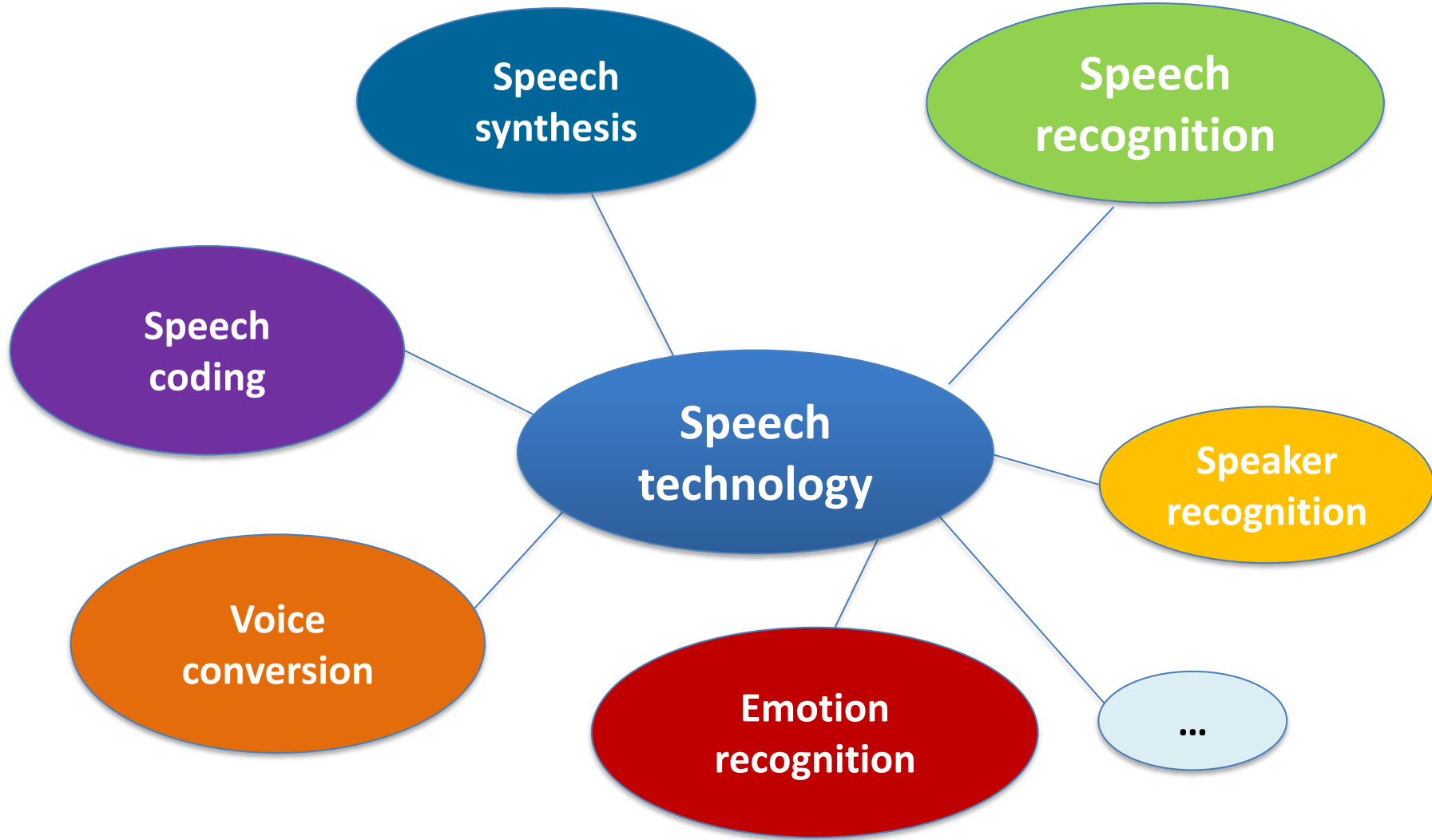
1) Analysis, 2) Learning, 3) Transformation



- Key Parameter: the spectral envelope (relation to timbre)

# Voice conversion examples

	Source	Target	GMM	DFWA	DFWE
<b>slt</b> → <b>clb</b> (FF)					
<b>bdl</b> → <b>clb</b> (MF)			 	 	

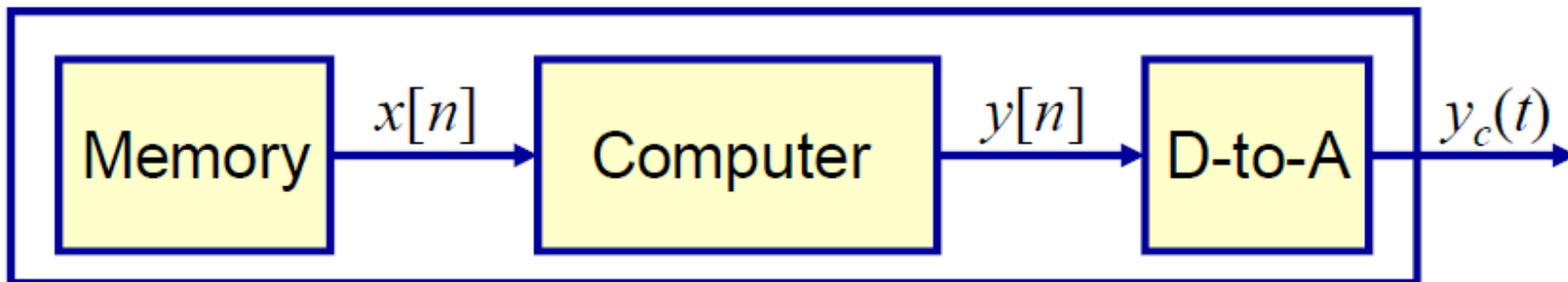


# **PERCEPTUAL CODING OF AUDIO SIGNALS**

# Apple iPod



- stores music in MP3, AAC, MP4, wma, wav, ... audio formats
- compression of 11-to-1 for 128 kbps MP3
- can store order of 20,000 songs with 30 GB disk
- can use flash memory to eliminate all moving memory access
- can load songs from iTunes store – more than 1.5 billion downloads
- tens of millions sold





# Compression

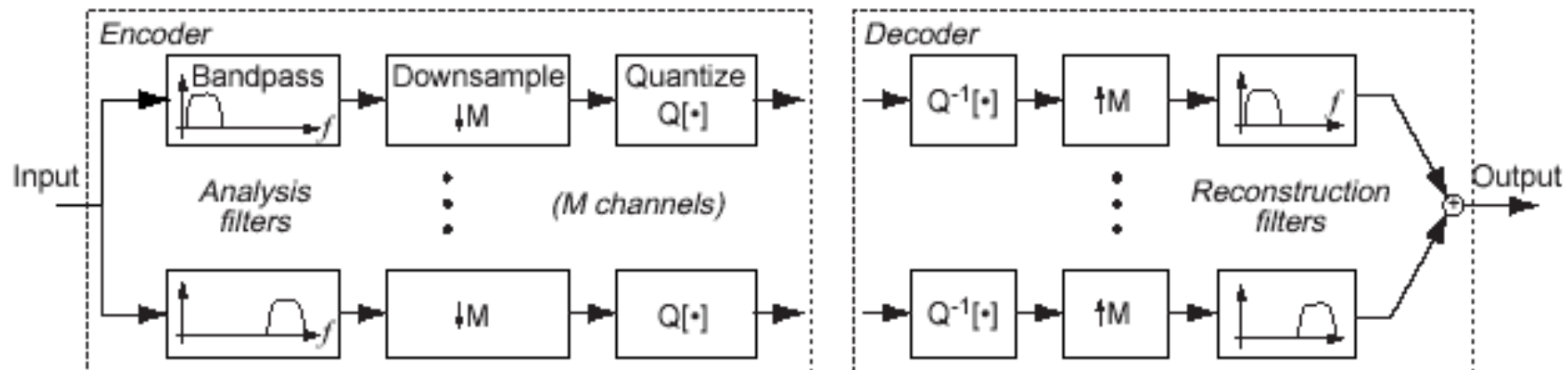
- High data rates, such as CD audio (4.32 Mb/s), are incompatible with internet & wireless applications.
- Audio data must somehow be compressed to a smaller size (less bits), while not affecting signal quality (minimizing quantization noise).
- **Perceptual Audio Encoding** is the encoding of audio signals, incorporating psychoacoustic knowledge of the auditory system, in order to reduce the amount of bits necessary to faithfully reproduce the signal.
  - MPEG-1 Layer III (aka mp3)
  - MPEG-2 Advanced Audio Coding (AAC)

# Perceptual coding

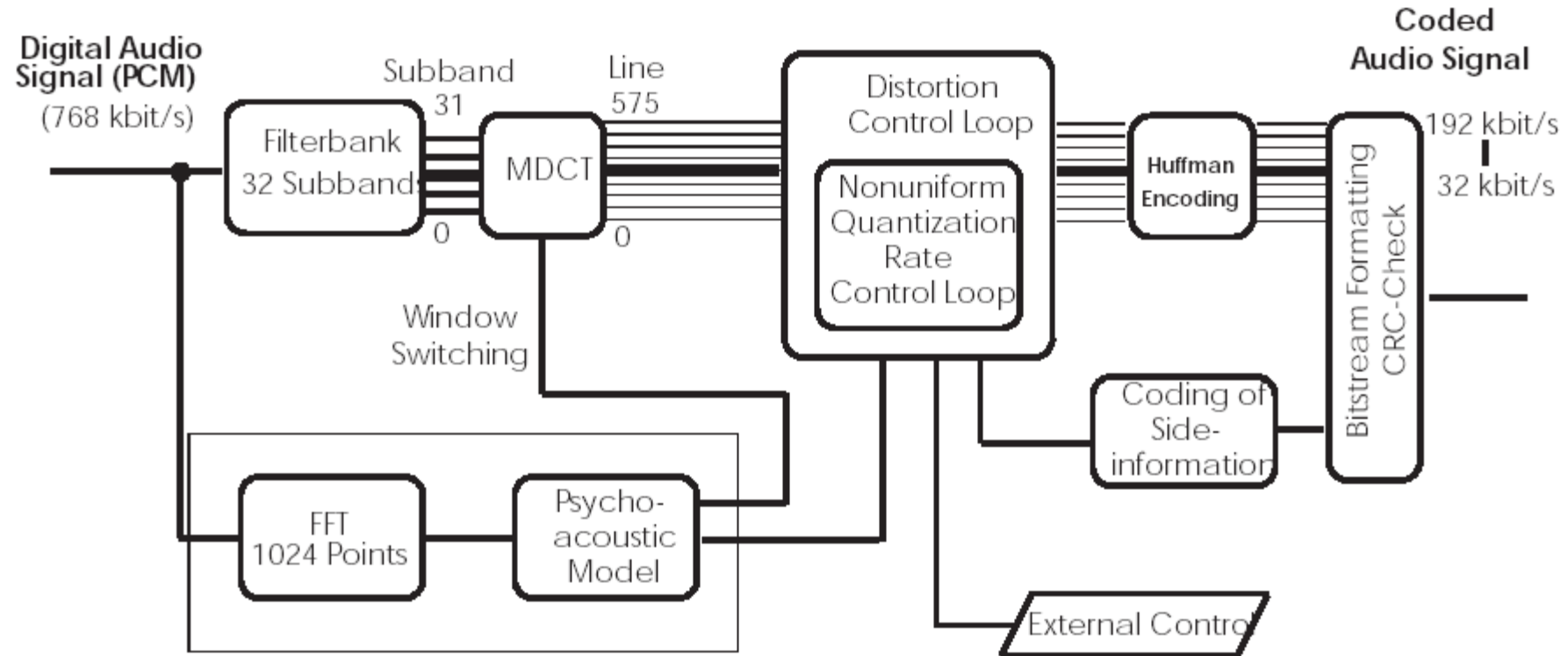
- Goal: compress audio (e.g. music) without quality loss
- Use properties of hearing
  - Critical bands
  - Hearing limitations
  - Masking
    - Time domain
    - Frequency domain

# Subband coding

- Analysis filter bank,  $M$  bandpass filters
- Quantize separately in different bands
  - quantization noise stay within band; gets masked



# MP3



# MP3 Bit Rate vs. Audio Quality

www.xeport.com

Song: You Are Number One

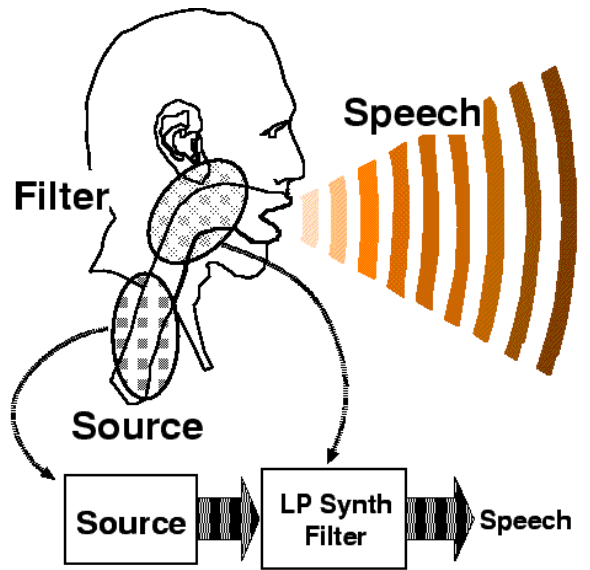
Bit Rate: 320kbps CBR

File Size: 1168kB

Sampling Rate: 44100Hz

Bit Depth: 32 bits

[video](#)



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The END

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