GSM speech coding

Wolfgang Leister
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Norsk Regnesentral

Sources

This part contains material from:

- Web pages Universität Bremen, Arbeitsbereich Nachrichtentechnik (ANT): Prof.K.D. Kammeyer, Jörg Bitzer, Frank Jordan, Volker Kühn
  - http://www.comm.uni-bremen.de/whomes/meyer/gsm_coder.html
- Sten Amundsen (INF-MKT presentation 2003)

GSM

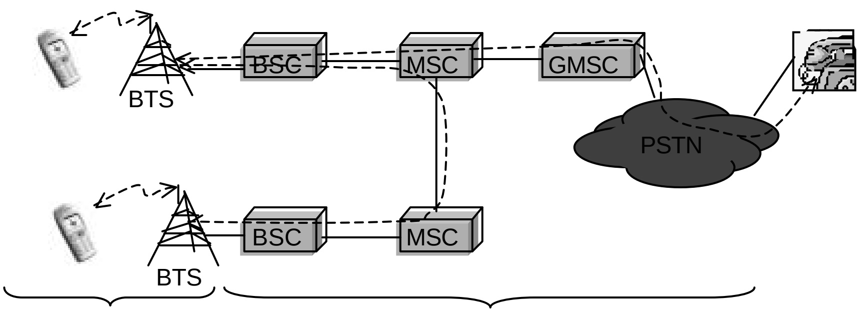
- Global System for Mobile communications,
  - originally developed for Europe
  - has now over 70% of the world market.
- Initially developed for operation in the 900MHz band and subsequently modified for the 850, 1800 and 1900MHz bands.
- GSM = Groupe Speciale Mobile (CEPT committee), 1982
- 1987: Memorandum to implement
- 1991: Start of operation

GSM

- Best possible use of frequencies
- lower bandwidth than PSTN, LAN, WAN, etc: 13kBit/s
- mobile networks suffer from interferences and dropouts
  - but: not disturb end user
- GSM uses block coding
  - PCM and MPEG-1, MPEG-2 uses continuous coding
- Originally GSM is designed for:
  - One channel 16 kbit/s
  - Noise: BER of 10^{-2}
  - Maximum coding delay: 30 ms

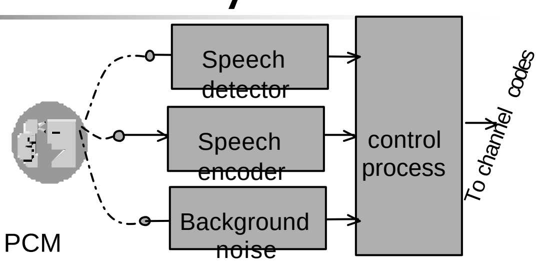
GSM system architecture

- radio access network and base net.
- Speech connection between GSM phone and PSTN or GSM



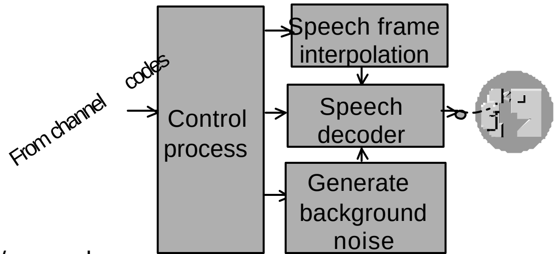
GSM speech coder (RPE-LTP) speech coding similar to PSTN

Sender Functionality



- Speech encoder receives PCM coded speech.
- Speech detector marks frames whether they contain speech.
- Background noise is sent in ”silent suppression” mode.

Receiver Functinality



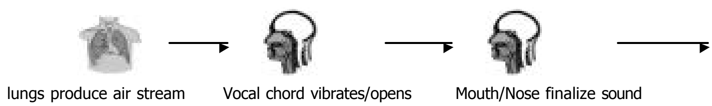
- Speech decoder gets 13 kbit/s speech
- Speech frame interpolation replaces frames lost in transmission (using last received frame)
- Generate background noise using noise frames.

Speech coding

- Speech:
  - bandwidth ca. 12 kHz
  - dynamics: 100 dB (whisper-shout)
  - 20 Bit resolution at 24kHz sampling rate: 480 kBit/s
- Understandable speech:
  - bandwidth 300Hz-3400Hz (telephone)
  - dynamics: 50 dB
  - 8 Bit resolution at 8kHz sampling rate: 64 kBit/s
- Handy: 13 kBit/s
  - coding of speech signal necessary (source coding)

What is speech?

- Speech Generation:



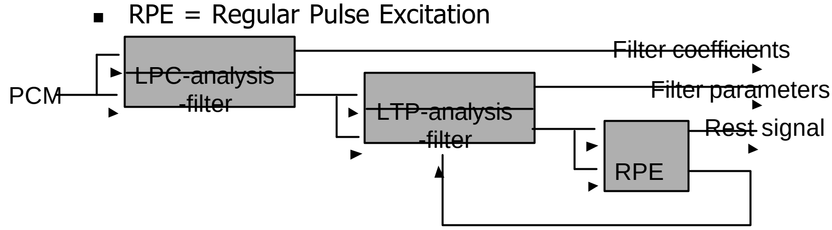
- Artificial Speech Model:

Source → All-Pol Filter → Synthetic Voice →

GSM speech encoder overview

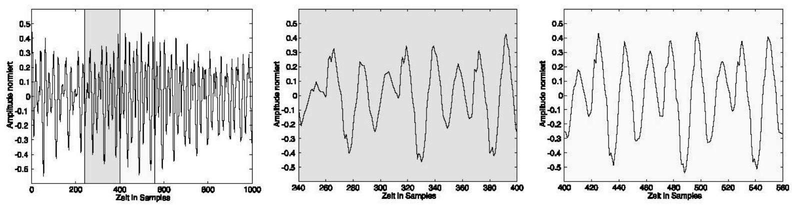
Five Steps:

- AD conversion
- PCM sample blocks
- LPC = Linear Predictive Coding
- LTP = Long-Term Prediction
- RPE = Regular Pulse Excitation



GSM Speech encoder

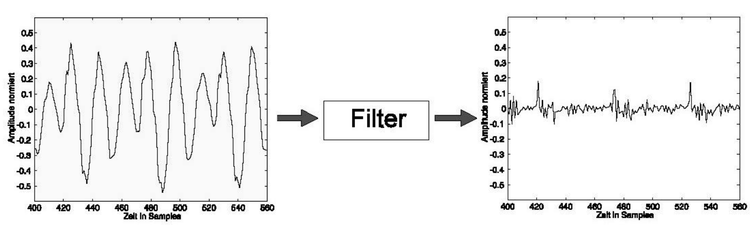
- Step 1: produce digital audio/speech signal
  (microphone, low-pass filter, A/D converter)
- Step 2: Split up in blocks of 20 ms (160 samples)
  statistical parameters do not change in short time intervals
  (assume: short-time stationary behaviour)



GSM Speech encoder

- Step 3: Linear Predictive Coding
  Find filter coefficients with parametric estimation, e.g., Burg Algorithm
  Find reflection coefficients for lower data rate, quantize logarithmically
  Filter signal with quantized coefficients
  LPC uses 36 Bit/160 Samples transfer rate

Parameter	1	2	3	4	5	6	7	8
Bits	6	6	5	5	4	4	3	3
Wert	-0.932	0.735	0.145	0.69	-0.16	0.36	-0.2	-0.4

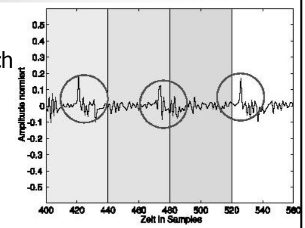


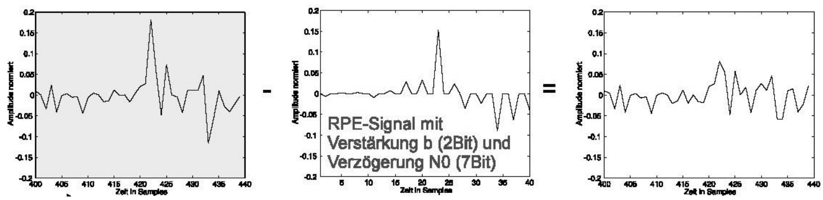
GSM Speech encoder

- Step 4: Long-Term Prediction

  Split blocks into four parts of 40 samples each
  There are still similarities in the encoding!

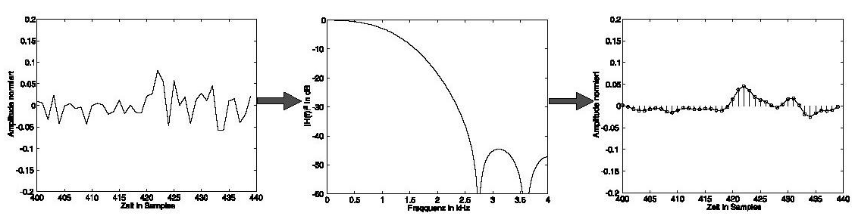
  For each block calculate difference to
  RPE from last step (cross-correlation)
  Calculate translation N0 and amplification b
  LPT uses (2+7)*4 Bit/160 samples transfer rate





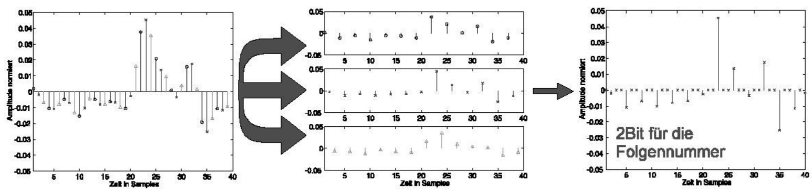
GSM Speech encoder

- Step 5: Regular Pulse Excitation
  - (a) Low pass filter with linear phase TP of degree 10



GSM Speech encoder

- Step 5: Regular Pulse Excitation
  - (b) Split rest signal into 3 polyphases
  - chose polyphase with largest energy



  - (c) Normation of max value of chose sequence (6bit)
  - quantize these 13 values with 3 bit linear

  - RPE uses (2+6+13*3)*4 Bit/160 samples transfer rate

GSM transfer rate

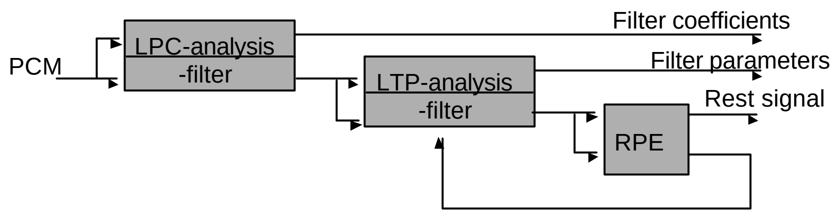
For 160 samples we need:

LPC filter coefficients		36	Bit
LTP filter coefficients	(7+2)*4	36	Bit
Index and Norm Step 5	(6+2)*4	32	Bit
Rest signal	13*3*4	156	Bit
		260	Bit

Data rate: 260bit/20ms = 13kbit/s

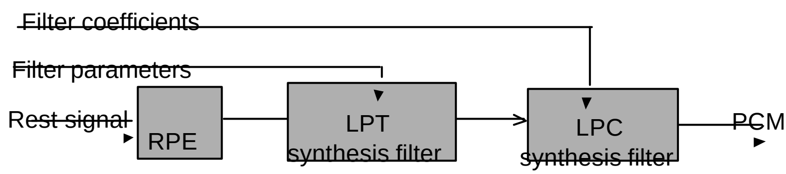
GSM speech coding

- Encoder consists of following parts:
  - Short term LPC (Linear Predection Coding) -analysis
  - Short term LPC-filter
  - Long term LTP (Long Term Prediction) -analysis
  - Long term LTP-filter
  - RPE (Regular Pulse Excitation) position and coding of rest signal



LPE-LPT speech decoder

- Decoder consists of three parts:
  - RPE decoding and position
  - LTP syntesis filter
  - LPC short time synthesis filter



GSM Sound examples

- Original
- GSM full-rate 13 kBit/s, error-free transmission
- Enhanced GSM full-rate 12,2 kBit/s, error-free transmission
- GSM half-rate 5.6 kBit/s, error-free transmission
- old Vocoder 5 kBit/s (no standard)

	original	GSM full-rate, 13 kBit	enhanced GSM full-rate, 13 kBit	GSM half-rate, 5.6 kBit	Old LPC-Vocoder (5kBit/s)	
Male voice, english, 8 kHz, 16 Bit linear						
Male voice, german, 8 kHz, 16 Bit linear						
Music sample, 8 kHz, 16 Bit linear						

Future Development

- GSM is an ETSI standard
- RPE-LPT standardised for GSM by ETSI
- Three RPE-LTP speech coder standards:
  - FR: Full rate: 13 kbit/s
  - HR: Half rate: 5.6 kbit/s (frequency usage better, worse speech quality)
  - EFR: enhanced full rate: 12.2 kbit/s (better speech quality than FR)
- New specifications by ETSI
- Defines dynamic way to change speech encoders

End of Part

Thank you for your attention!