IoT, Course introduction

Internet of Things a.a. 2021/2022

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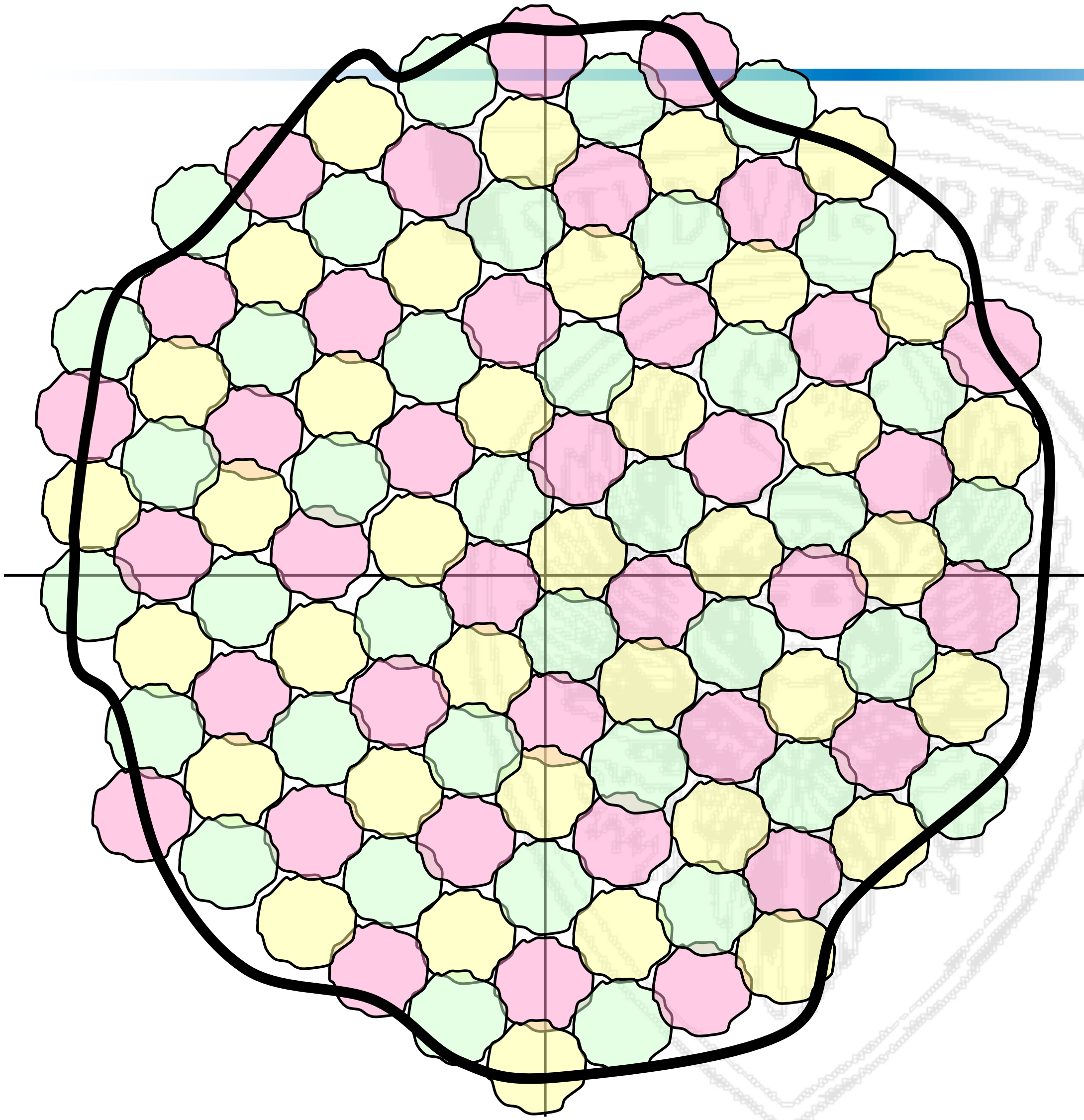
3.5 – Procedures

Cellular systems & GSM
Wireless Systems, a.a. 2021/2022

Mobility management

Cellular coverage (microcells)



many BS

Very low power!!

Unlimited capacity!!

Usage of same spectrum

(12 frequencies)

(4 freq/cell)

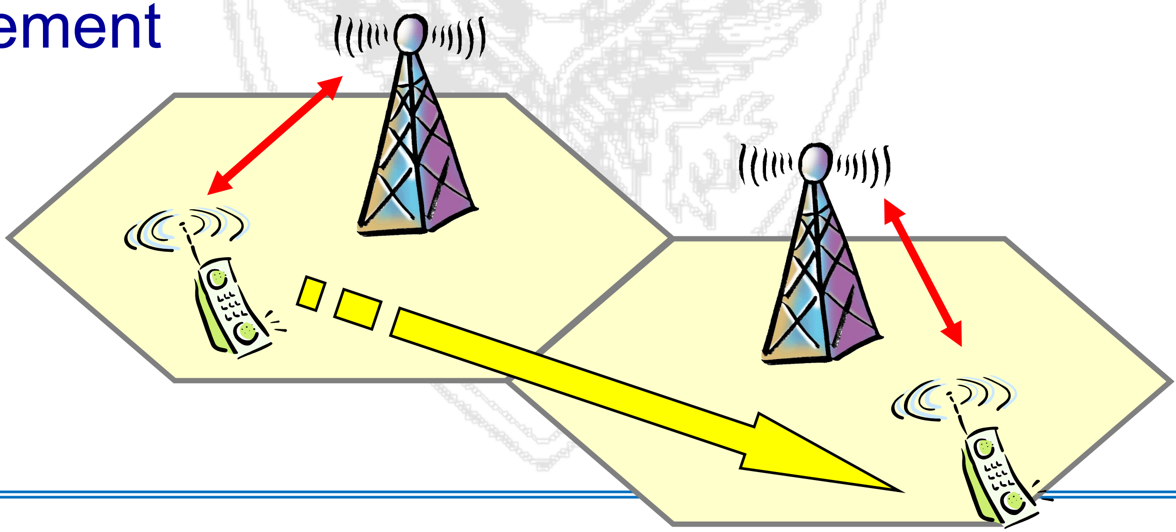
Disadvantage:

mobility management

additional infrastructure costs

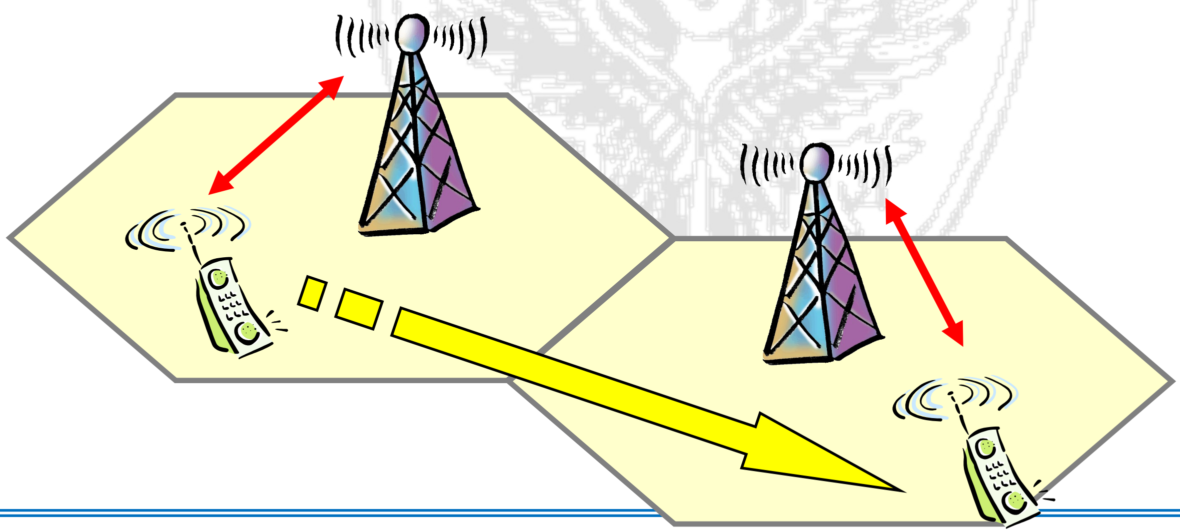
Mobility management

- cellular networks, users can move around in the system and then move from one cell to another
- This obviously poses problems of routing information (or more simply of the calls in the case of voice service)
- All procedures that the network puts in place to enable mobile users to be reached by a communication and maintain active communication even in the presence of a change of the cell go under the name of mobility management



Mobility management

- Users of cellular systems WHILE MOVING can:
  - call out
  - be called
  - converse
- And there should be some "intelligence" that supports this.



Mobility management

- In the case of circuit service mobility management procedures differ depending on whether the user is moving in the IDLE state (no active circuit) or ACTIVE (in conversation)

- ACTIVE: there is an active circuit that needs to be re-routed after every change of the cell (handover)

- IDLE: the user must be able to be located to be able to establish a call to/from him/her (*Location Update, Cell Selection, Cell Reselection*)

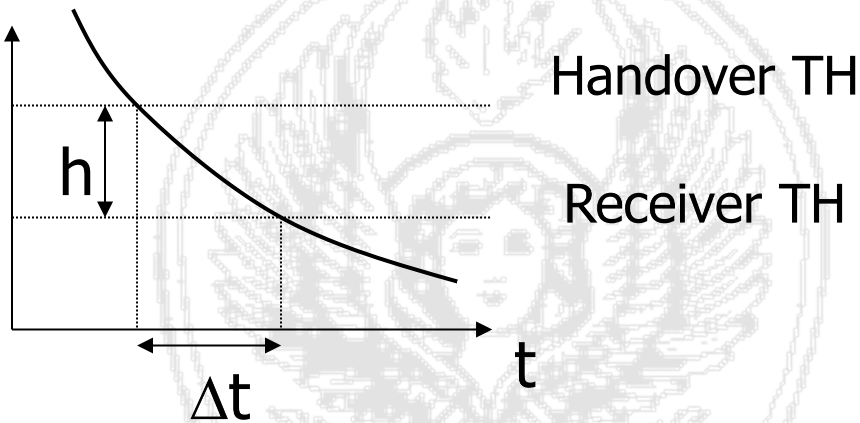
Handover

- Procedure by which a mobile terminal in conversation changes the base station it is affiliated to
- In network-controlled handoff and mobile assisted handoff (NCHO and MAHO) the procedure is always initiated by the network, on the basis of measurements (received signal strength, quality, etc.) carried out by both the network and the user side
- Handover procedures must be efficient and fast
- We will see in the case of GSM how handover procedures are managed from the point of view of network signaling and of the routing of the circuit

Handover

When to trigger an handover?

- The choice of the thresholds of activation of the handover procedure is a critical factor



- If h is too small It is too small and you risk loosing the connection
- If h is large the number of requests for handover increases, so also the signaling traffic in the network

Handover

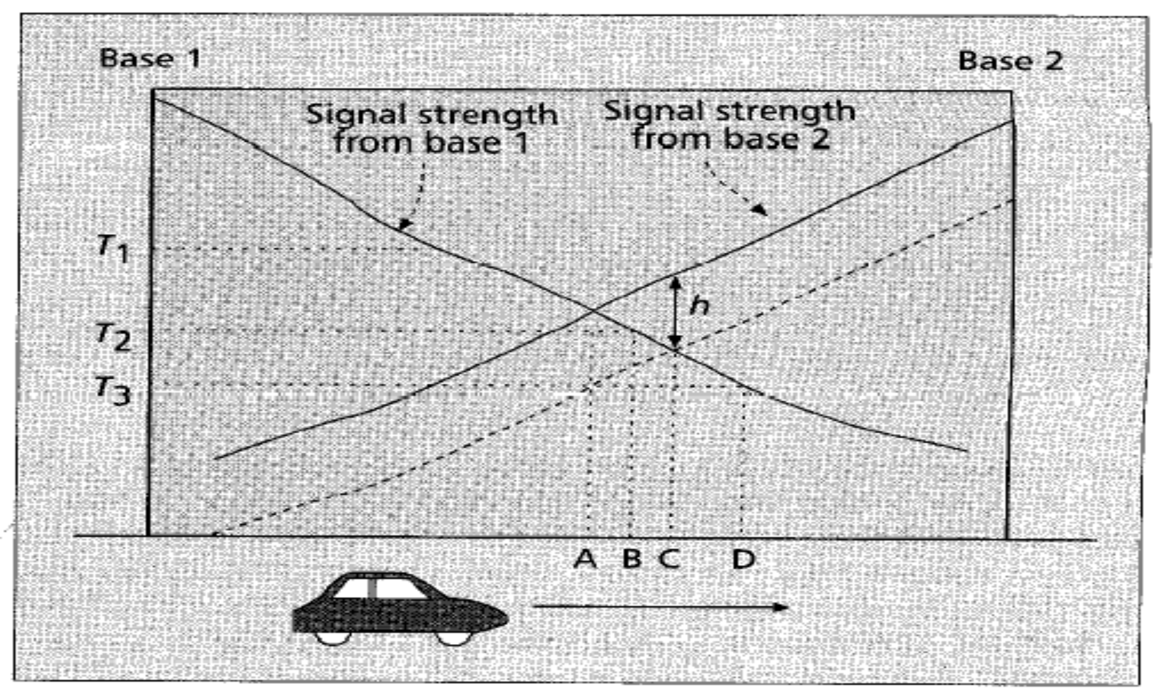
When to trigger an handover?

- There are several methods

1 - method of the strongest signal

– the handover occurs at point A

due to the fluctuations of the signal many cell changes are possible (ping-pong effect)



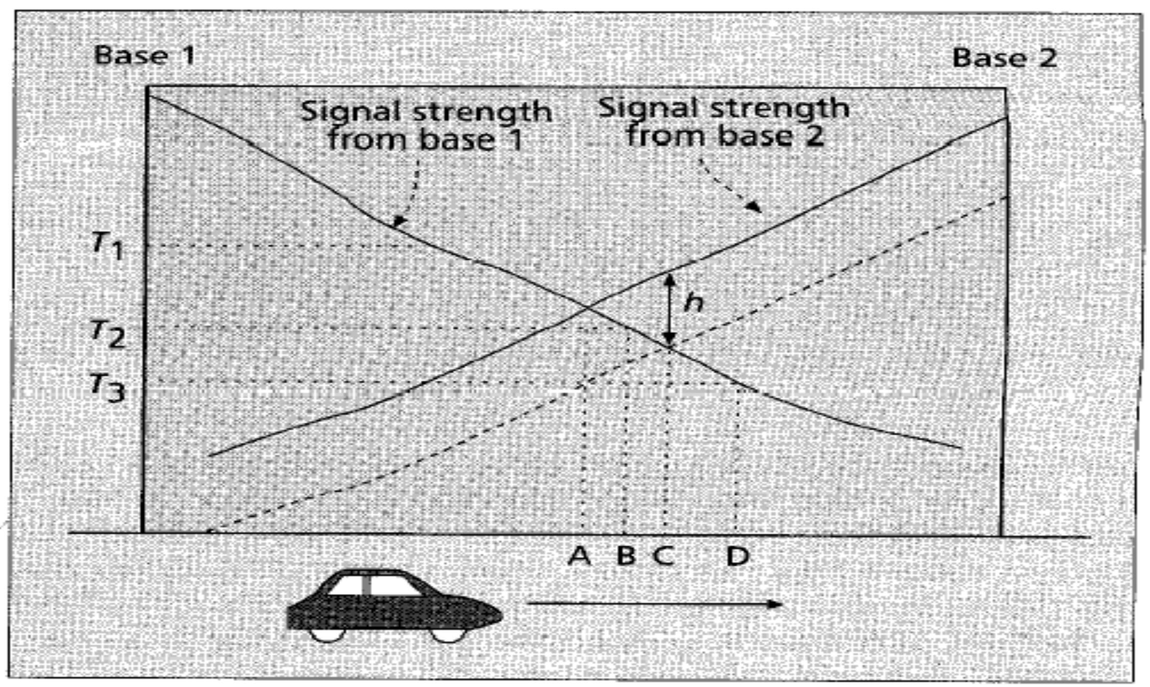
Handover

When to trigger an handover?

- There are several methods

2 - method of the strongest signal with the threshold

- if the signal received from the previous BS is less than a threshold (as. eg. T2) and the power of another BS is stronger; the handover occurs at point B



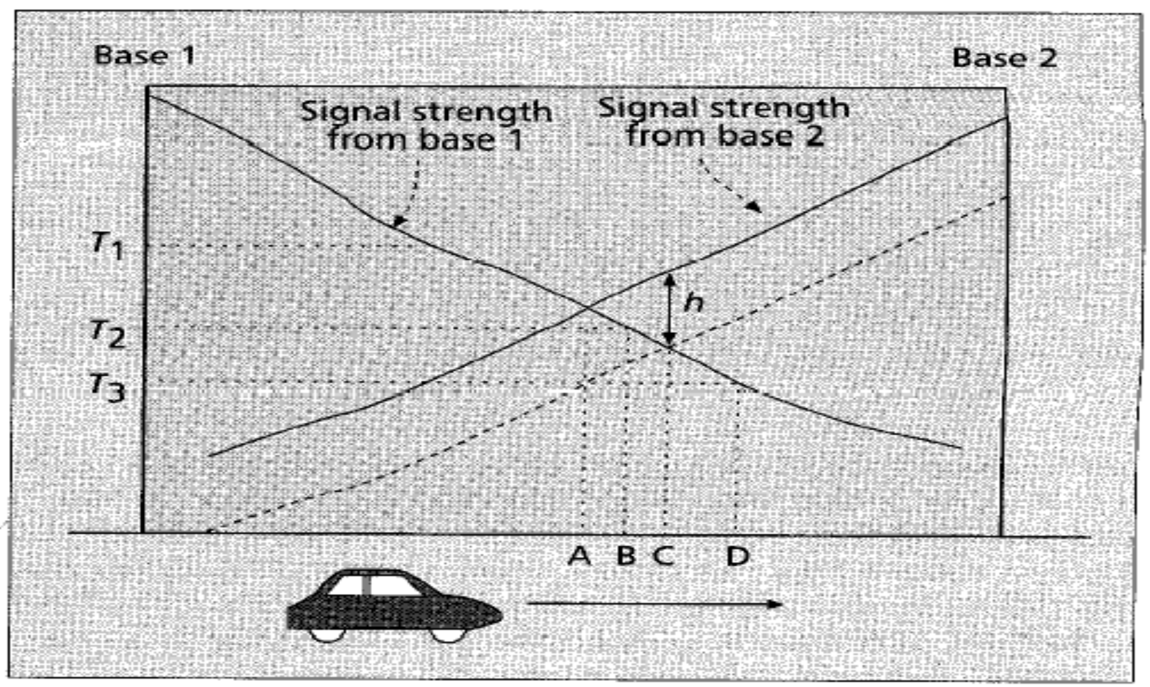
Handover

When to trigger an handover?

- There are several methods

3 - method of the strongest signal with hysteresis

- – if the power of the other BS is stronger than a value of h; the handover occurs at the point C



Handover performance

- When there is a handover the channel in the old cell is released and the new channel is requested;
  - Problem: a channel in the new cell may not be available
- We define the probability of rejecting an handover (Pdrop) as the probability that a handover request can not be met and the blocking probability (Pblock) as the probability of rejecting a new call
- In systems that deal with requests for handover as the new incoming requests (call setup) Pdrop = Pblock
- In fact it is better to block an incoming call that loosing one active
- You can think of better treat requests for handover

Handover performance guard channels technique

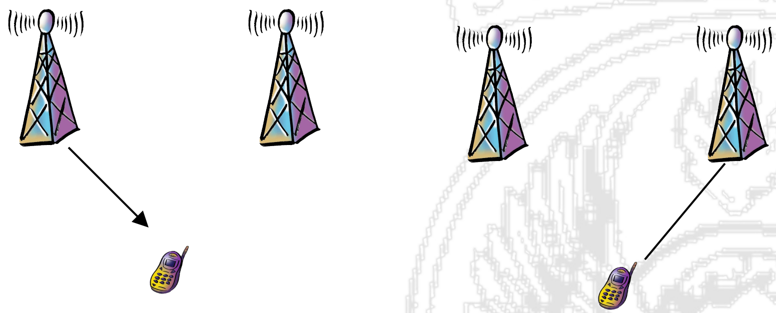
- Guard Channels
  - A number of channels is reserved for handover requests
  - Pdrop becomes lower but the capacity of the system is lower
  - System dimensioning is critical and requires accurate estimates of the traffic dynamics (how many channels should I reserve for handover requests?)

Mobility management

- Other Options
  - Queuing priority scheme
    - ✓ Handoff area: area within which the MS can hear both base stations. If no channels are available in the new BS the user will continue to be interconnected to the old BS; the request for handover to the new BS is buffered and served as soon as a channel is freed.
  - Subrating scheme
    - ✓ If there are no channels available at the new Base Station a channel previously allocated to a call is divided into two channels each half rate, allowing both calls to go forward.

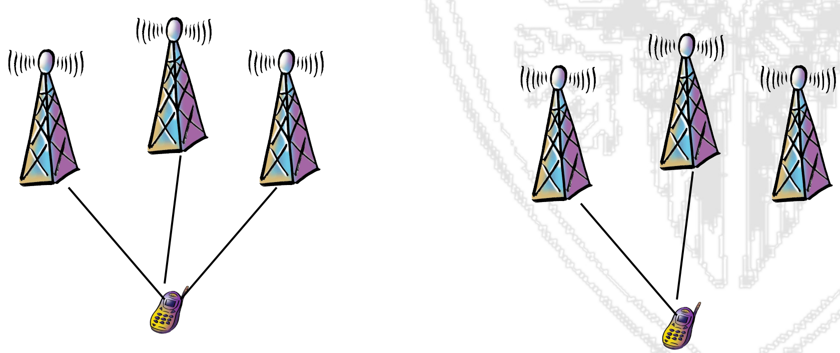
Types of handover

- Hard Handover (GSM-2G)



Removal and establishment of a new radio link

- Soft Handover (UMTS-3G)



Leveraging on the user macrodiversity, the user is simultaneously connected to several base stations

Handover Procedure

Handover e GSM

- The handover procedure is initiated by the network, based on measurements provided by the MS
- When the MS connects to a cell, the BSC sends to it a list of “alternative channels” (the BCCH of 6 adjacent cells) whose signal strength should be monitored by the MS;
- The results of such measurements is transmitted by the MS to the BSC using the SACCH channel every 480 msec
- An handover may be started by the BSC based on measurements performed by both the MS and the BTS

Handover parameters

- The procedure requires
  - A set of rules to determine whether an handover is necessary
  - Dedicated procedures to commute the communication from the original radio channel to the new channel
- It should be transparent to the user

Handover parameters - MS

- Signal strength on the BCCH carrier of adjacent cells (RXLEVNCELLn)
- Signal strength on the active TCH channel (RXLEV)
- Quality of the active TCH channel (RXQUAL)

Handover parameters - BTS

- Signal strength from the MS on the traffic channel (RXLEV)
- Quality of the traffic channel from the MS (RXQUAL)
- Distance of the MS (Timing Advance)

Reasons for handovers

- Low quality transmissions (RXLEV and/or RXQUAL below threshold)
- The distance between the MS and the BTS is above a given threshold (timing advance)
- Motivated by traffic (high load on the cell)
- Control and maintenance

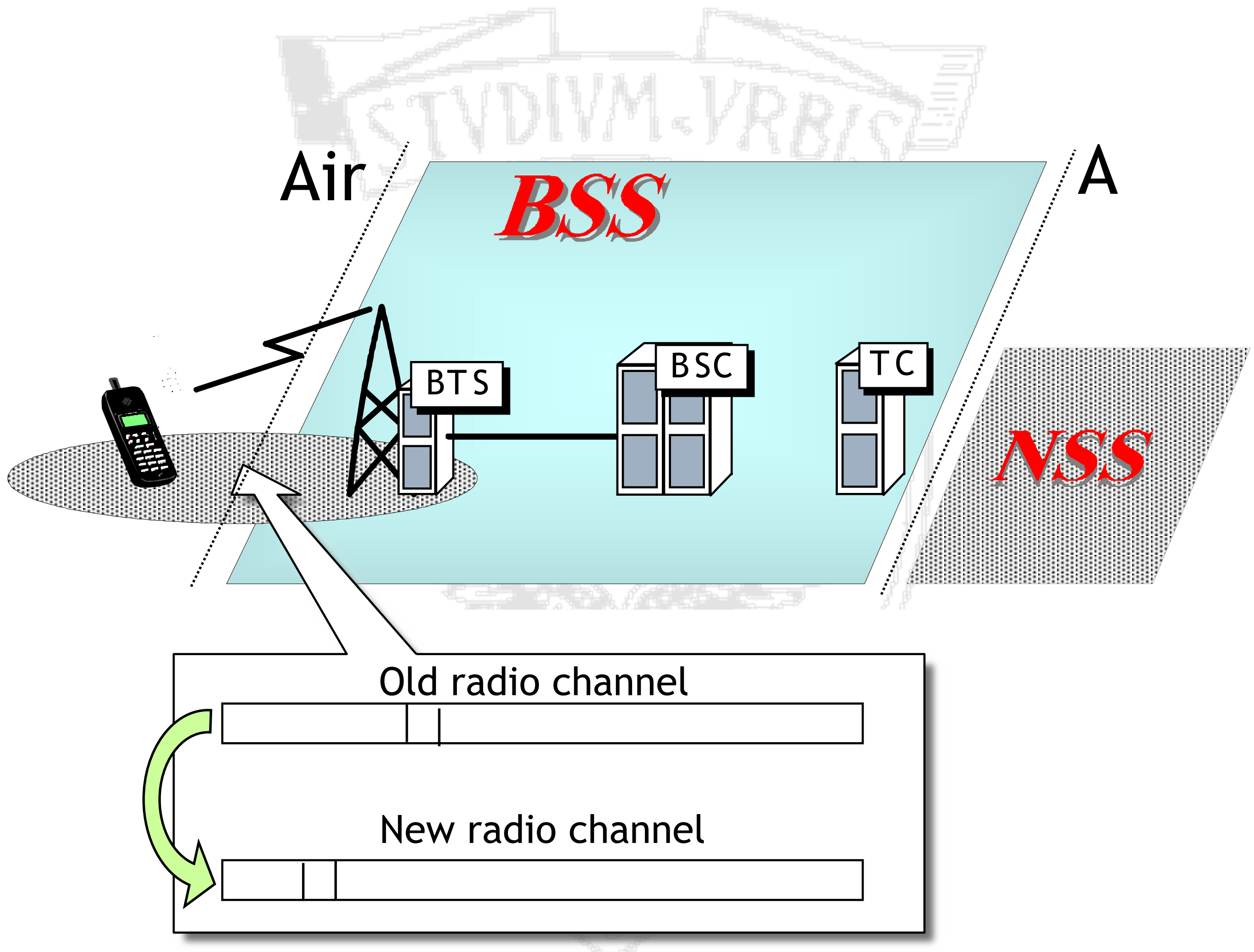
Types of Handovers

4 types of Handovers

- Intra Cell - Intra BSC
- Inter Cell - Intra BSC
- Inter Cell - Inter BSC
- Inter MSC

Handovers must be performed quickly! (<=100 ms)

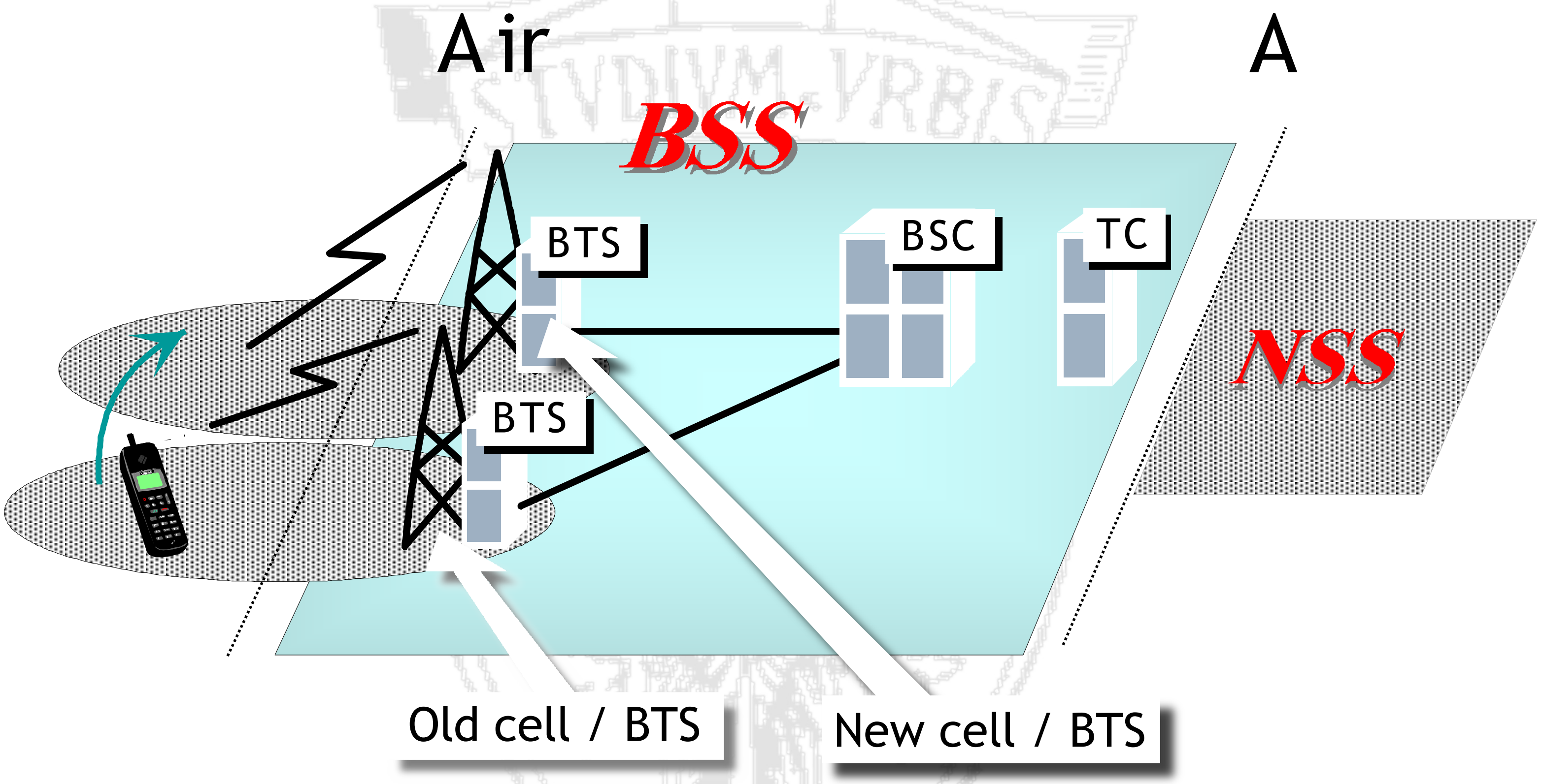
Intra Cell –
Intra BSC Handover



Intra Cell – Intra BSC Handover

- Simpler handover, decided by the BSC only
- A new traffic channel is allocated, usually the frequency within the BTS is modified as well
- Triggered by:
  - Low-quality TCH, high received signal strength
  - No adjacent BTS can provide better quality

Inter Cell – Intra BSC Handover



The MS moves to a new cell under the same BSC

Inter Cell-
Intra BSC Handover

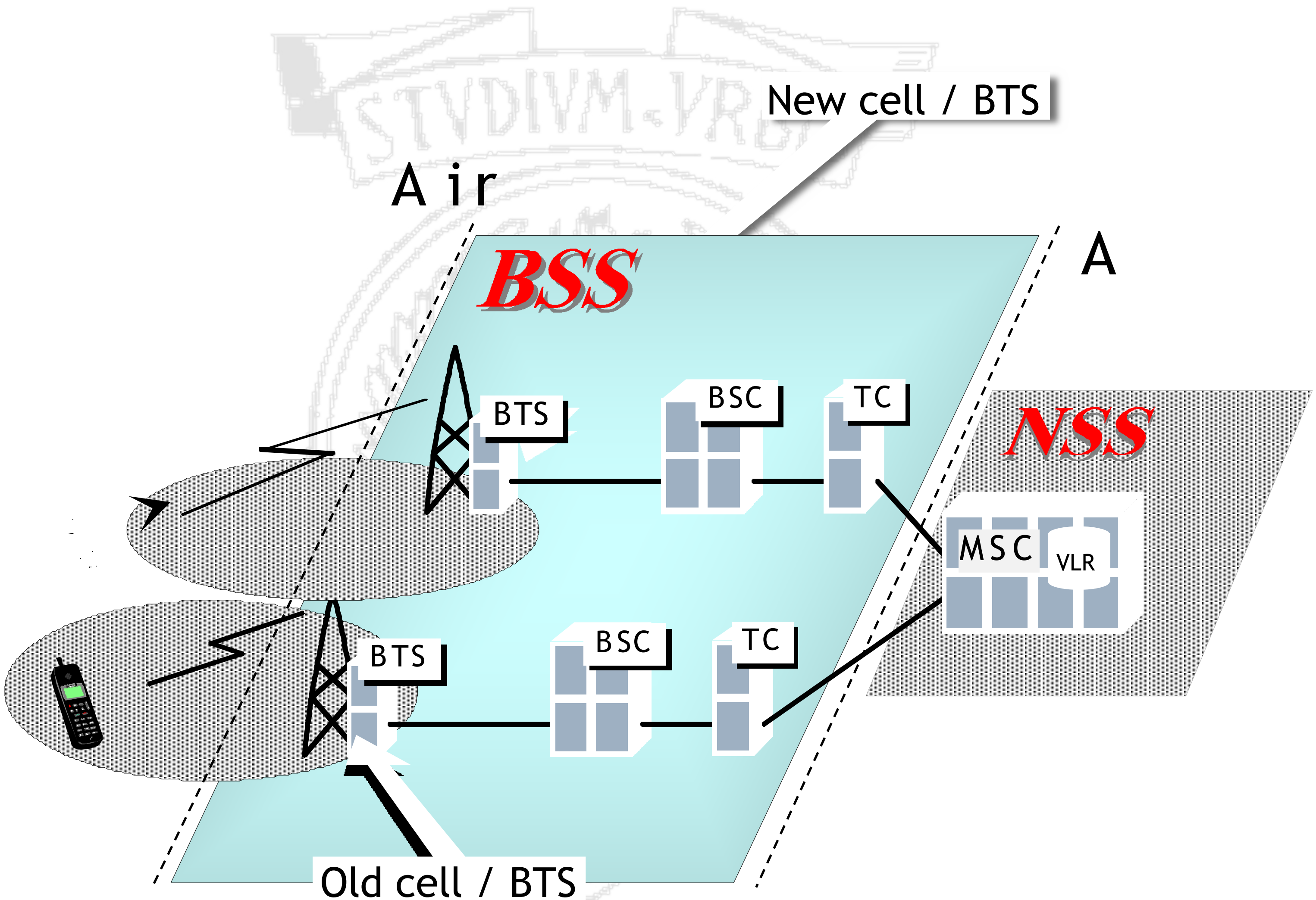
The handover procedure is fully controlled by the BSC

- The BSC identify the best BTS and the best TCH for the MS, based on MS and BTS measurements
- The BSC connects to the new BTS and requires the allocation of a new TCH
- The BSC signals to the MS (using the logical channel FACCH) to use the new TCH. The old radio carrier is released.
- The MS starts sending traffic on the new TCH
- The old connection is released

Additional Steps

- After the handover the MS must acquire information about the new adjacent cells. It uses the Slow Associated Control CHannel (SACCH)
- If the LA is changed by the handover, a Location Procedure must be triggered by the MS

Inter Cell – Inter BSC Handover

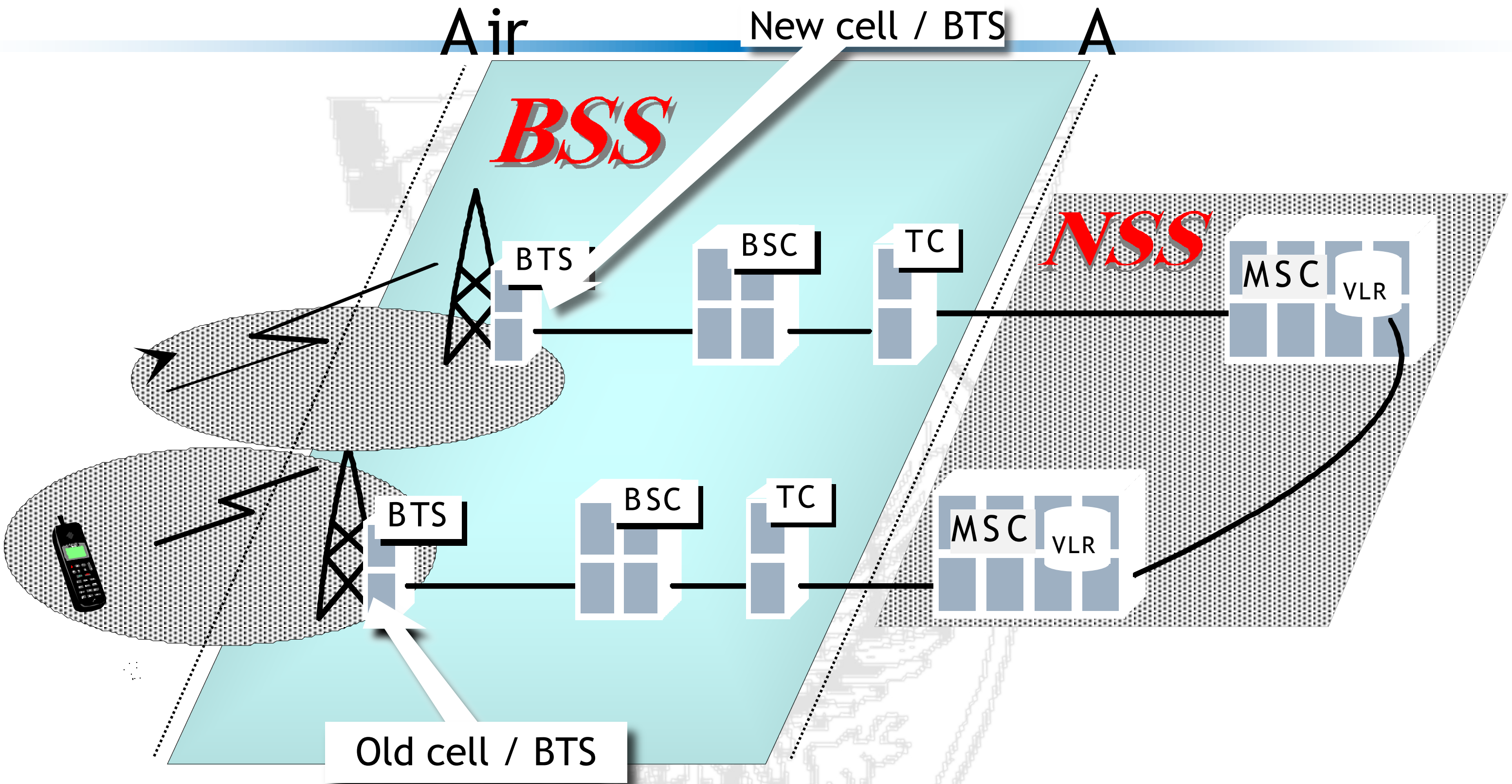


Inter Cell-
Inter BSC Handover

The handover procedure is initiated by the BSC

- The BSC identifies the best BTS and the best TCH for the MS
- The current BSC sends a message to the MSC/VLR, as the new BTS is controlled by another BSC
- The MSC creates a connection to the new BSC
- The new BSC reserves a radio channel for the MS. The old carrier is released
- The new BSC sends a command to the MS, which should now use the new radio channel (TCH)
- The MS starts sending traffic on the new channel. The connection is routed by the MSC towards the new BSC
- The old connection is released
- If before if the LAI change a location update must be performed

Inter MSC Handover



Handover is more complex because different MSC/VLR are involvede

- The call is routed by the initial MSC to the final MSC

Inter MSC Handover (1)

The handover procedure is initiated by the BSC

- The current BSC decides an handover towards a BTS controlled by another MSC/VLR
- The current BSC sends an handover command to the initial MSC/VLR
- The initial MSC/VLR sends a request to the final MSC/VLR
- The final MSC/VLR allocates an HandOver Number (HON), which is transmitted to the initial MSC/VLR

Inter MSC Handover (2)

- The destination MSC/VLR starts a connection to the new BSC
- A traffic channel is reserved to the MS by the new BSC
- The initial MSC/VLR sends an handover command to the MS by using the FACCH channel of the old BSC and BTS
- The MS switches to the new channel and starts sending traffic over the new TCH
- The old connection is released

HandOver Number

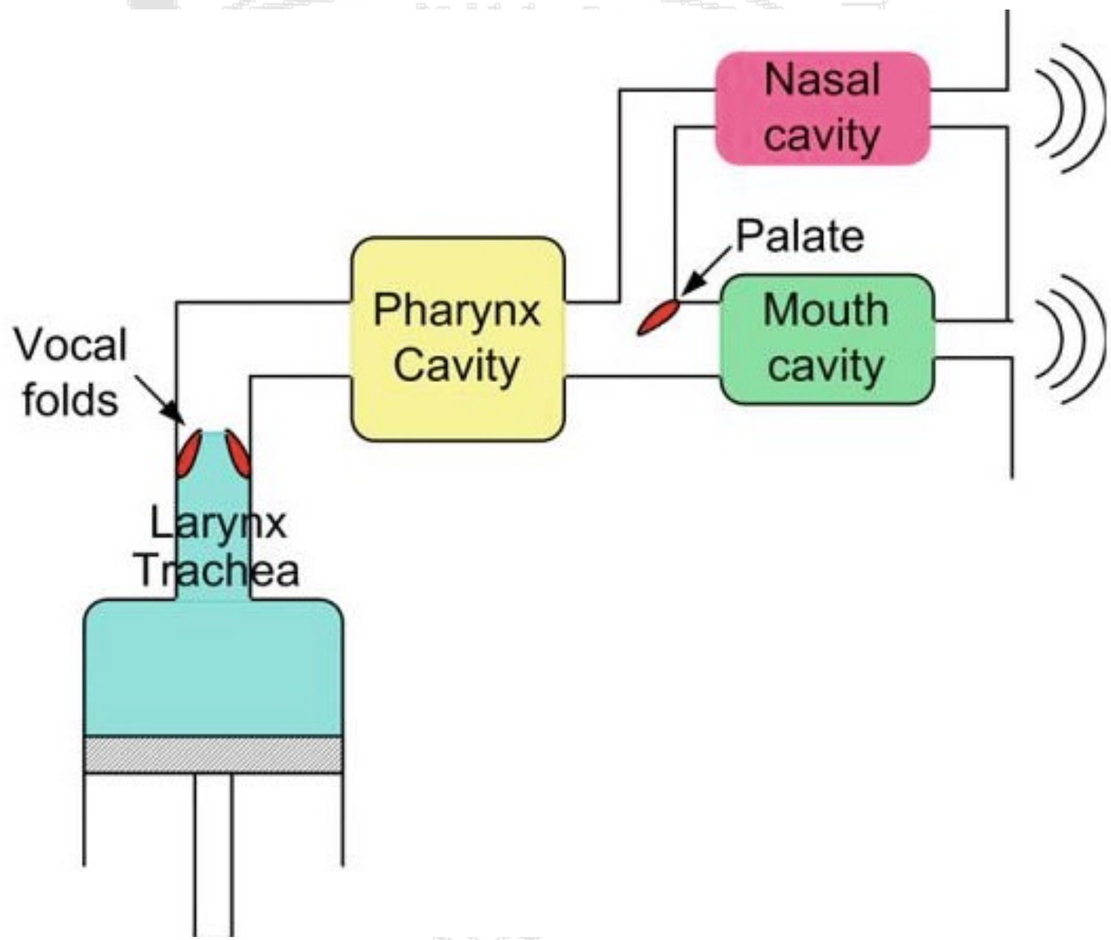
- Same format as MSRN and MSISDN
- HON = CC + NDC + SN
  - CC = Country Code
  - NDC = National Destination Code
  - SN = Subscriber Number
- SN points to a database
  - in case of MSISDN located in the HLR
  - in case of HON and MSRN located in VLR
- HON contains enough information to allow the GMSC to route the call towards the destination MSC
- In this case other network elements (HLR) must receive information on the change of MSC/VLR

MS is switched off

- When a MS is switched off, it sends to the network a ***IMSI detached*** message
- The MSC/VLR flags the user as detached
- Paging is no longer performed until the MS is switched on again

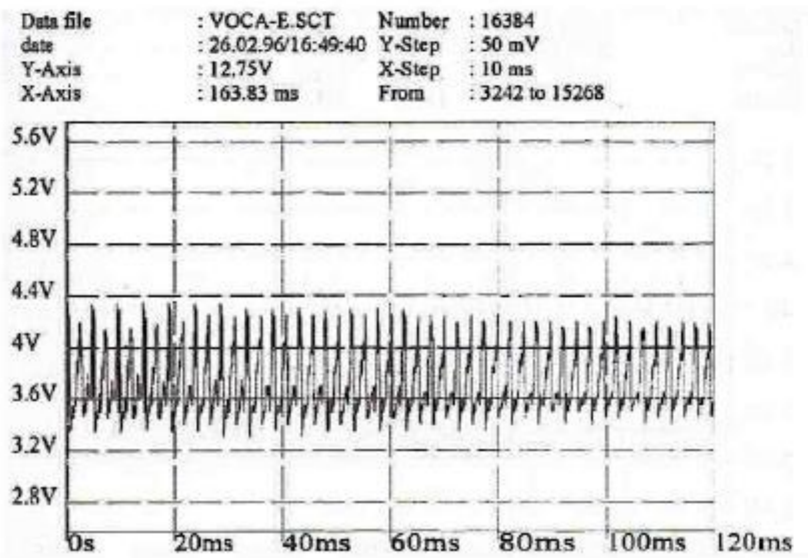
Voice Coding

Speech signals

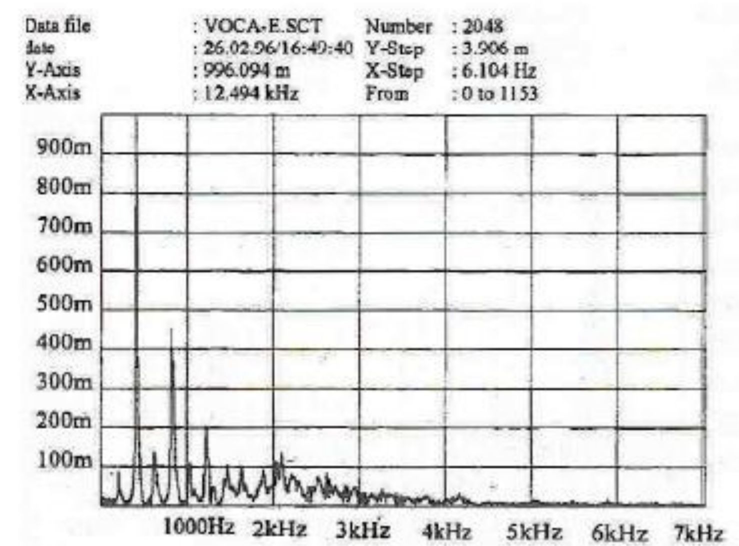


Voice coding: voiced sound

Time- frequency features, vowel «e»



Signal in time



Signal spectrum (frequency components)

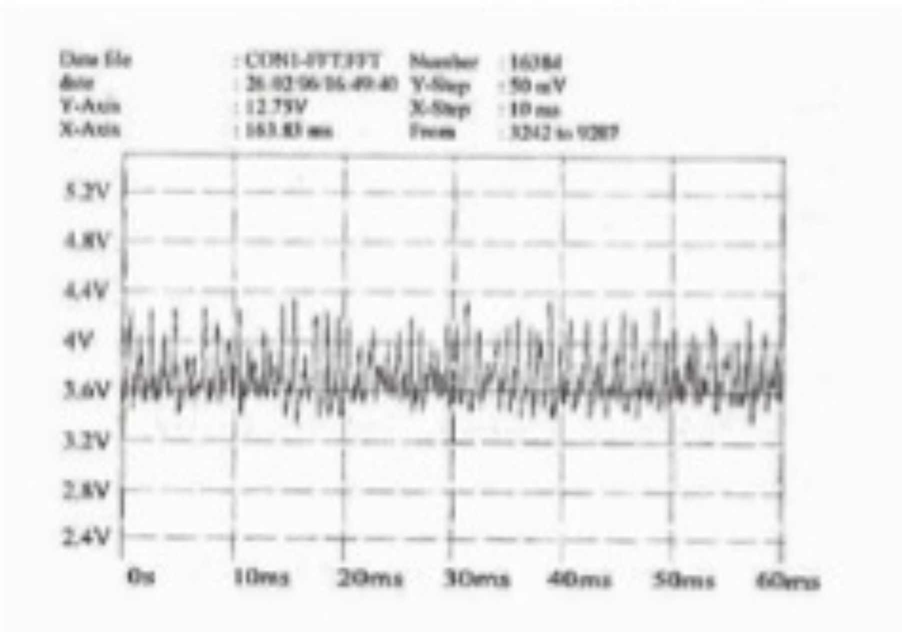
- Sounds produced by the vibrations of the vocal folds
- Features:

1)Periodic (pitch period); 2) High amplitude; 3) Slow variation of the signal

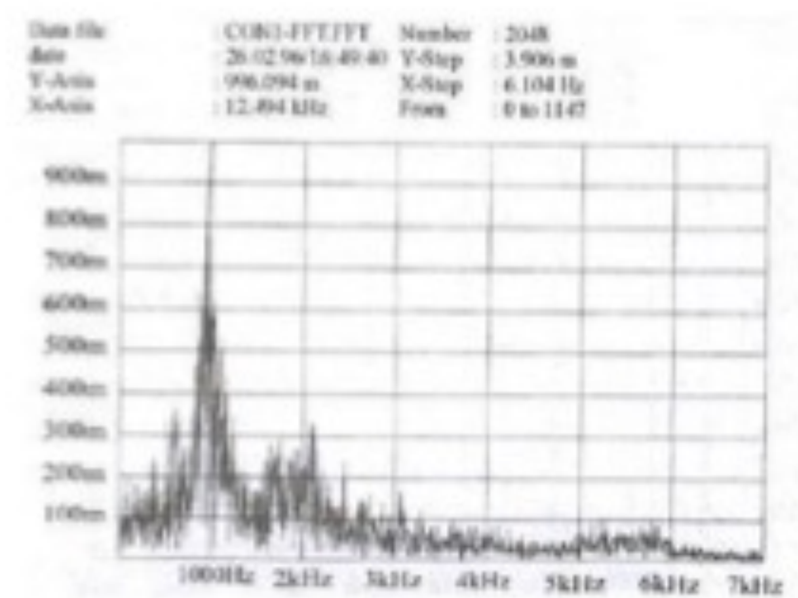
4) low number of frequencies around which the energy is concentrated (formant frequencies) 5) formant frequencies are low frequencies

Voice coding: voiceless sound

Time- frequency features, consonant f



Signal in time

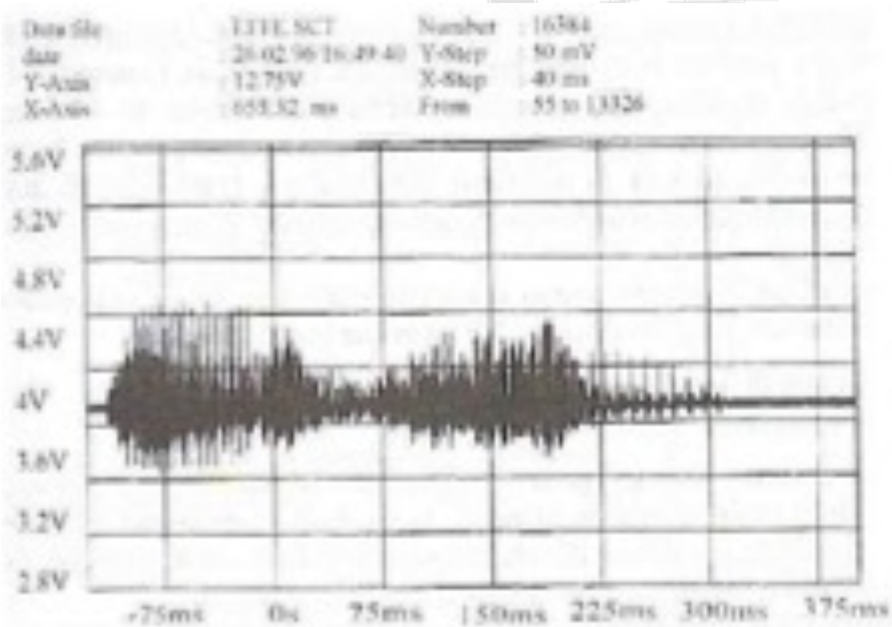


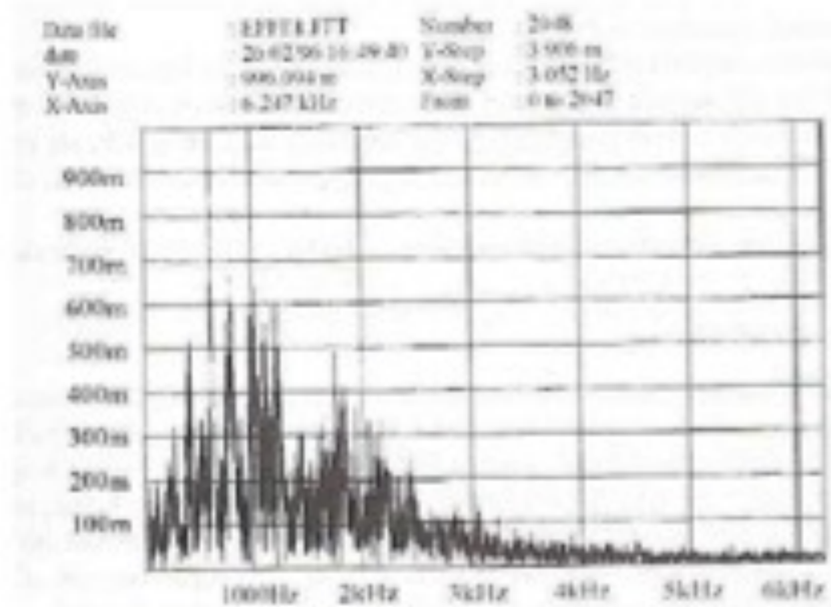
Signal spectrum (frequency components)

- Features: 1)Randomic pattern ; 2) Lower amplitude; 3) Energy concentrated also at higher frequencies.

Voice coding

One word: effe





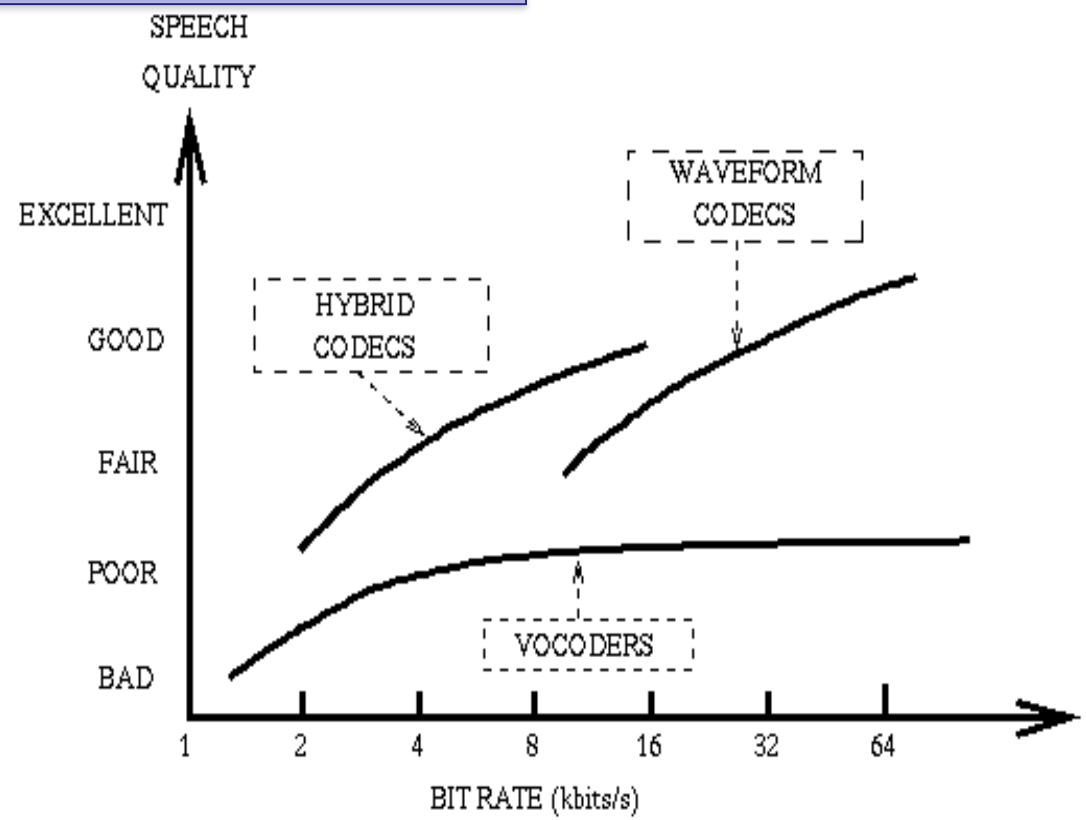
- Vowels and Consonants have different amplitudes.
- The most significant frequency components are located between 300Hz and 3400Hz, with (small) spectral components till 7KHz

Voice coding

Speech signal is translated into a sequence of bits

Mean Opinion Score

- Waveform codecs
- Source codecs (vocoders)
- Hybrid codecs



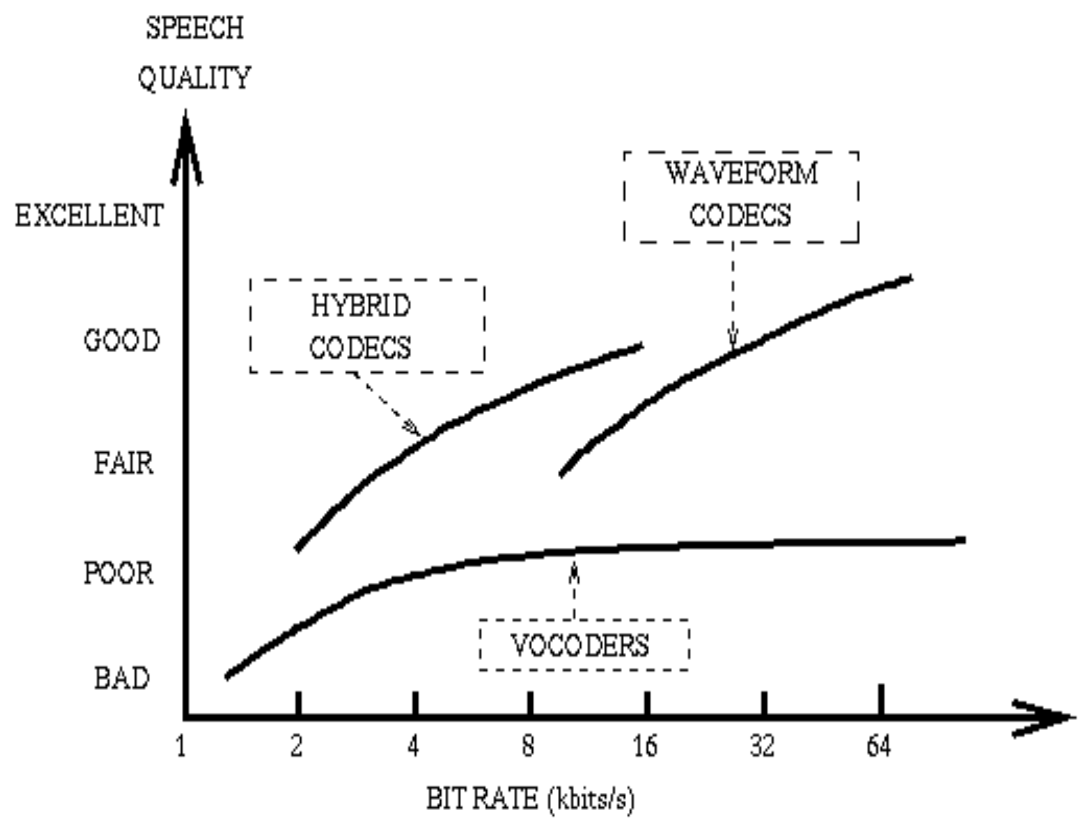
Voice coding

Speech signal is translated into a sequence of bits

Digitization of an analog signal ←

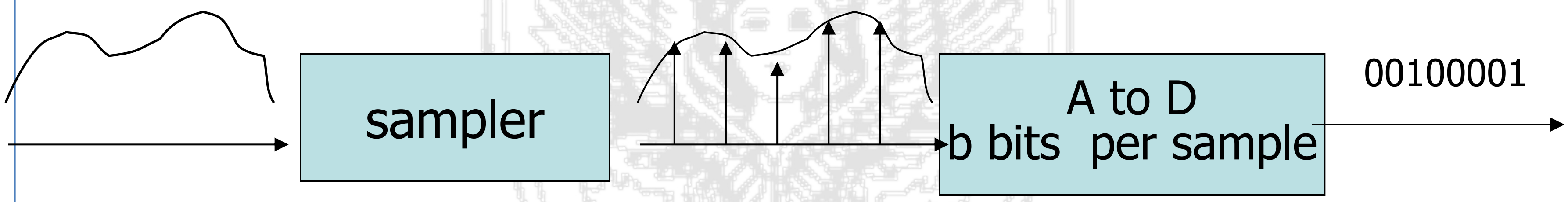
They make an explicit description of the waveform input (es. PCM)

- Waveform codecs
- Source codecs (vocoders)
- Hybrid codecs



Waveform codecs

- no a priori knowledge of how the signal was generated
- Information needed
  - Signal bandwidth (speech signal < 4 KHz)
  - maximum tolerable quantization noise



high quality, low complexity, low delay (1 sample),

robustness to errors and background noise

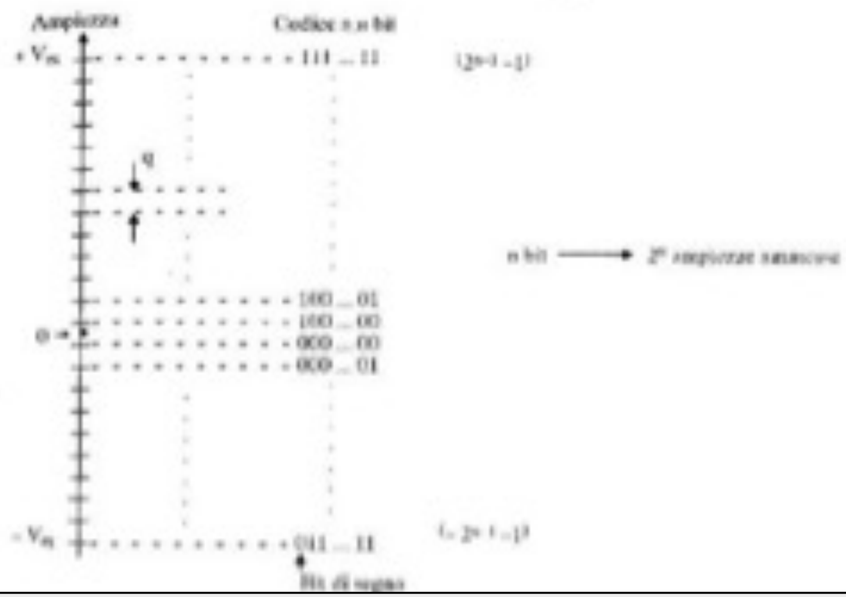
Waveform coding: Pulse Code Modulation (PCM)

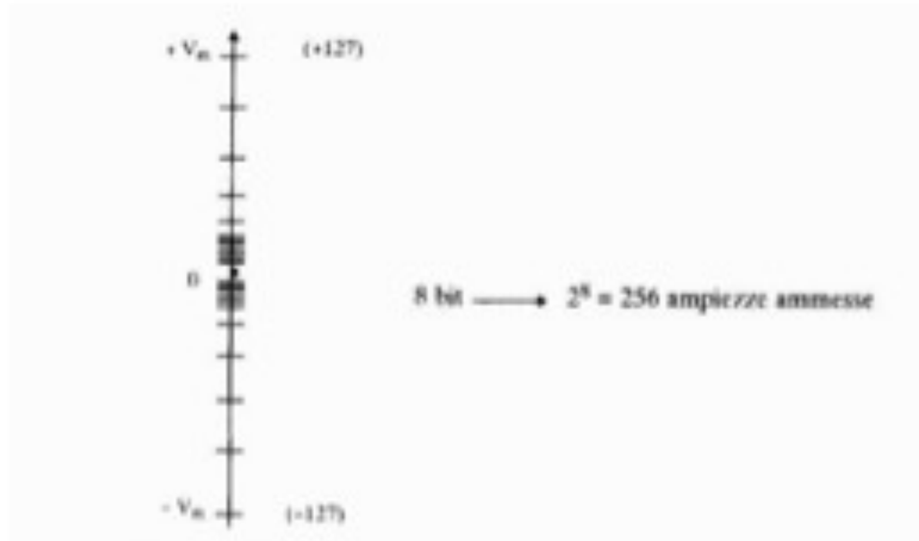
- standardized by ITU in 1960: G.711
- We assume B = 4 kHz, and the sampling frequency Bc = 8 kHz, 8 bit / sample, 64 kb / s
- Two different quantization rules (logarithmic)
  - for America (m-law) and for Europe (A-law)
  - standard conversion rules

Quantization

Uniform quantization

Non - Uniform quantization



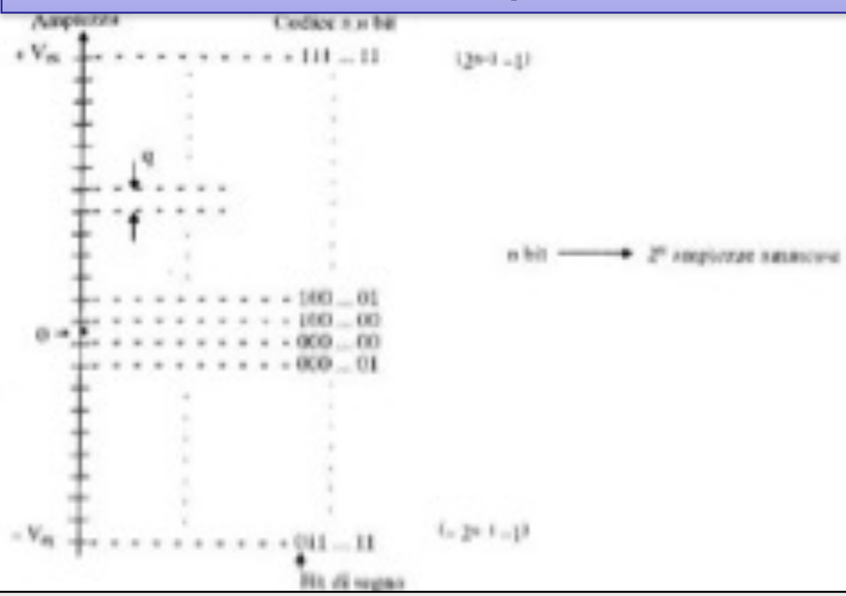


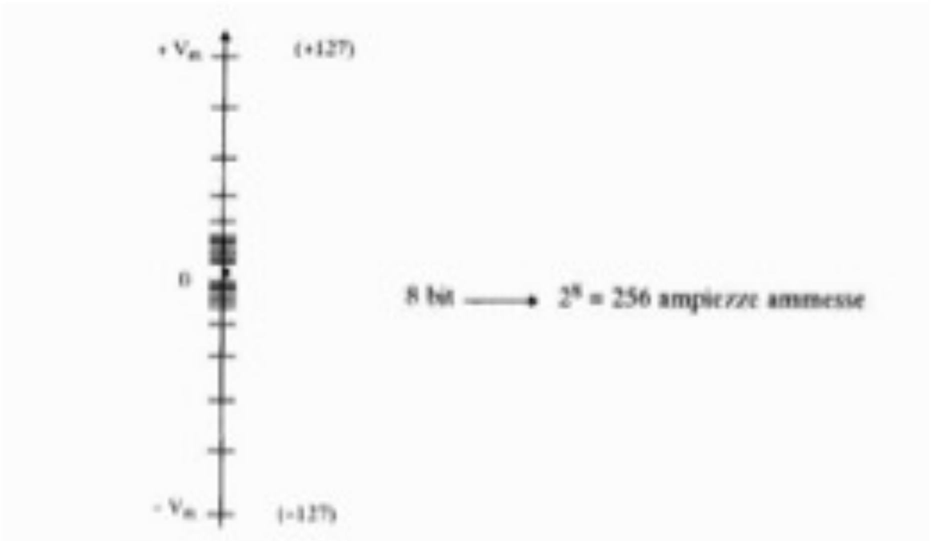
- Quantization error is fixed (<q/2, where q is the quantization step)
- 12 bit per sample are needed to achieve a quantization error low enough also for small values

Quantization

The axis of the amplitudes is divided into equal intervals

Non - Uniform quantization





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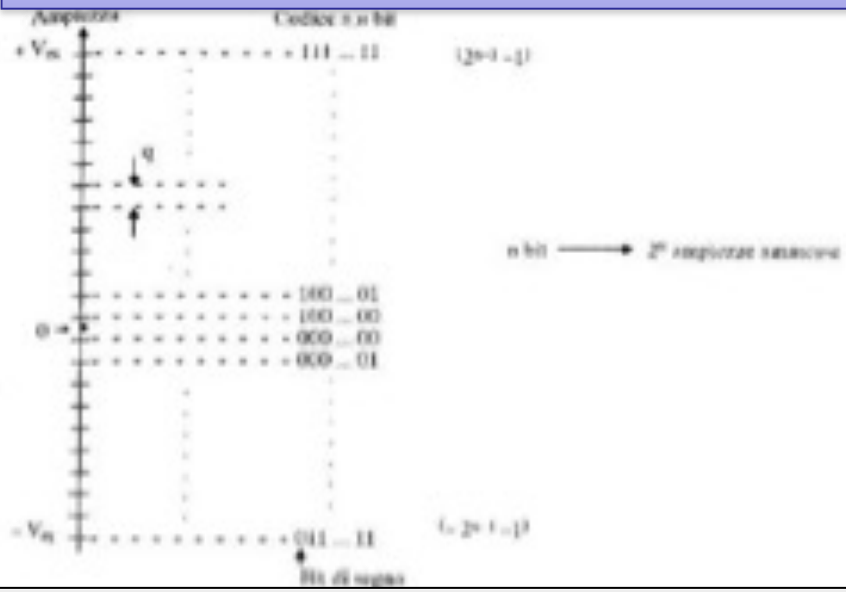
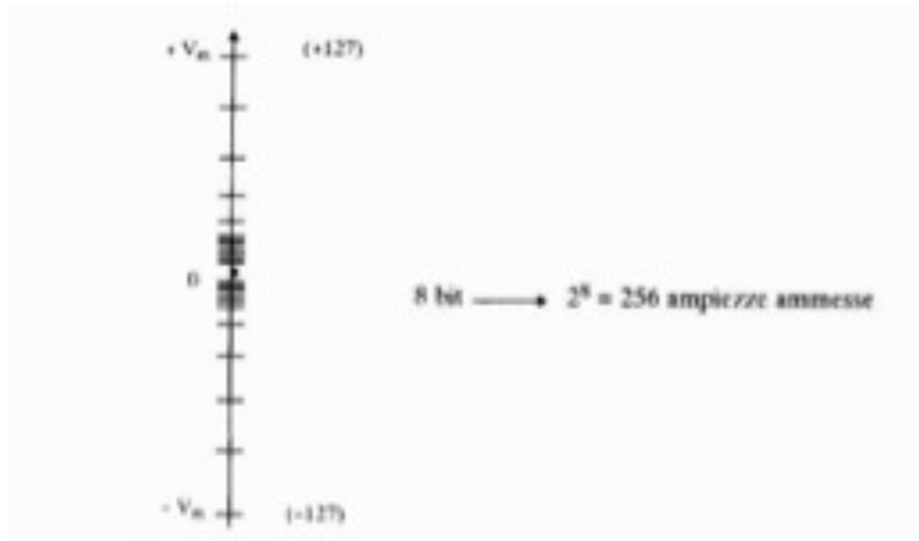
- Many relatively small values
- Higher quantization errors can be tolerated in case of high values
- 8 bits per sample results in excellent perceived quality

Quantization

The axis of the amplitudes is divided into equal intervals

Non - Uniform quantization





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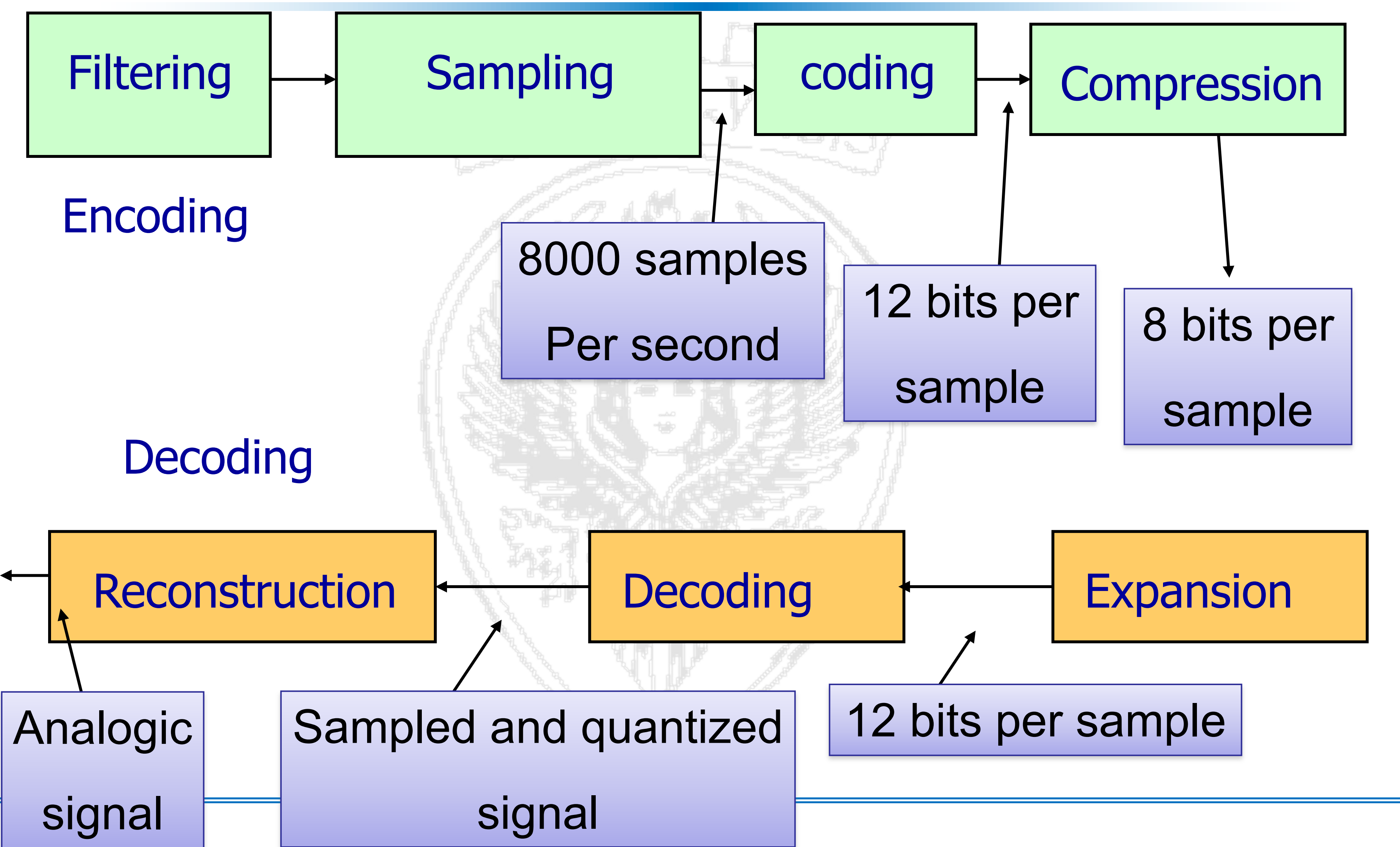
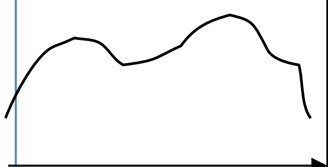
compression

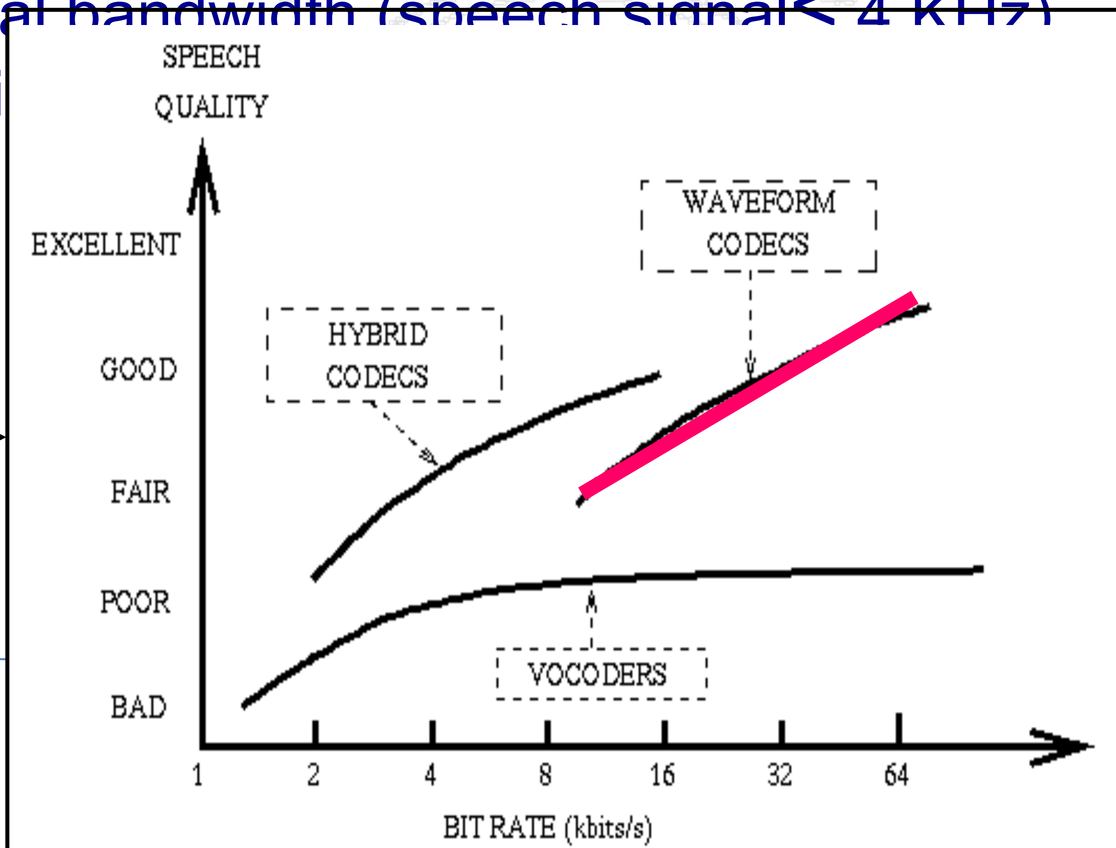
Fasi della codifica/decodifica



Waveform codecs

- no a priori knowledge of how the signal was generated
- Information needed
  - Signal bandwidth (speech signal< 4 KHz)
  - maxi





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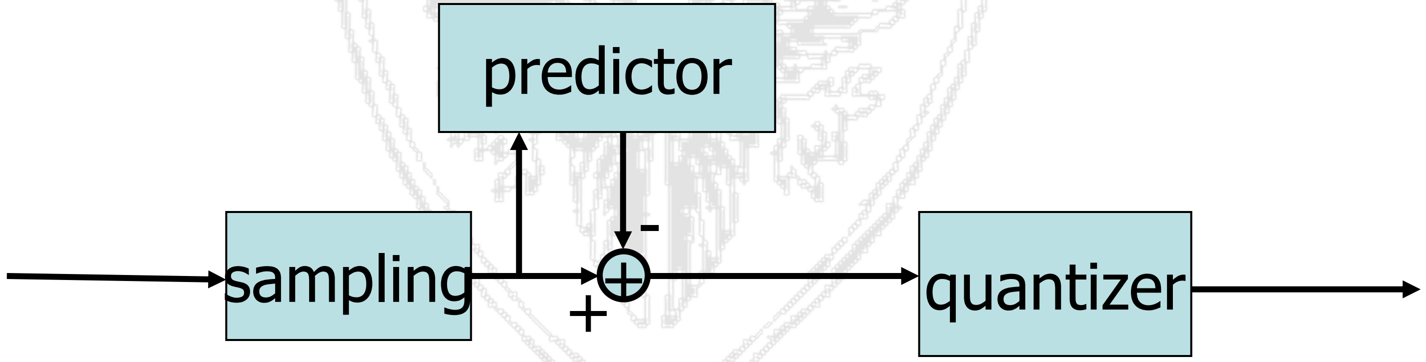
Differential PCM (DPCM)

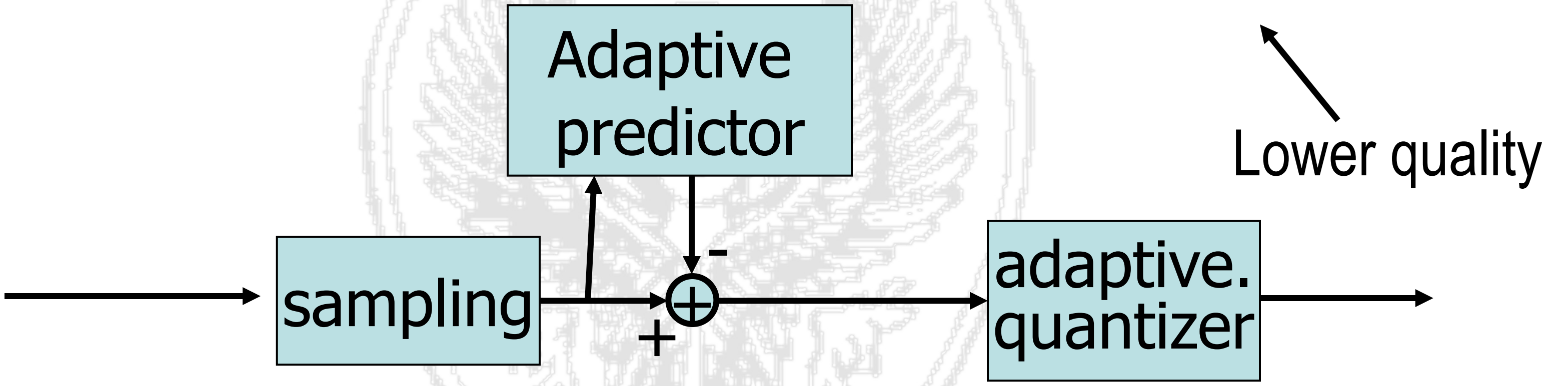
- the subsequent voice samples are correlated
- You can use prediction methods for evaluating the next sample known previous
- transmitting only the difference between the predicted value and the actual value
- because of the correlation the variance of the difference is smaller and it is possible to encode it with a smaller number of bits



Adaptive DPCM (ADPCM)

Cordless
DECT

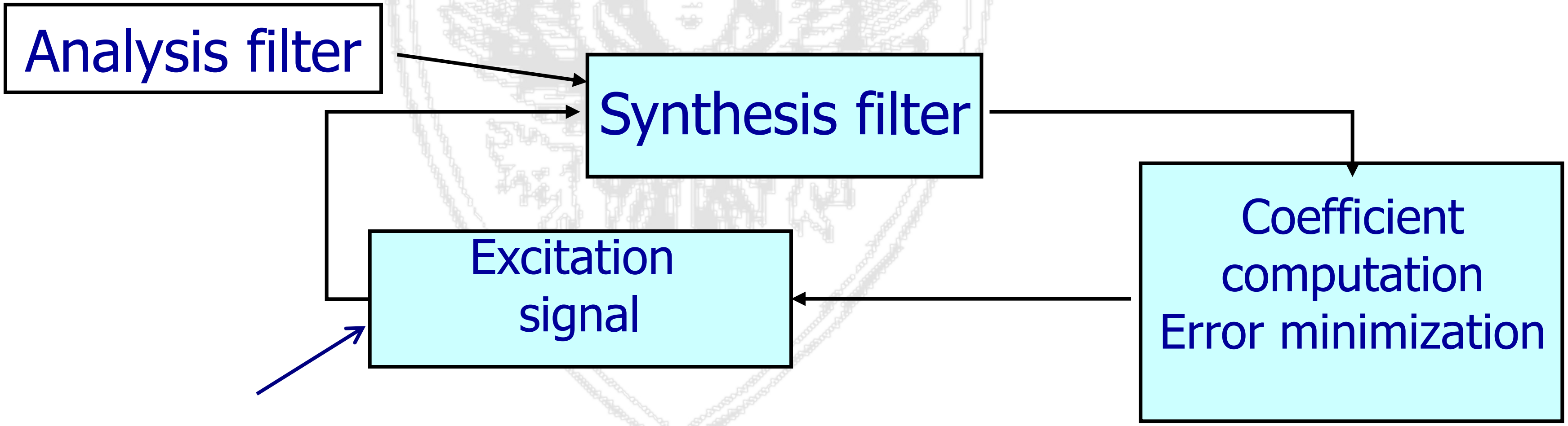
- performance improves if the predictor and quantizer are adaptive
- standardized in 1980 by ITU ADPCM 32 kbit / s G.721
- subsequently ADPCM at 40, 32, 24, 16-kbit / s G.726 and G.727



Benefits: Reducing the rate of emission while achieving equal quality (from 64 Kbps to 32 Kbps) 2) enable higher quality given a fixed data rate available per voice channel)

Source codecs (vocoders)

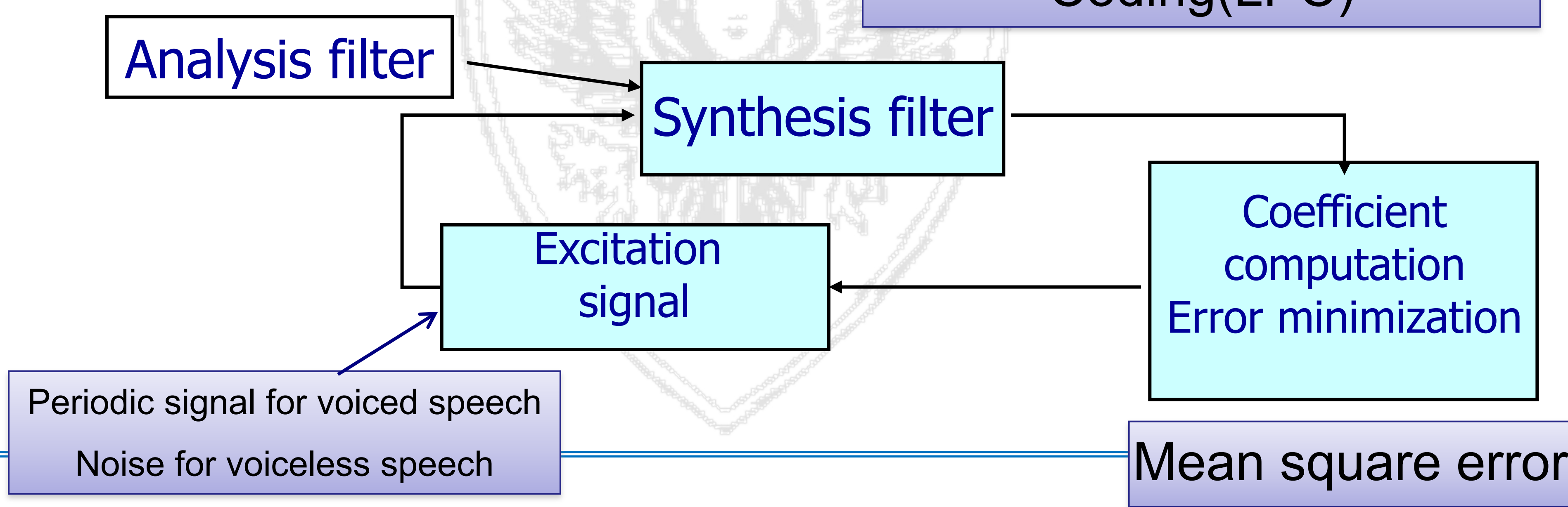
- They are based on models for the generation of the human voice
- models allow us to "remove redundancy" from the vocal segments to obtain an information base sufficient to reproduce the original voice signal (Idea: If we know the structure of the signal little information features will be enough to rebuild it)
- high complexity
- delays on average higher
- sensitive to errors, noise and non-human sounds



Source codecs (vocoders)

- Instead of trying to encode the waveform itself, vocoding techniques try to determine parameters about how the speech signal was created and use these parameters to encode the signal
- To reconstruct the signal, these parameters are fed into a model of the vocal system which outputs a speech signal.
- Filter: Transfer function $H(z)=A(z)/B(z)$

Example: Linear Predictive Coding(LPC)



Linear Vocoder (LPC)

- They encode the parameters of the synthesis filter and the excitation sequence
- in decoding a synthesizer uses the received parameters to reproduce the signal
- **The voice sample is approximated by a linear combination of a number of past samples**

- high delays: segmentation, analysis, synthesis
- quality intelligible but not natural (limits in the model + problems with background noise)
- low bit rate: <2.4 kbit / s

Main voice coding

Codebook Excited Linear prediction

LTP= Long term prediction

Coding	Year	Bit rate (kbit/s)	Frame size (ms)	Look ahead (ms)
G.711 PCM	1972	64	0.125	0
G.726 ADPCM	1990	32	1	0
G.722 Subband ADPCM	1988	48-64	0.125	1.5
G.728 LD-CELP	1992-94	16	0.625	0
G.729 CS-ACELP	1995	8	10	5
G.723.1 MP-MLQ	1995	6.3	30	7.5
G.723.1 ACELP	1996	5.3	30	5
RPE-LTP (GSM)	1987	13	20	0

Hybrid

Regular pulse excitation-residual signal is undersampled

The sequence of departure from which the decoder must start to rebuild the speech signal is not a pseudorandom sequence but is representative of the "real signal"

Codebook Excited Linear Prediction

- Tries to overcome the synthetic sound of vocoders by allowing a wide variety of excitation signals, which are all captured in the CELP codebook.
- To determine which excitation signal to use, the coder performs an exhaustive search.
  - For each entry in the codebook, the resulting speech signal is synthesised and the entry which created the smallest error is chosen.
- The excitation signal is encoded by the index of the corresponding entry (Vector Quantisation)
- CELP techniques allow bit rates of even 4.8 kbps.

Main voice coding

Codebook Excited Linear prediction

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G.728 LD-CELP	1992-94	16	0.625	0
G.729 CS-ACELP	1995	8	10	5
G.723.1 MP-MLQ	1995	6.3	30	7.5
G.723.1 ACELP	1996	5.3	30	5
RPE-LTP (GSM)	1987	13	20	0

Hybrid

Regular pulse excitation-residual signal is undersampled

The sequence of departure from which the decoder must start to rebuild the speech signal is not a pseudorandom sequence but is representative of the "real signal"