Cellular systems & GSM

Wireless Systems, a.a. 2014/2015

Un. of Rome "La Sapienza"

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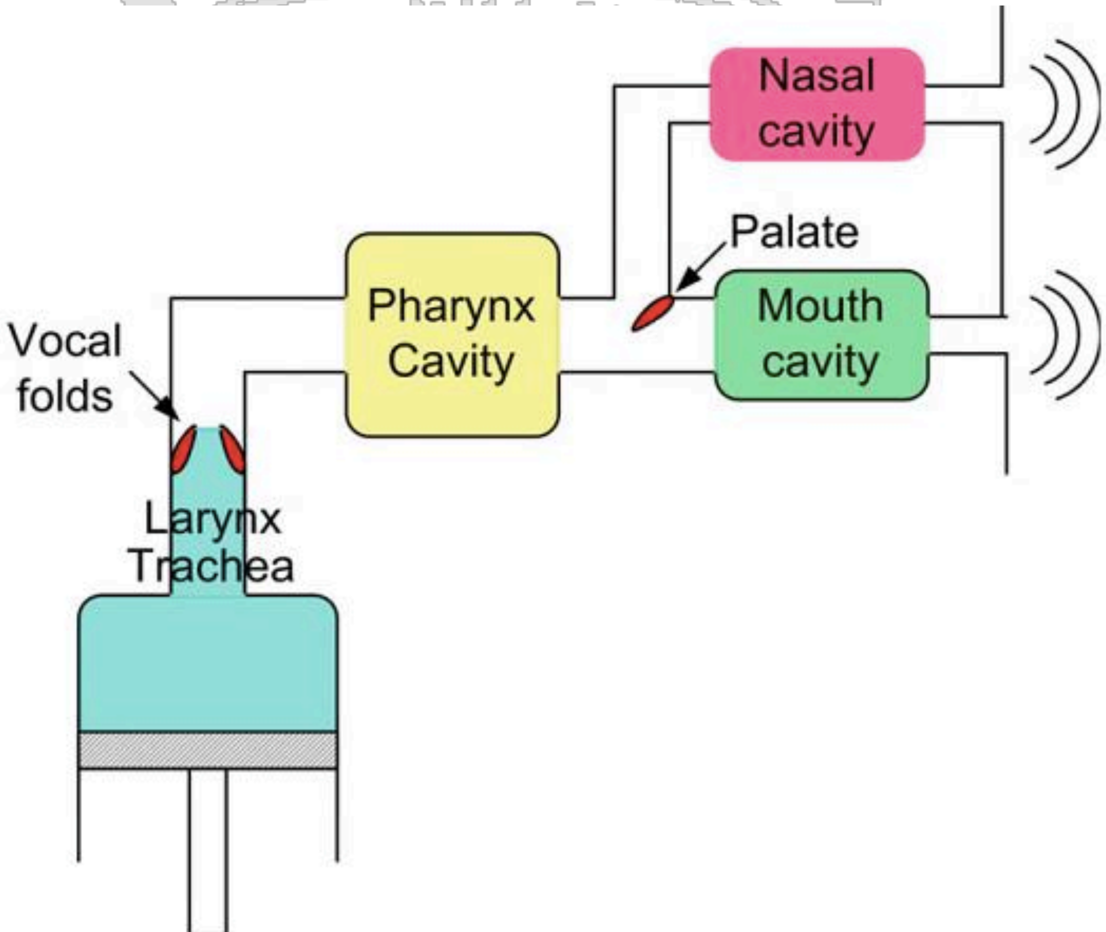
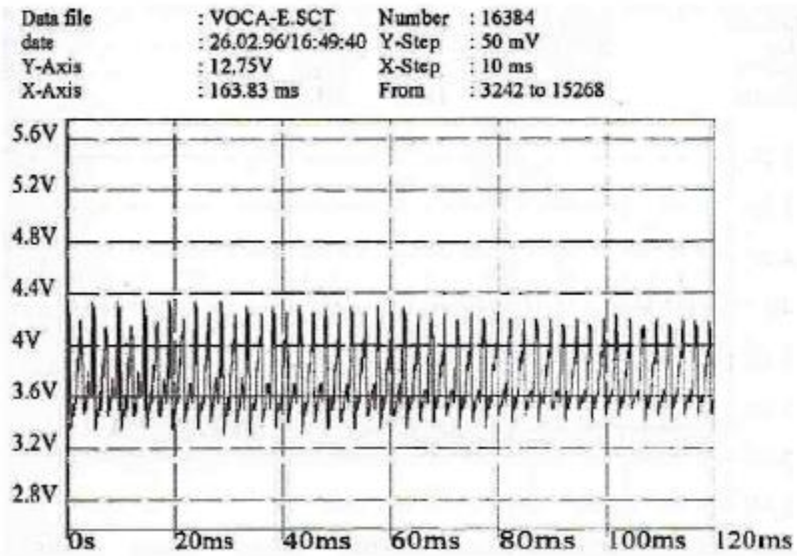
Voice Coding

Speech signals

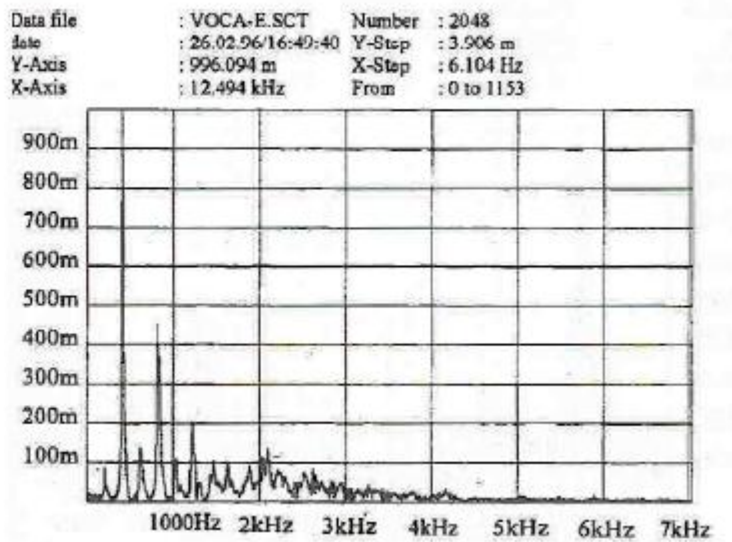


Voice coding: voiced sound

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

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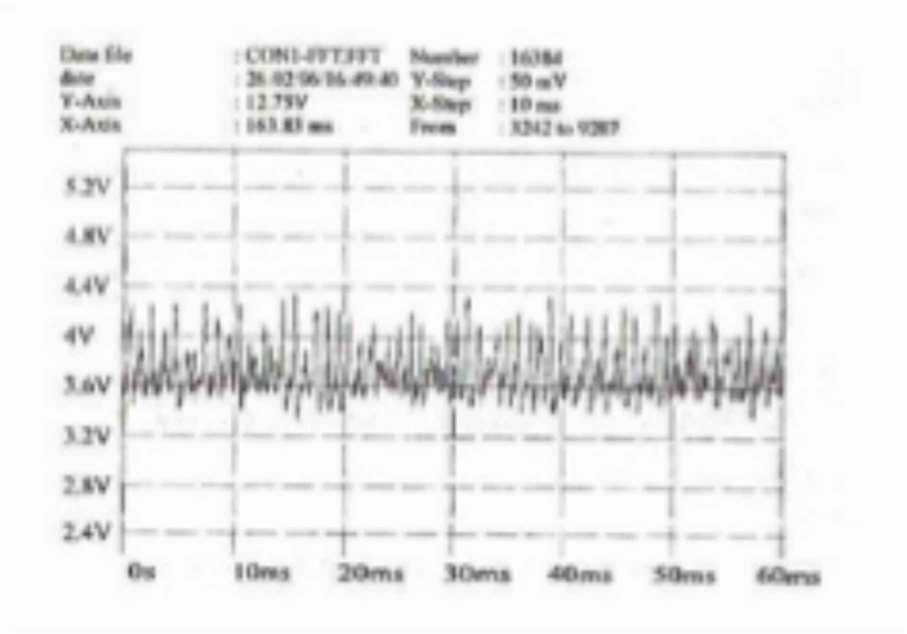
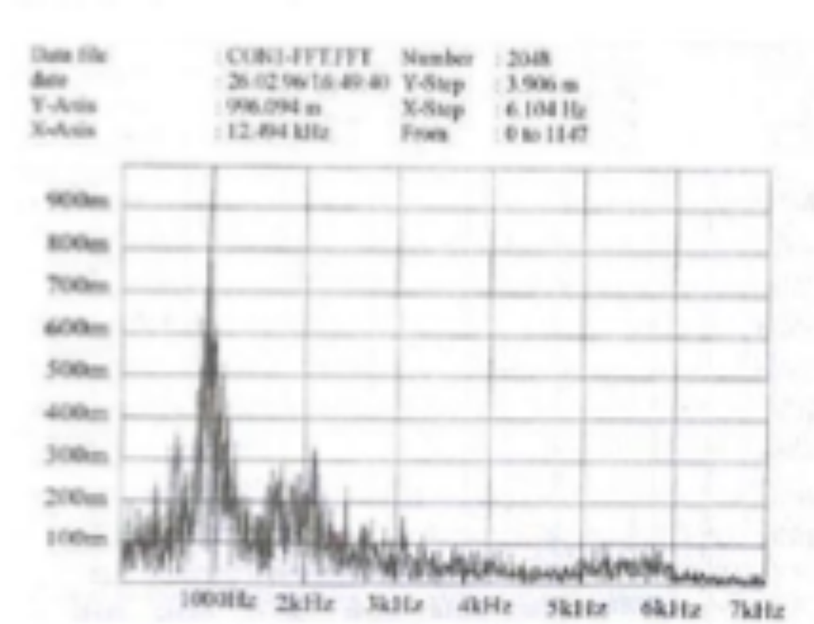
Voice coding: voiceless sound

Time- frequency features, consonant f



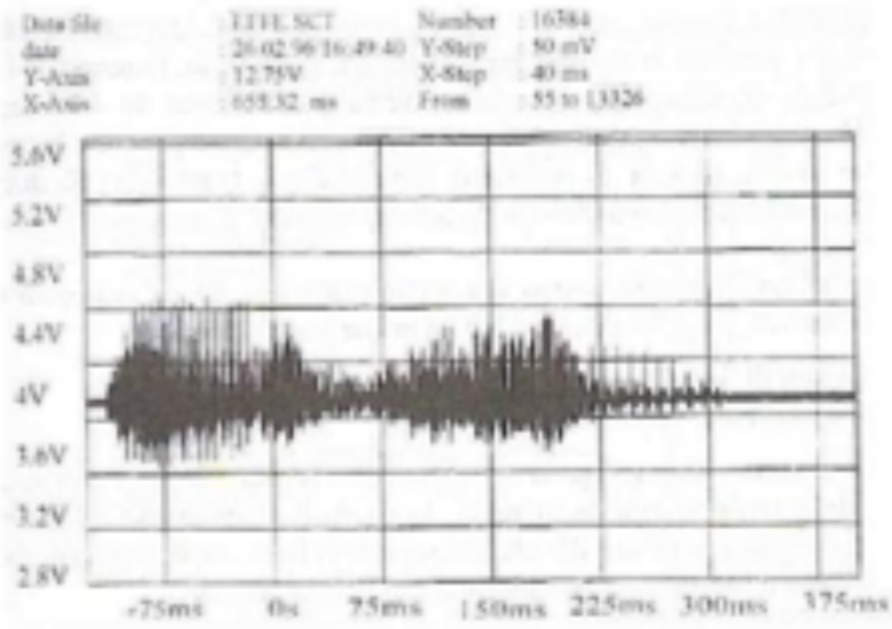
Signal in time

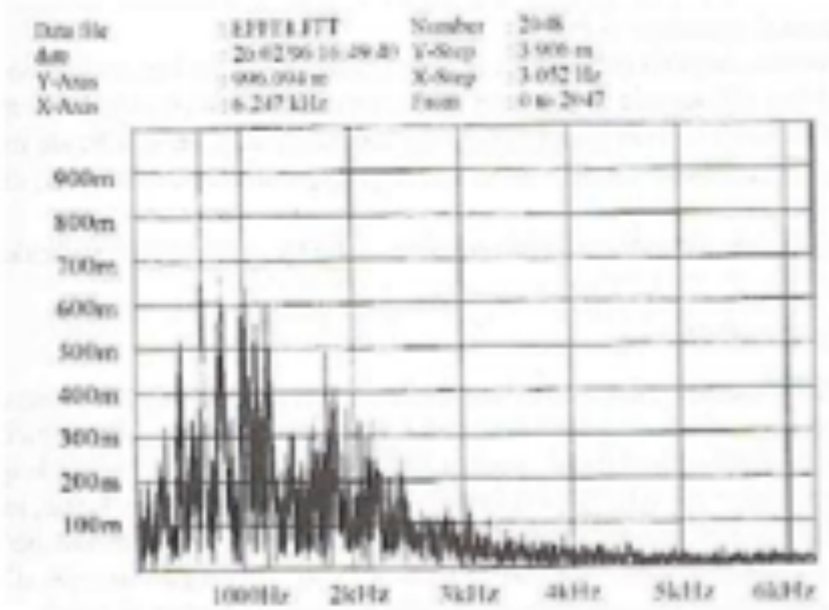


Signal spectrum (frequency components)

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

One word: effe



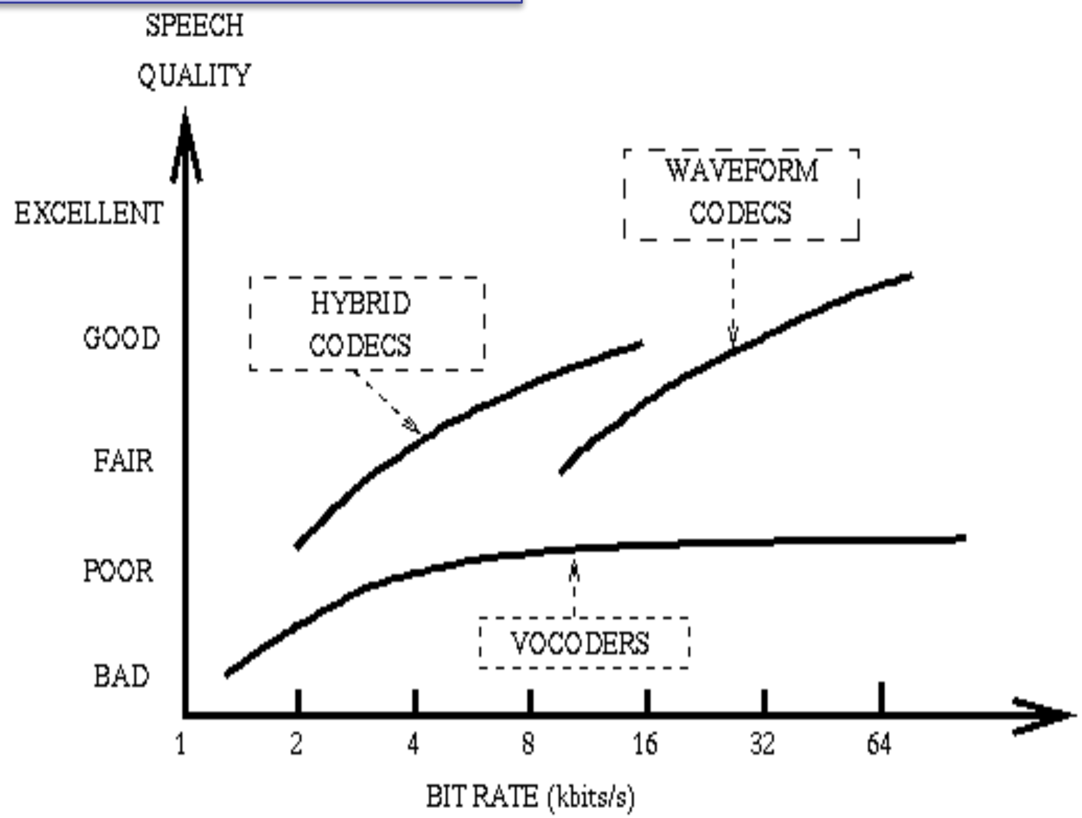


- Vowels and Consonants have different amplitudes.
- The most significant frequency components are located between 300Hz and 3400Hz, with (small) spectral components till 7KHz

Speech signal is translated into a sequence of bits

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

Mean Opinion Score

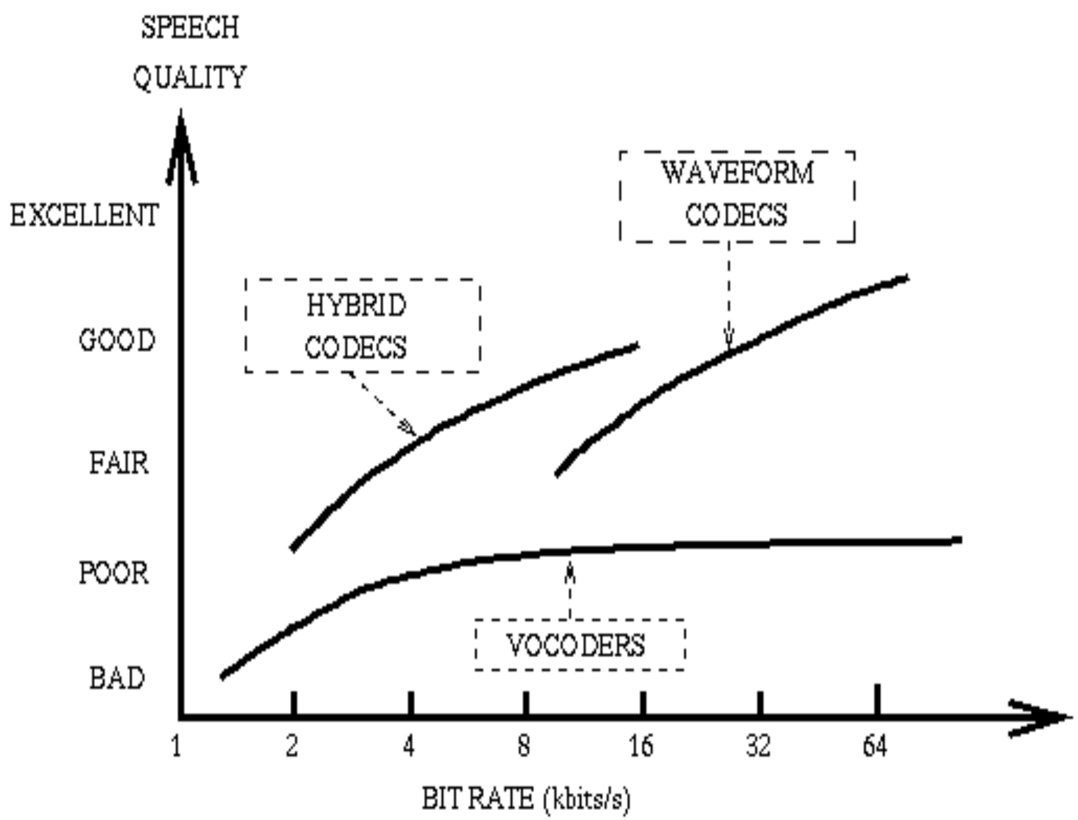


Speech signal is translated into a sequence of bits

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

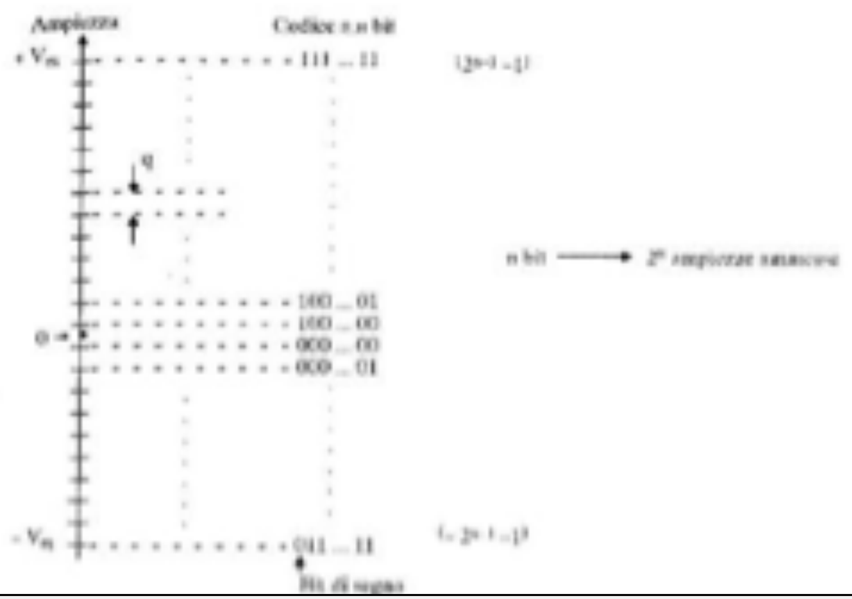
Waveform coding: Pulse Code Modulation (PCM)

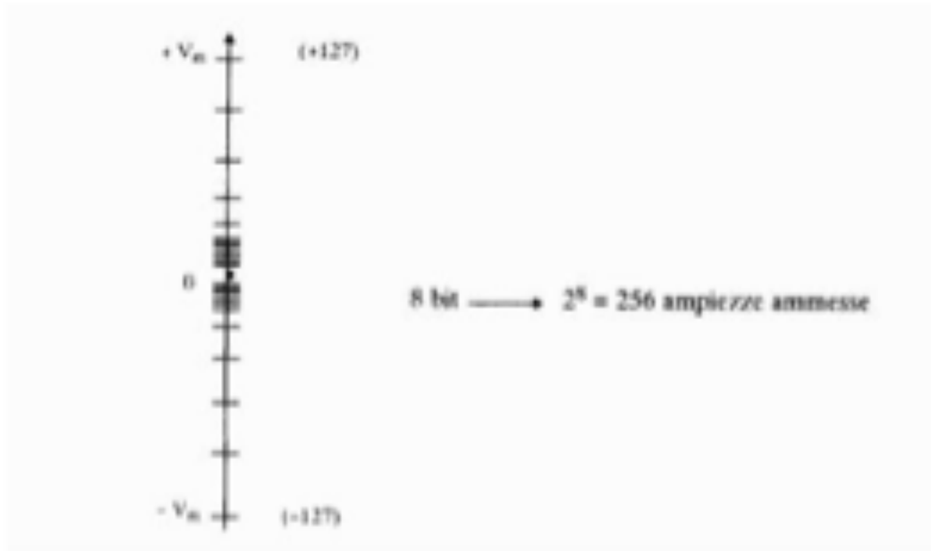
- standardized by ITU in 1960: G.711
- We assume B = 4 kHz, and the sampling frequency Bc = 8 kHz, 8 bit / sample, 64 kb / s
- Two different quantization rules (logarithmic)
  - for America (m-law) and for Europe (A-law)
  - standard conversion rules

Quantization

Uniform quantization

Non - Uniform quantization

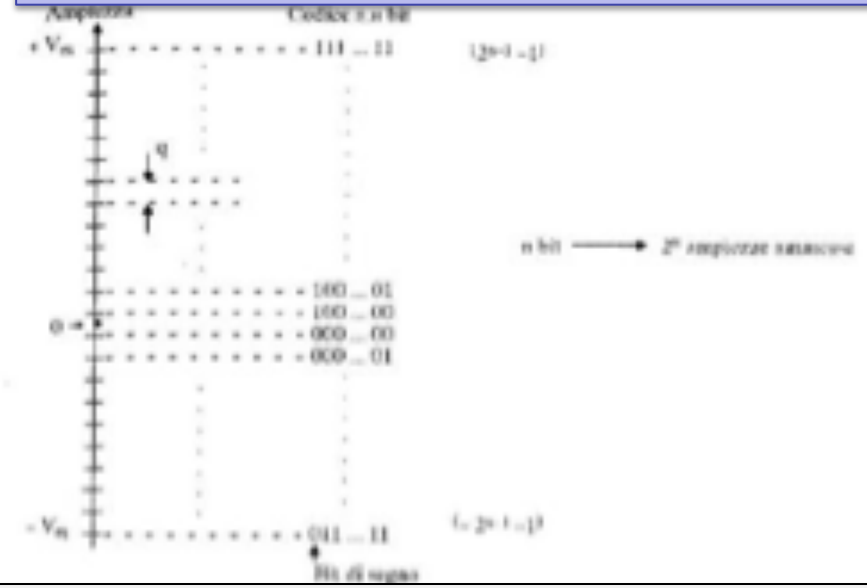




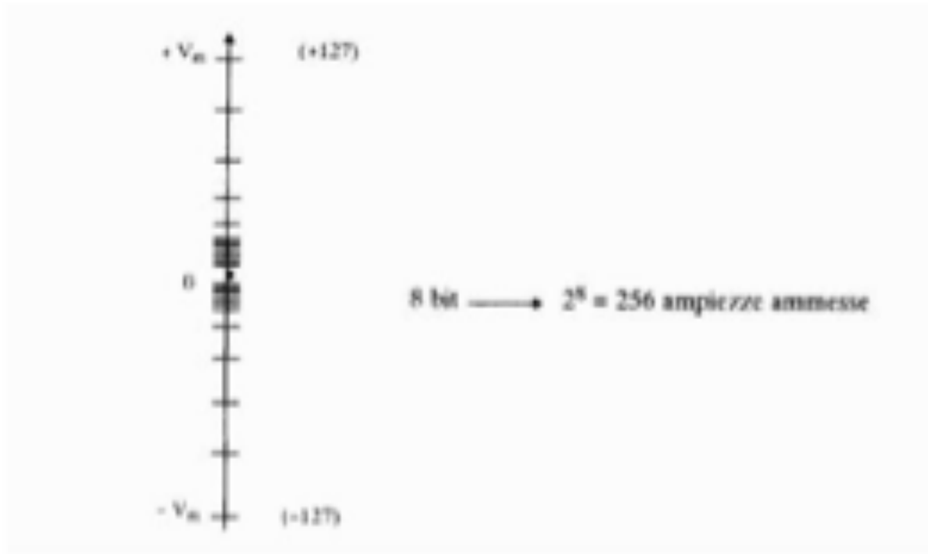
- Quantization error is fixed (<q/2, where q is the quantization step)
- 12 bit per sample are needed to achieve a quantization error low enough also for small values

Quantization

The axis of the amplitudes is divided into equal intervals



Non - Uniform quantization



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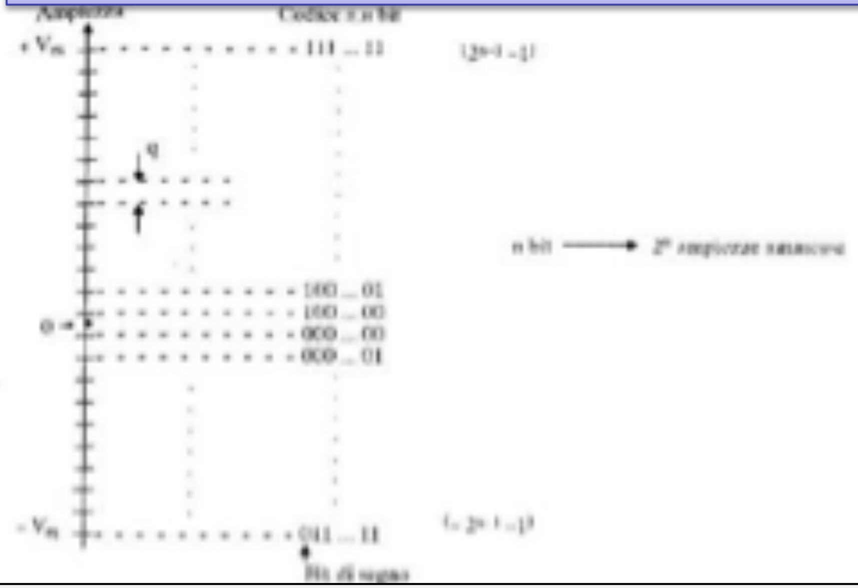
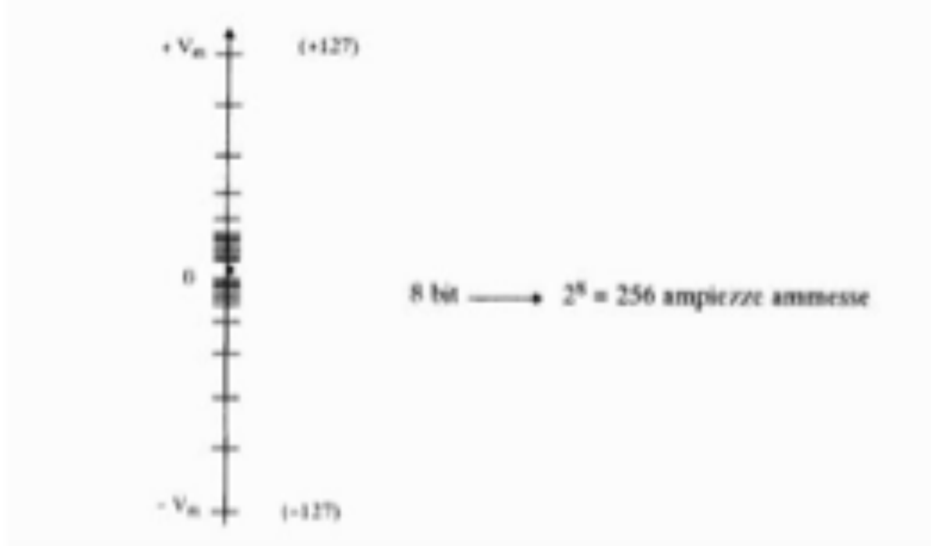
Non - Uniform quantization





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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

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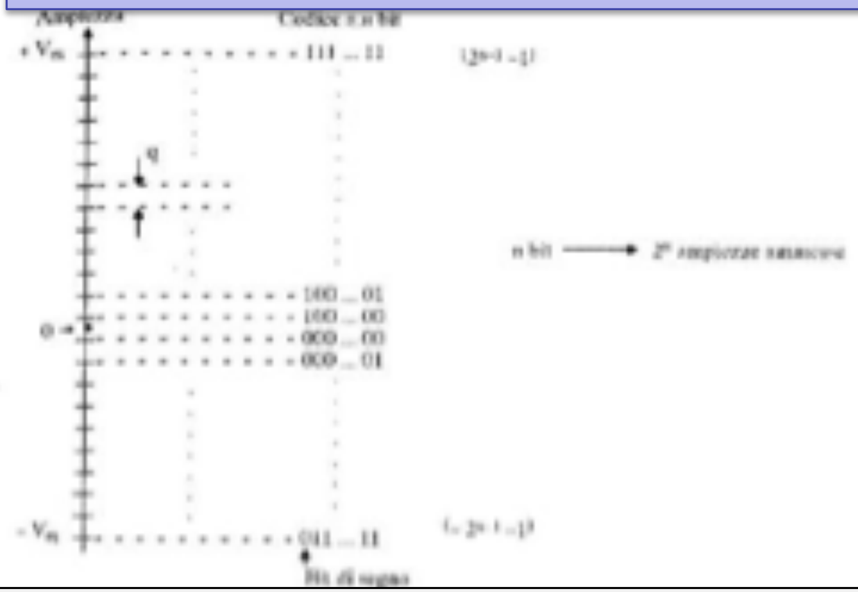
The axis of the amplitudes is divided into equal intervals



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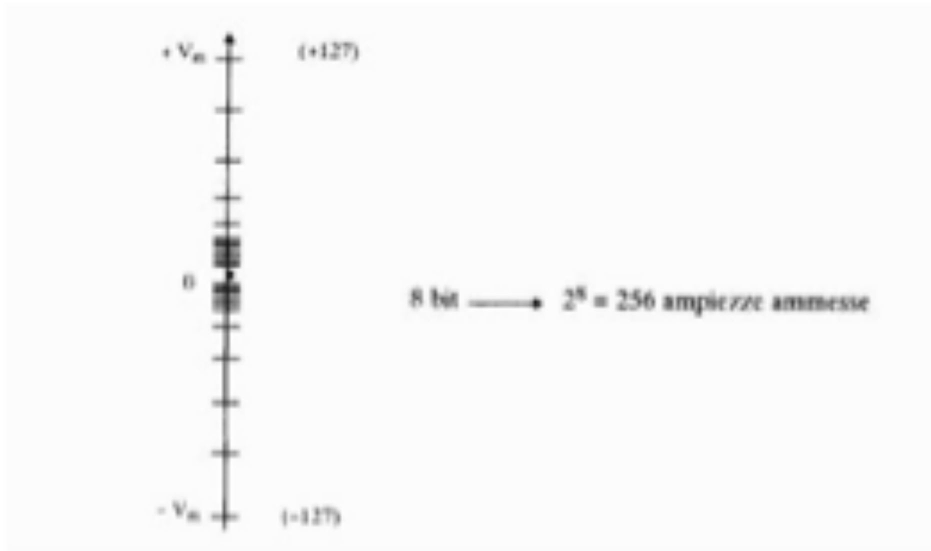

- Many relatively small values
- Higher quantization errors can be tolerated in case of high values
- 8 bits per sample results in excellent perceived quality ← compression

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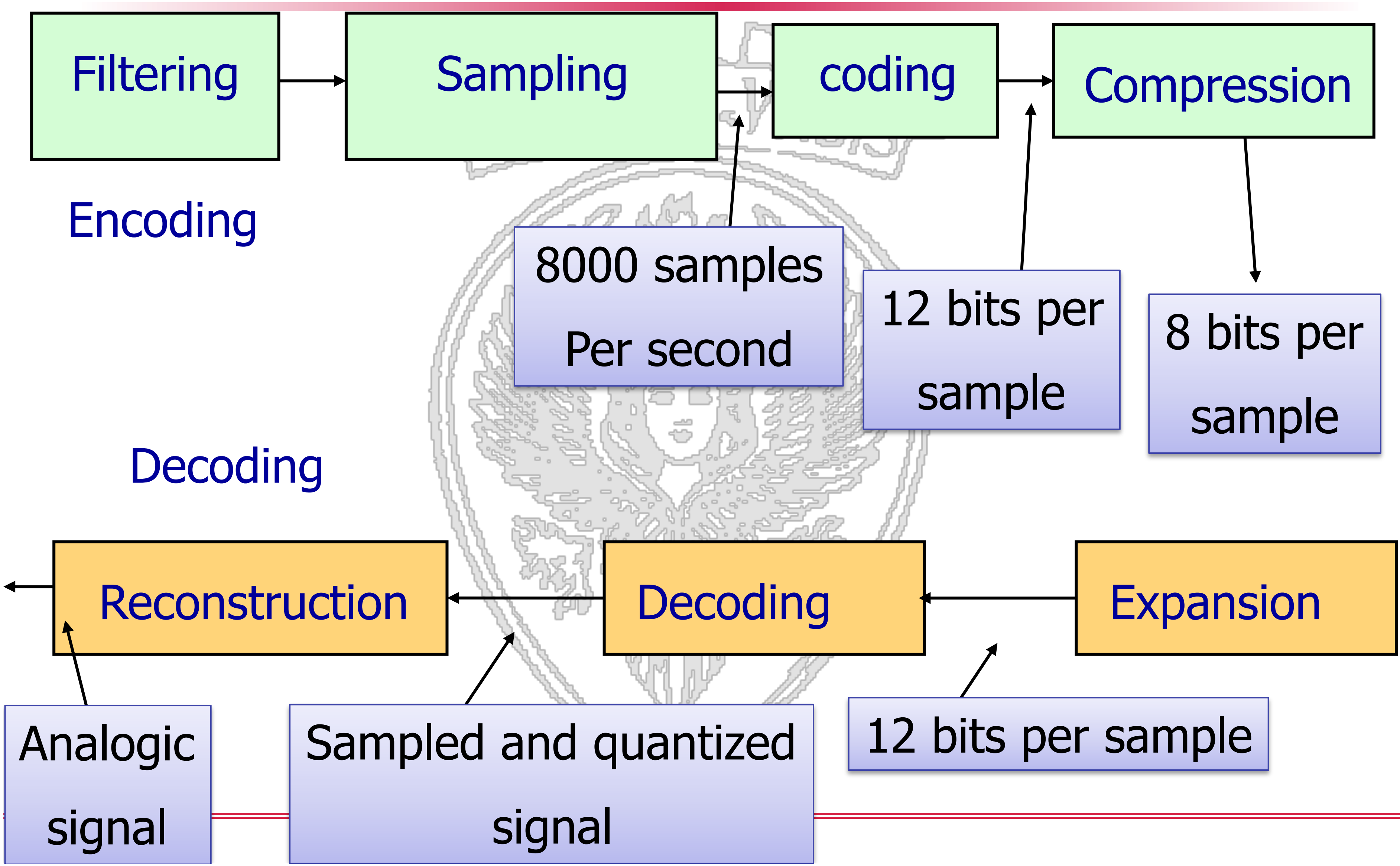
Fasi della codifica/decodifica



Waveform codecs

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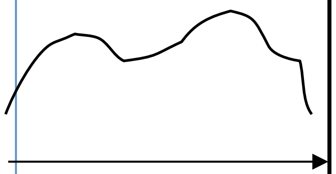

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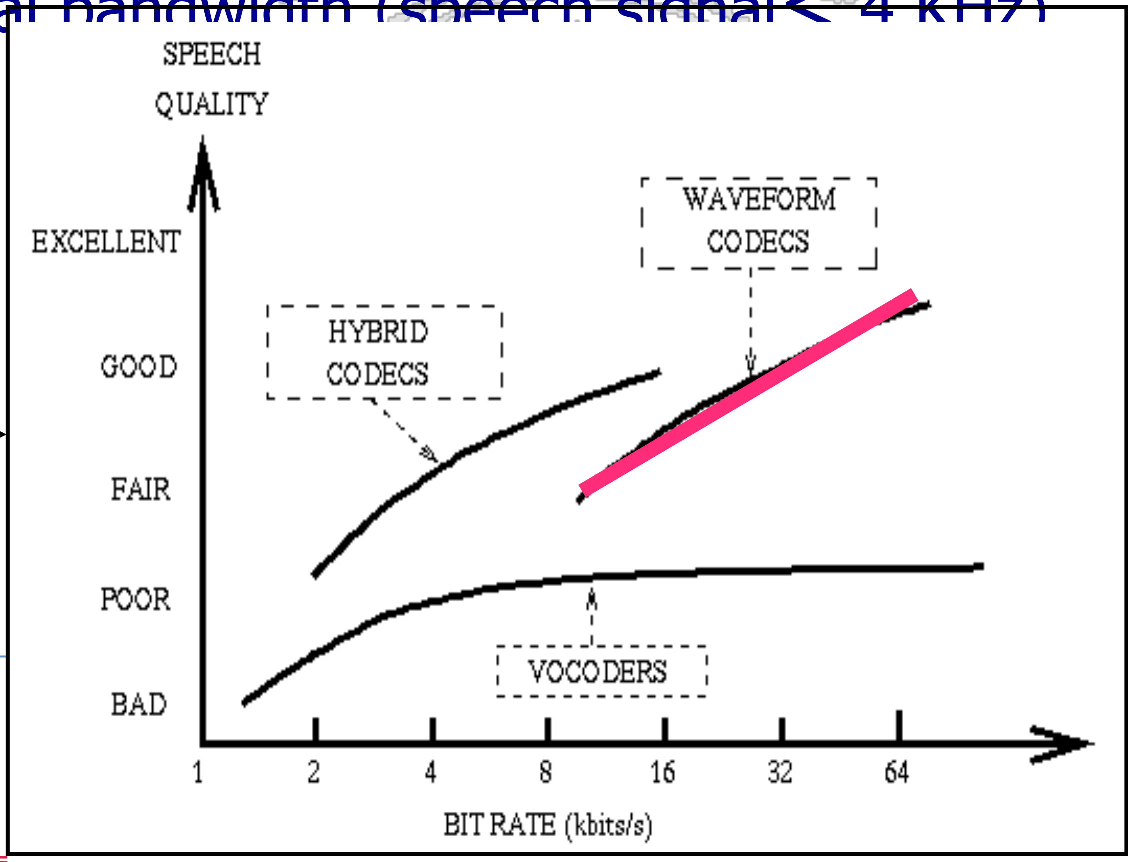

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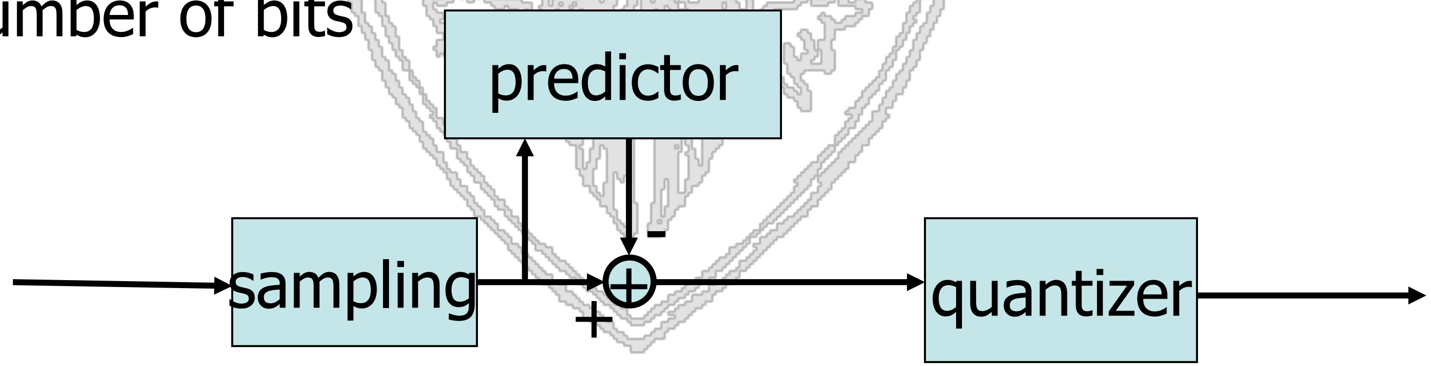
Differential PCM (DPCM)

- 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



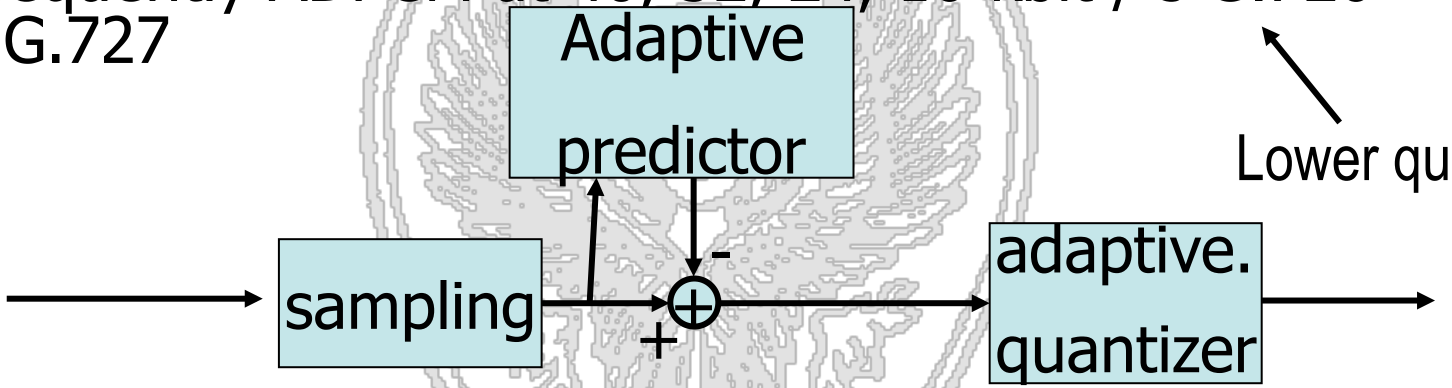
Adaptive DPCM (ADPCM)

- 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

Cordless

DECT

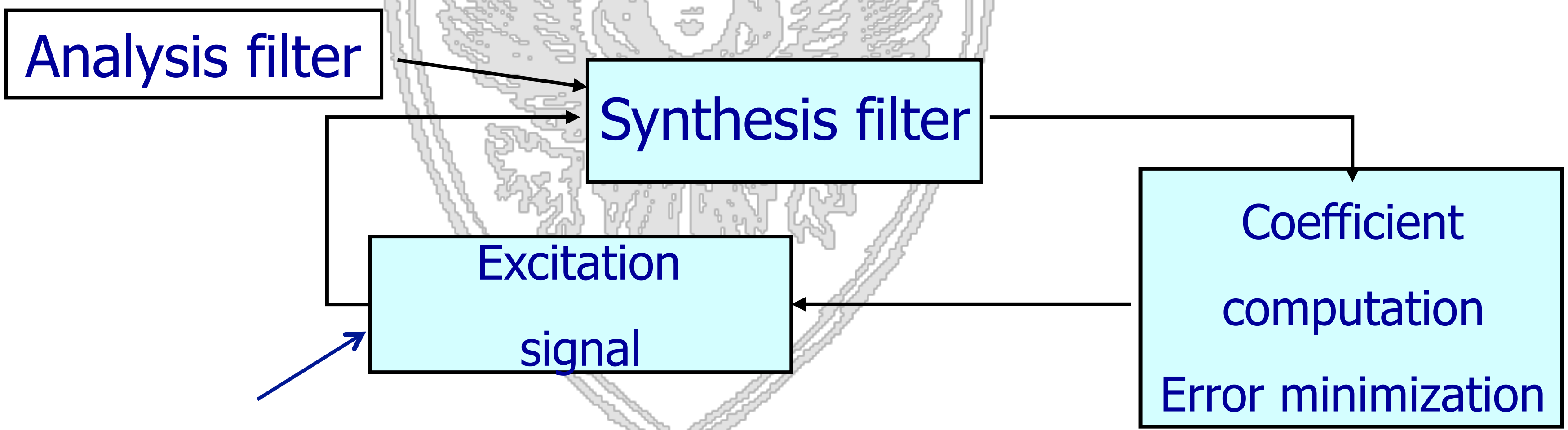
Lower quality



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)

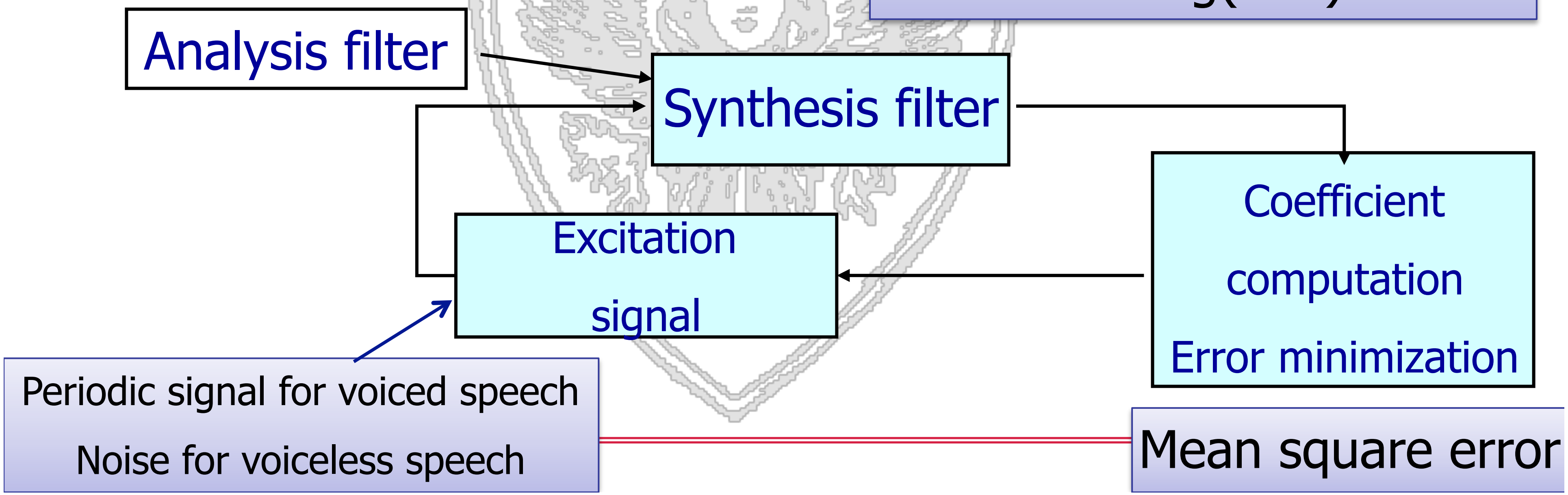
- They are based on models for the generation of the human voice
- models allow us to "remove redundancy" from the vocal segments to obtain an information base sufficient to reproduce the original voice signal (Idea: If we know the structure of the signal little information features will be enough to 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

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

Codebook Excited Linear prediction

LTP= Long term prediction

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.

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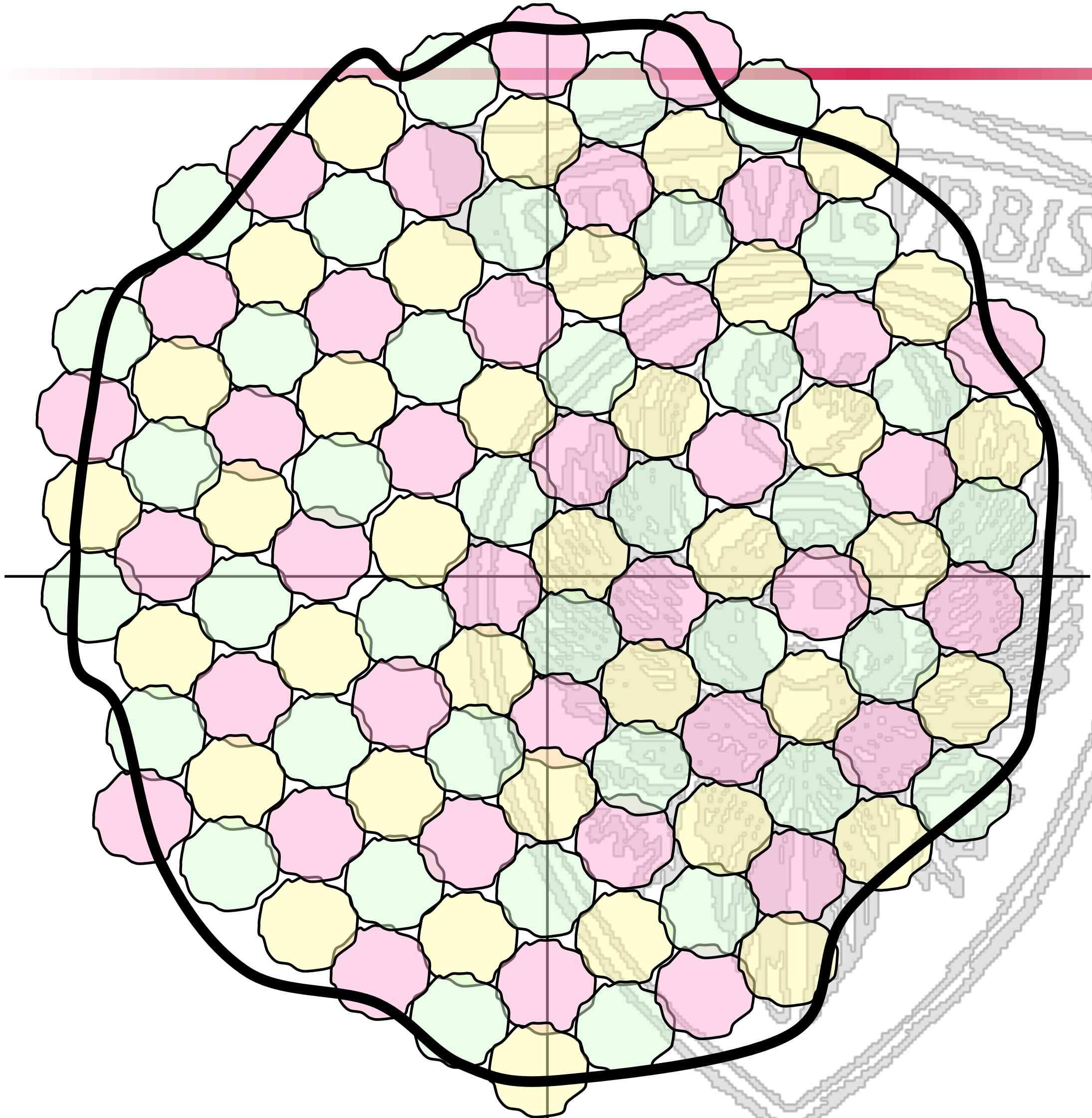
Mobility management

Cellular coverage (microcells)



many BS

Very low power!!

Unlimited capacity!!

Usage of same spectrum

(12 frequencies)

(4 freq/cell)

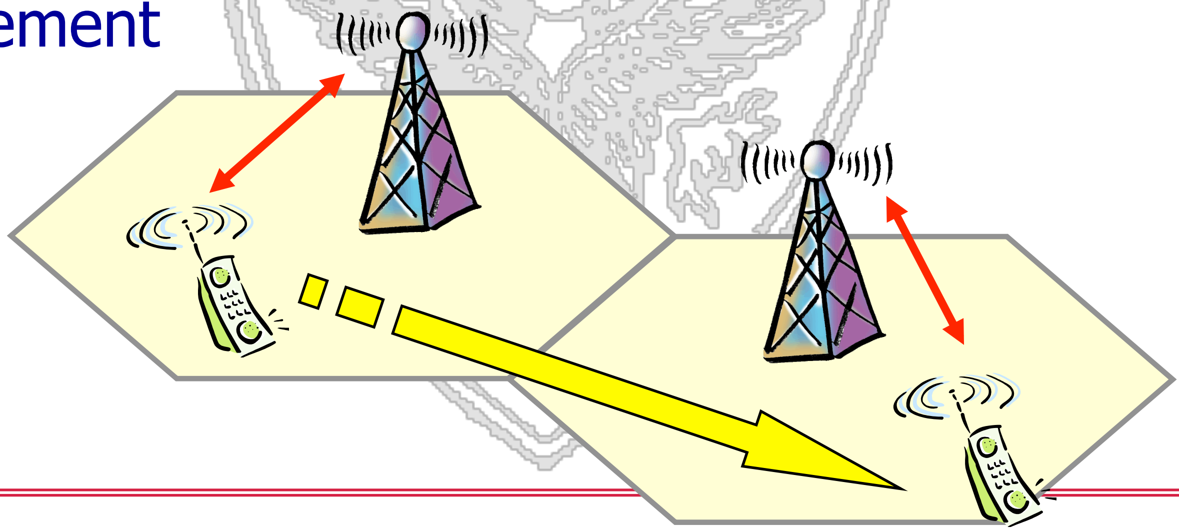
Disadvantage:

mobility management

additional infrastructure costs

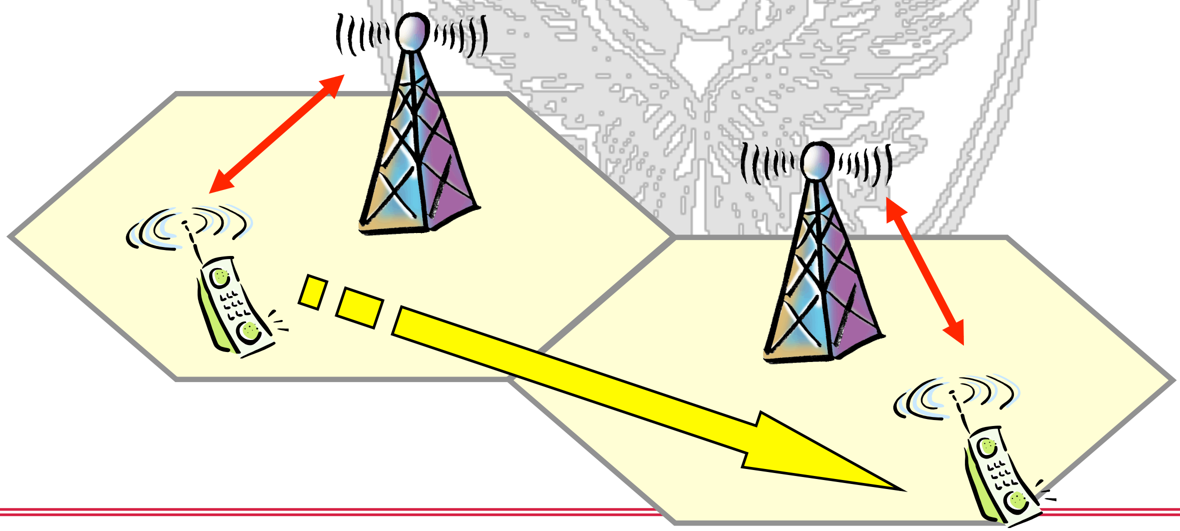
Mobility management

- 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)

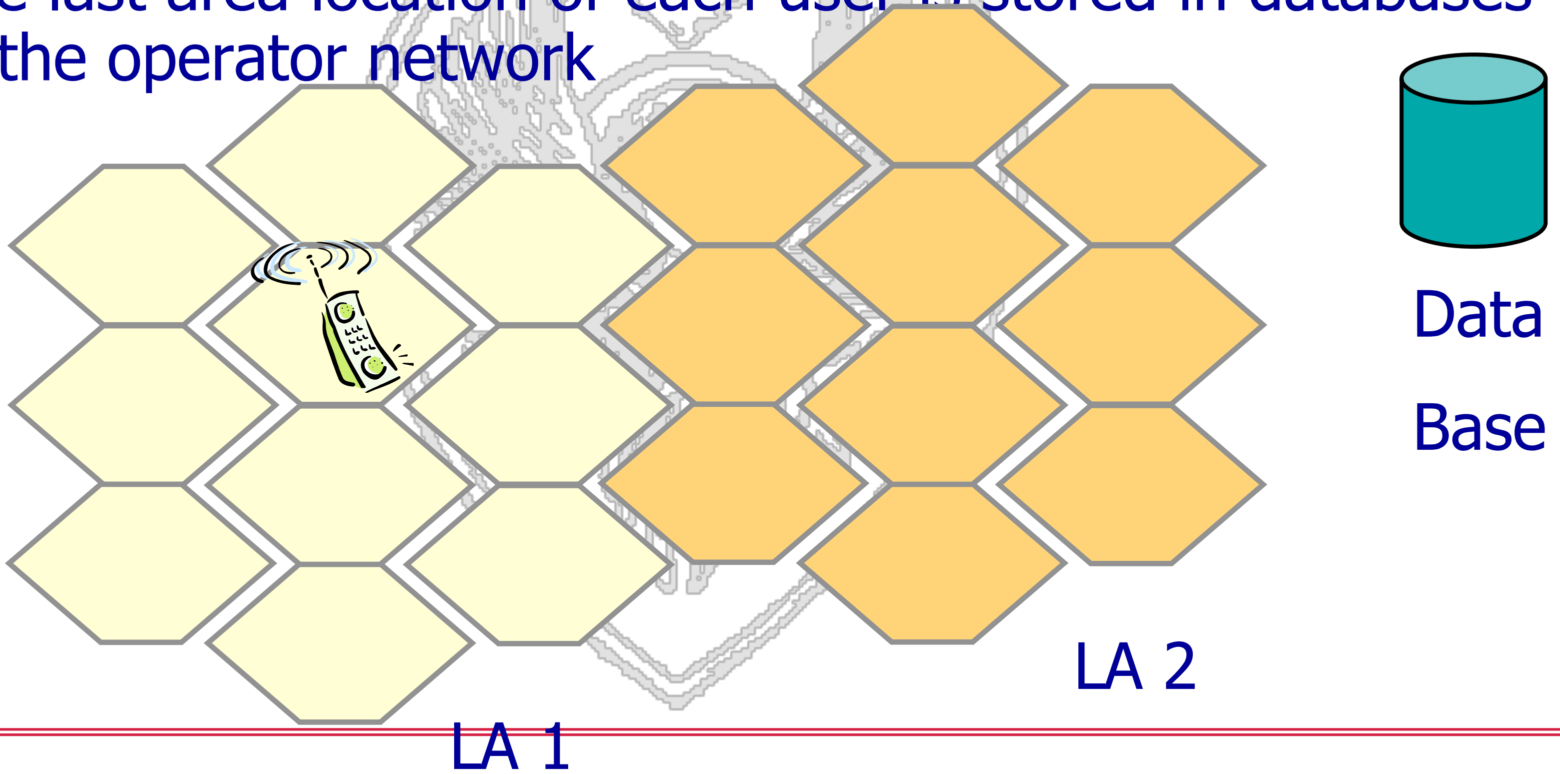
- IDLE: the user must be able to be located to be able to establish a call to/from him/her (*Location Update, Cell Selection, Cell Reselection*)

Cell selection

- A mobile terminal in idle mode "locks" to a cell on the basis of the signal received from the base station
- On a suitable common control channel the radio base station transmits the information of the system that, among other things, specifies its identifier
- The mobile terminal scans the radio frequencies to decode the control channel of the base stations in the area
- The terminal selects the base station from which it receives the strongest signal
- Periodically we continue to take steps (if the MS is in IDLE) on the signal received from the base station adjacent; if you get better from a different base station you make a cell reselection

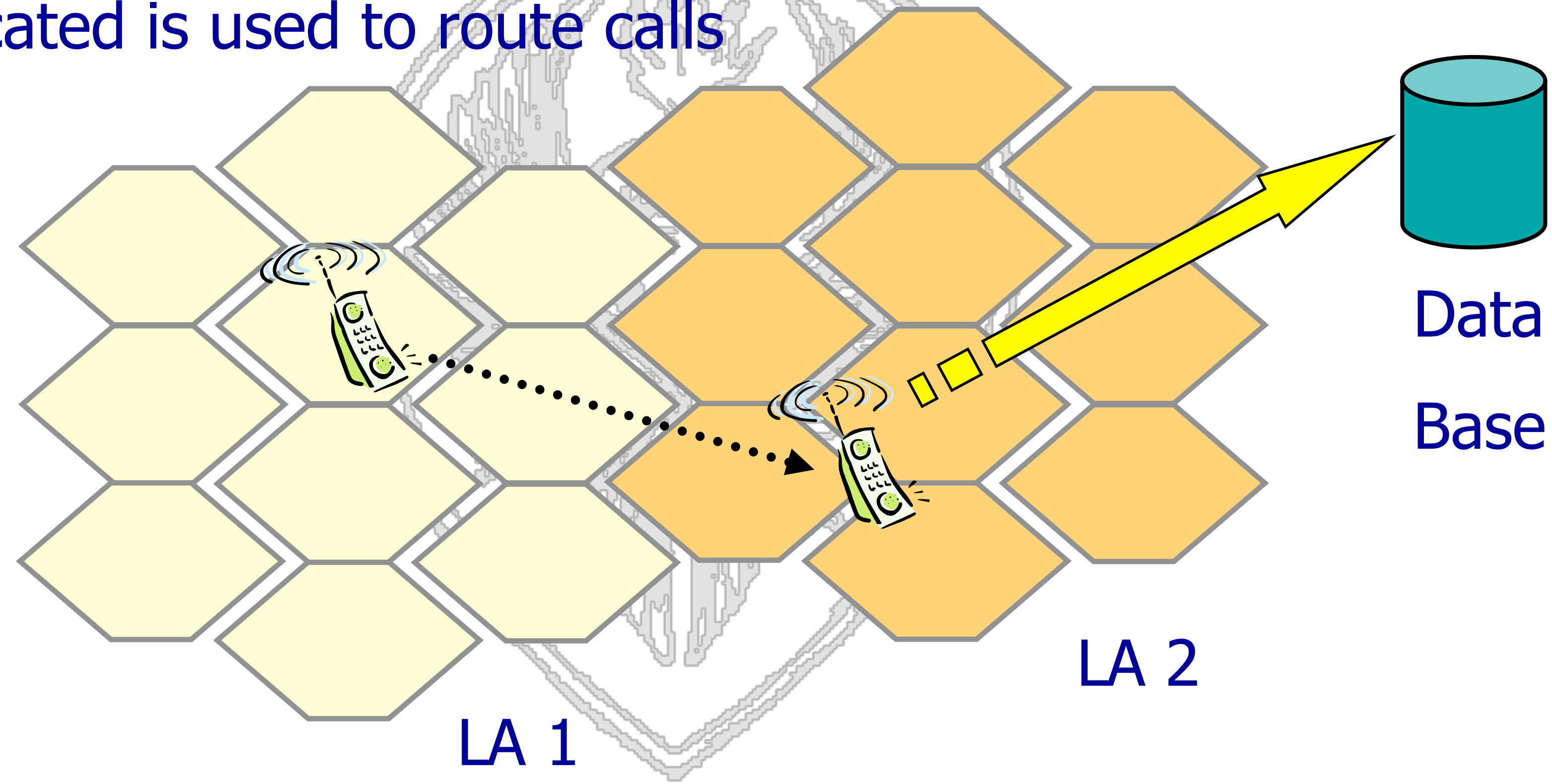
Location Update

- Location Area: topological entity that is hierarchically superior to the cell (group of several cells)
- An IDLE user is tracked by the system based on Location Area (and not on the basis of its cell)
- The last area location of each user is stored in databases of the operator network



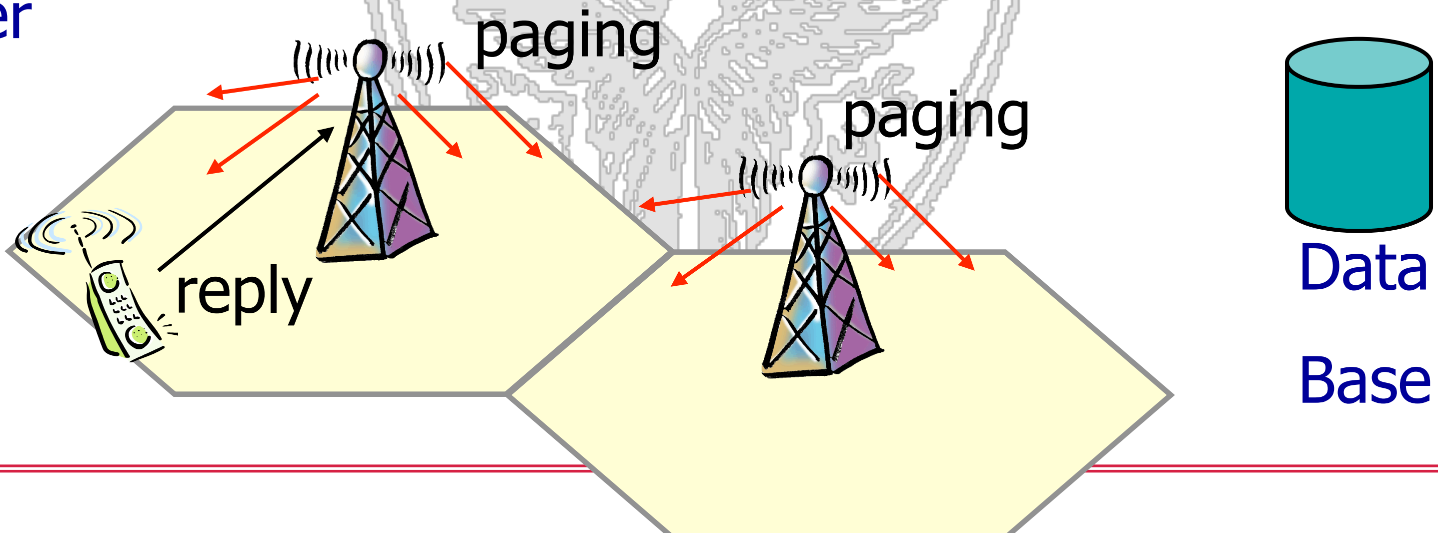
Location Update

- If a user in the IDLE state moves from one LA to another, it triggers a Location Update procedure
- The information about the LA in which a user is located is used to route calls



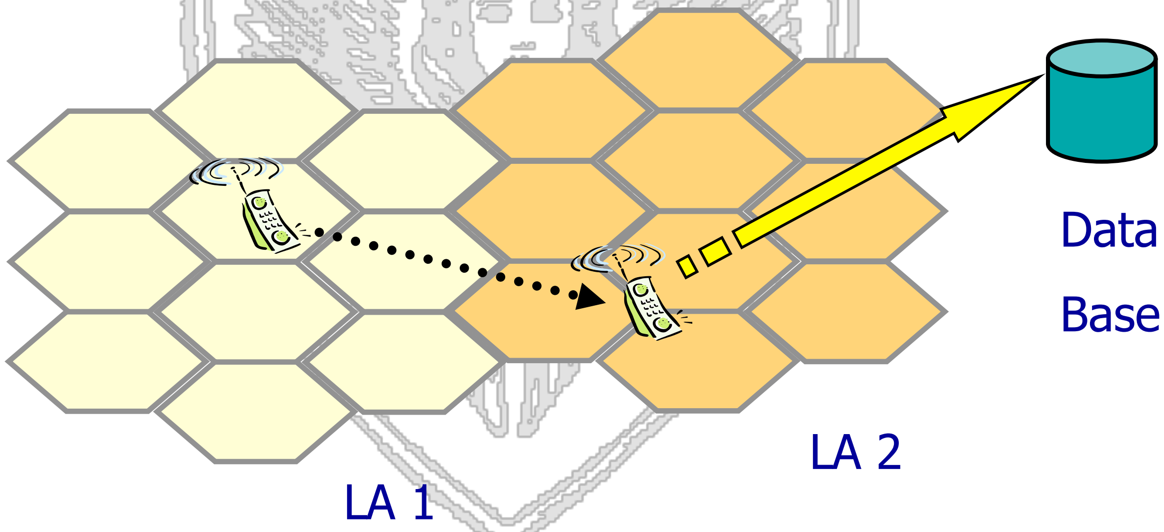
Paging

- When a call arrives for the mobile user the operatore is queried for the current user LA
- Once we know the location area where the user is located, the network initiates a paging procedure
- Each base station in the LA sends a control message broadcast with the ID of the user ID
- When the mobile terminal answers the network knows the cell and routes the call through the cell BTS till the mobile user



Paging vs. Location Update

- QUESTION:
- How big should the Location Area be?
  - small
  - large
- What drives in one direction, what in the other?

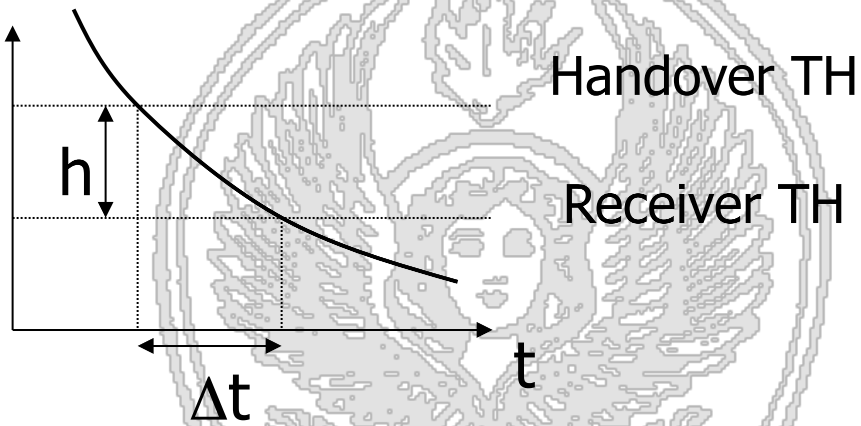


Handover

- 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

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 Δt 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

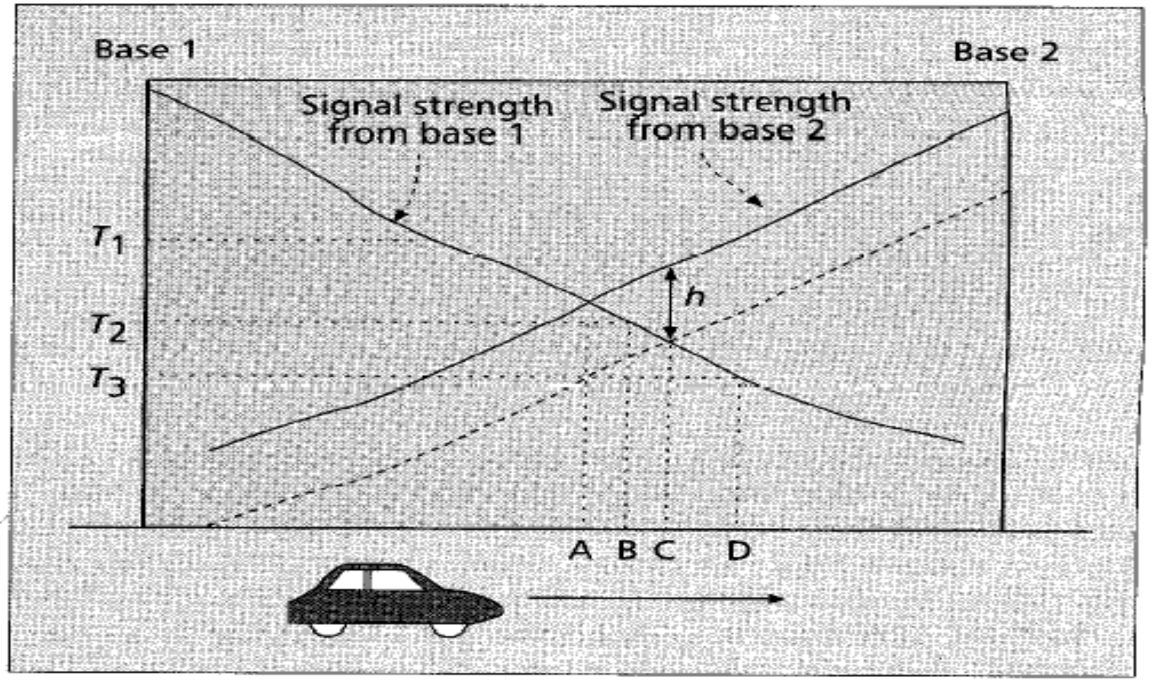
When to trigger an handover?

- 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)

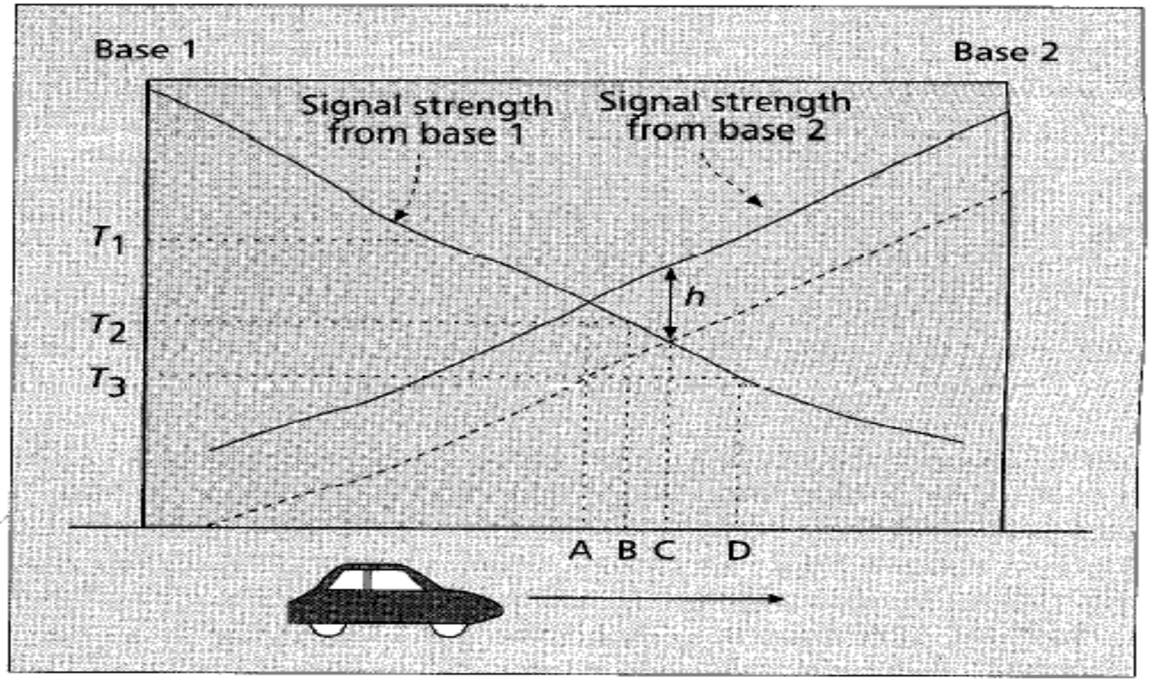


When to trigger an handover?

- 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

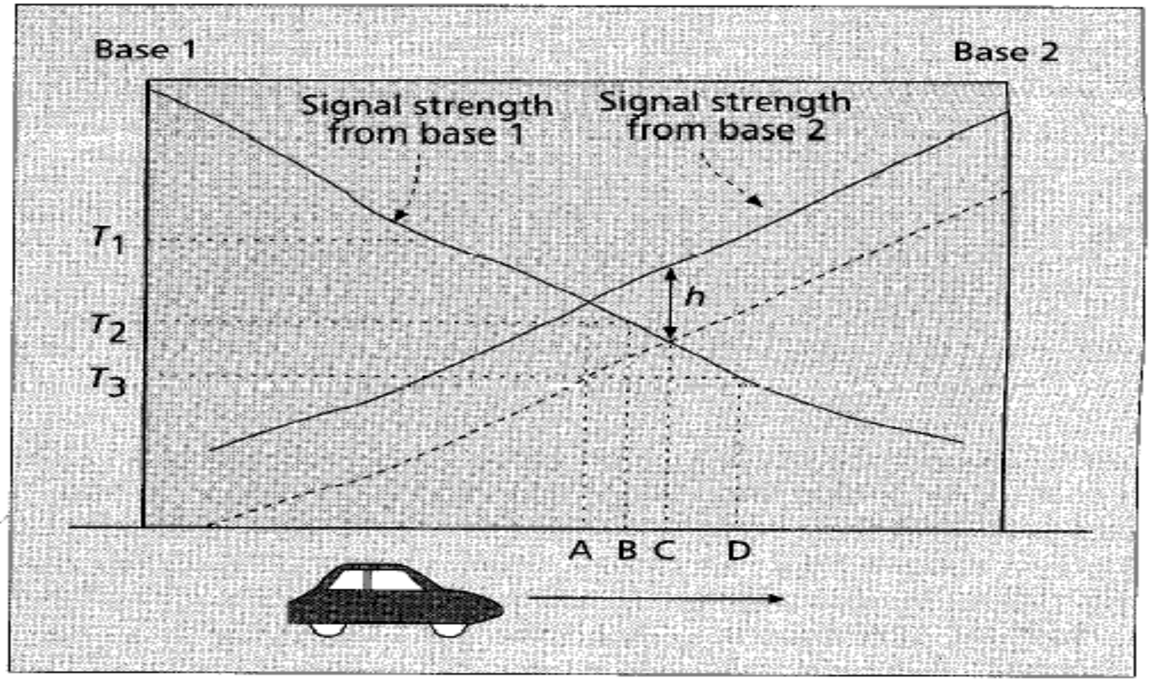


When to trigger an handover?

- There are several methods

3 - method of the strongest signal with hysteresis

- if the power of the other BS is stronger than a value ≥ 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

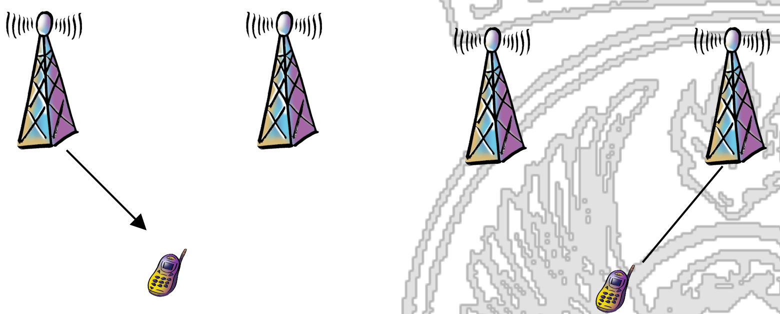
Handover performance guard channels technique

- 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?)

- 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.

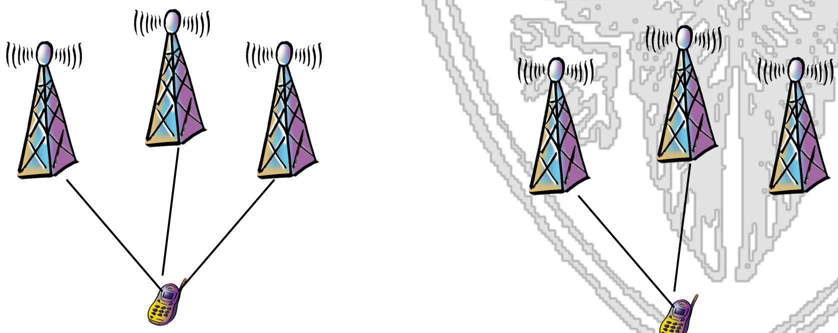
Types of handover

- Hard Handover (GSM-2G)



Removal and establishment of a new radio link

- Soft Handover (UMTS-3G)



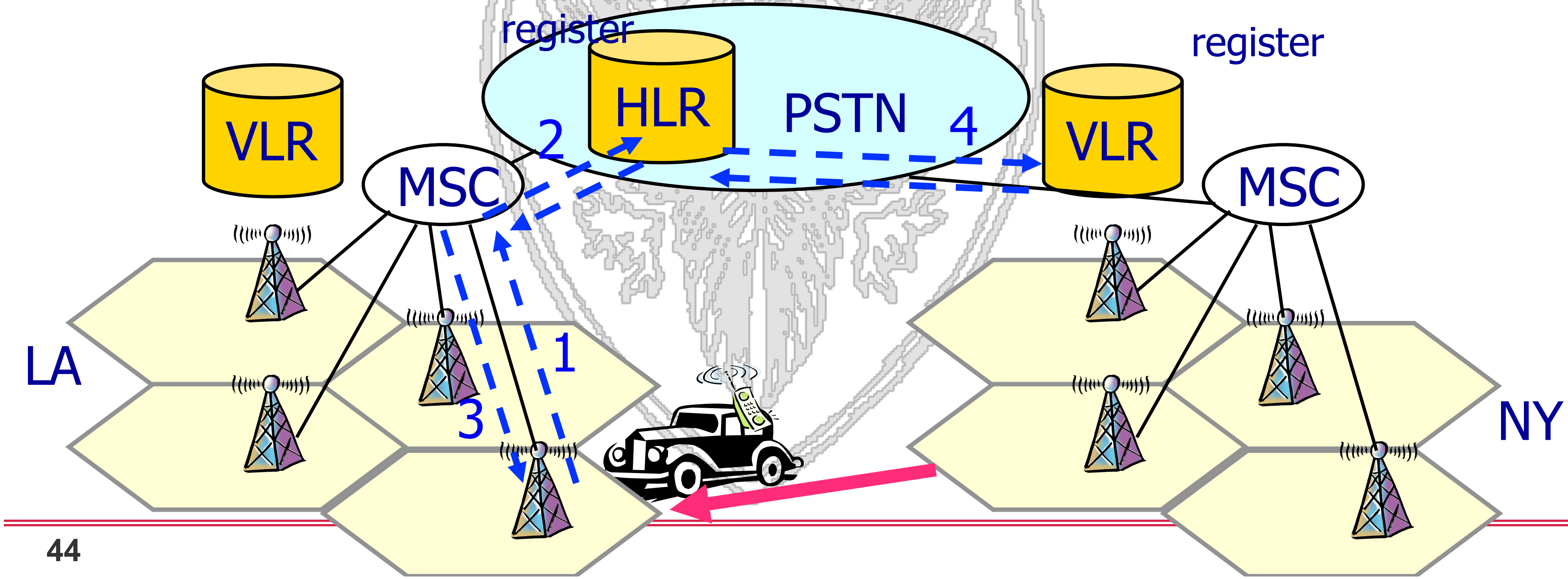
Leveraging on the user macrodiversity, the user is simultaneously connected to several base stations

Roaming

1. Upon arrival in a new LA, the user must register with the new VLR
2. The new VLR informs the HLR of the user's new location. The HLR sends back an ack with information such as the user's profile
3. The new VLR informs the user of the successful registration
4. The HLR sends a deregistration message to the old VLR

HLR: Home location register

VLR: Visiting location register



Call set up : example

1. MS→fixed phone through the VLR of the MS
2. Fixed phone→MS:through the gateway MSC of the MS the caller contacts the HLR and , throught it, the VLR currently managing the user
3. VLR provides information such as a routing number, LA of the user
4. Call set up