IoT, Course introduction

Internet of Things a.a. 2019/2020

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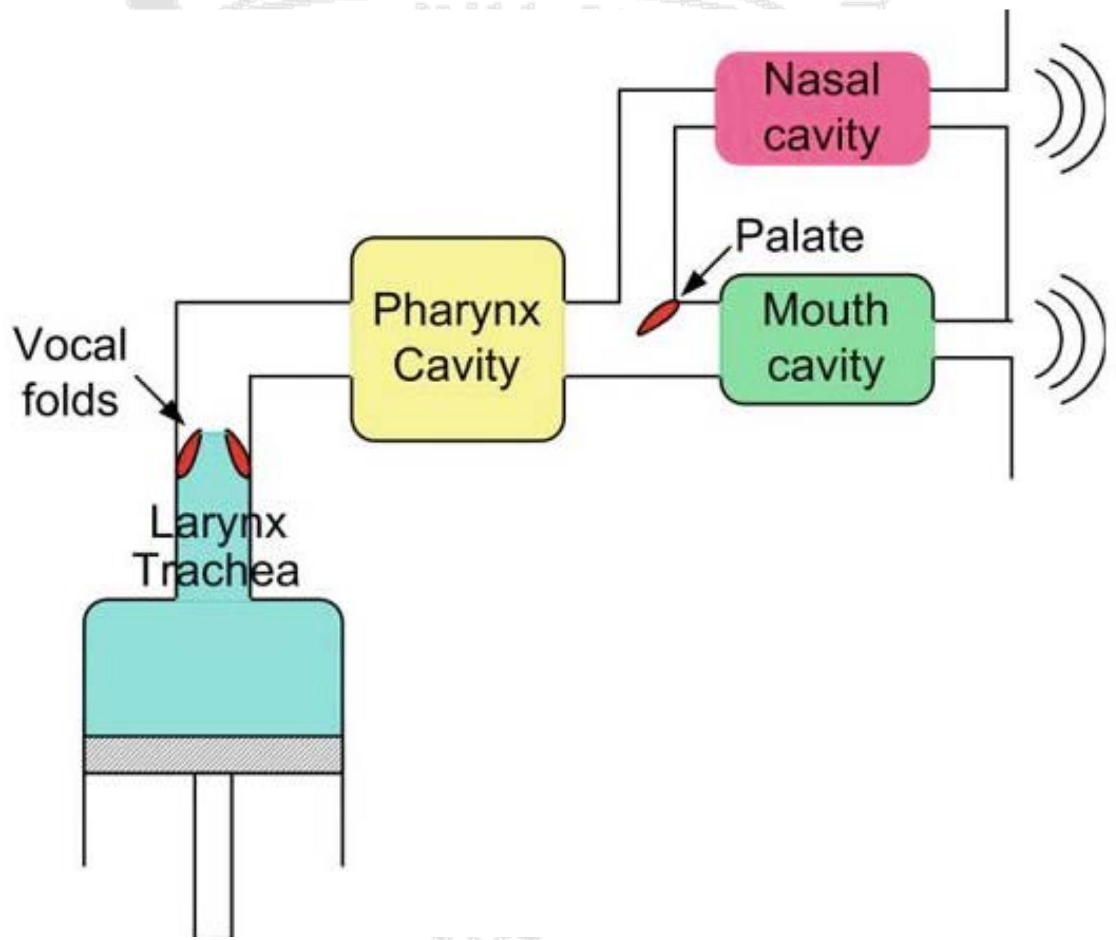
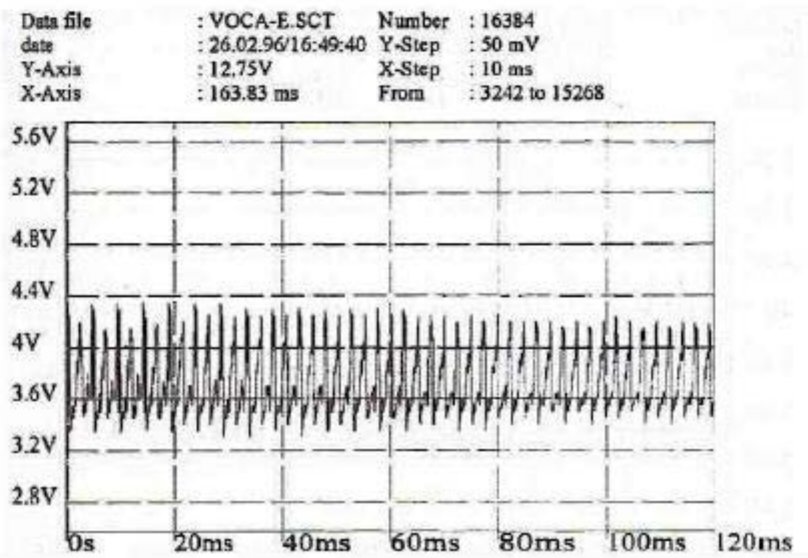
Voice Coding

Speech signals

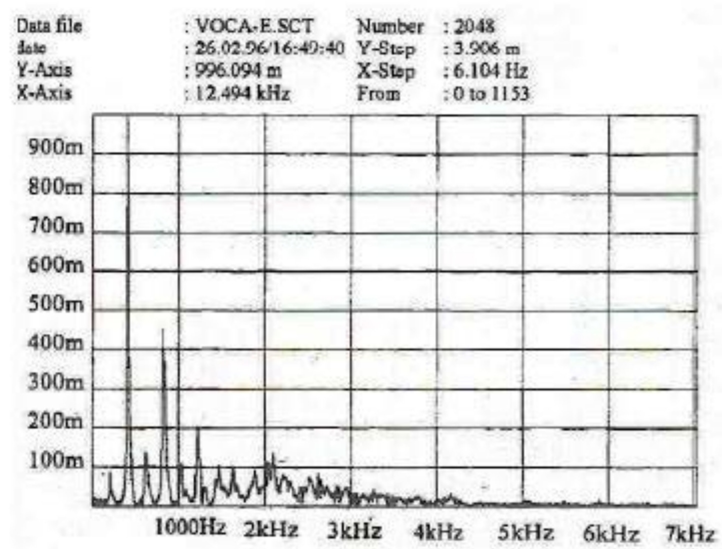


Voice coding: voiced sound

Time- frequency features, vowel «e»



Signal in time



Signal spectrum (frequency components)

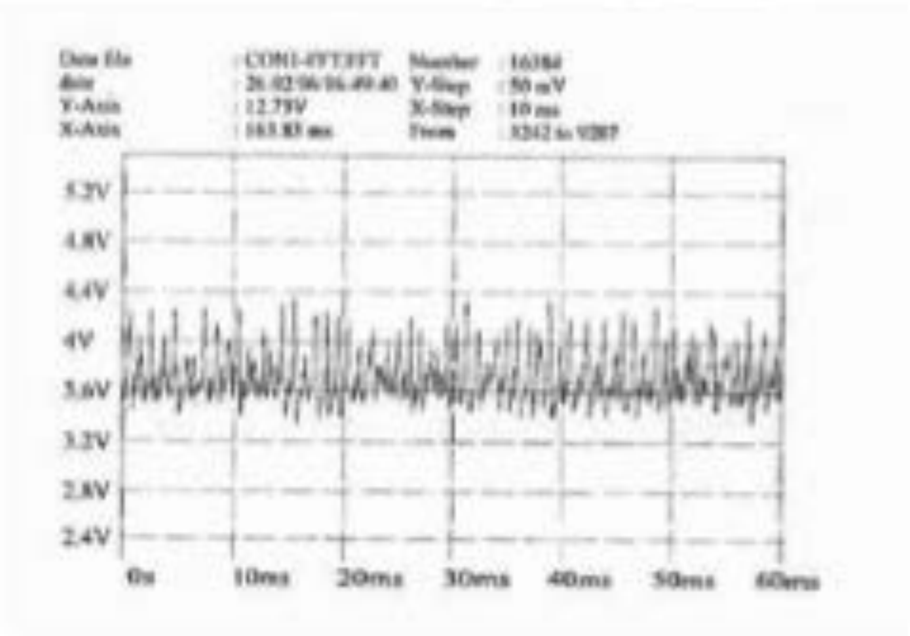
- Sounds produced by the vibrations of the vocal folds
- Features:

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

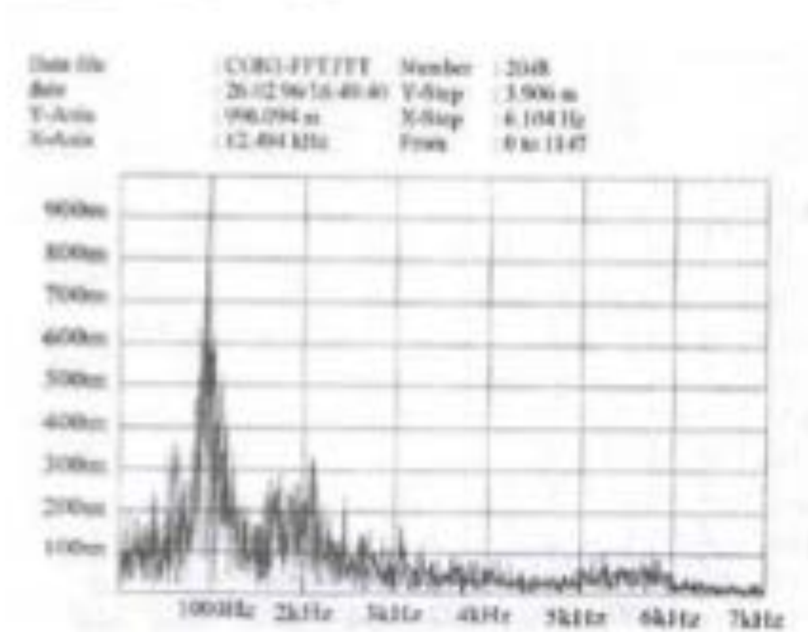
  4) low number of frequencies around which the energy is concentrated (formant frequencies) 5) formant frequencies are low frequencies

Voice coding: voiceless sound

Time- frequency features, consonant f



Signal in time

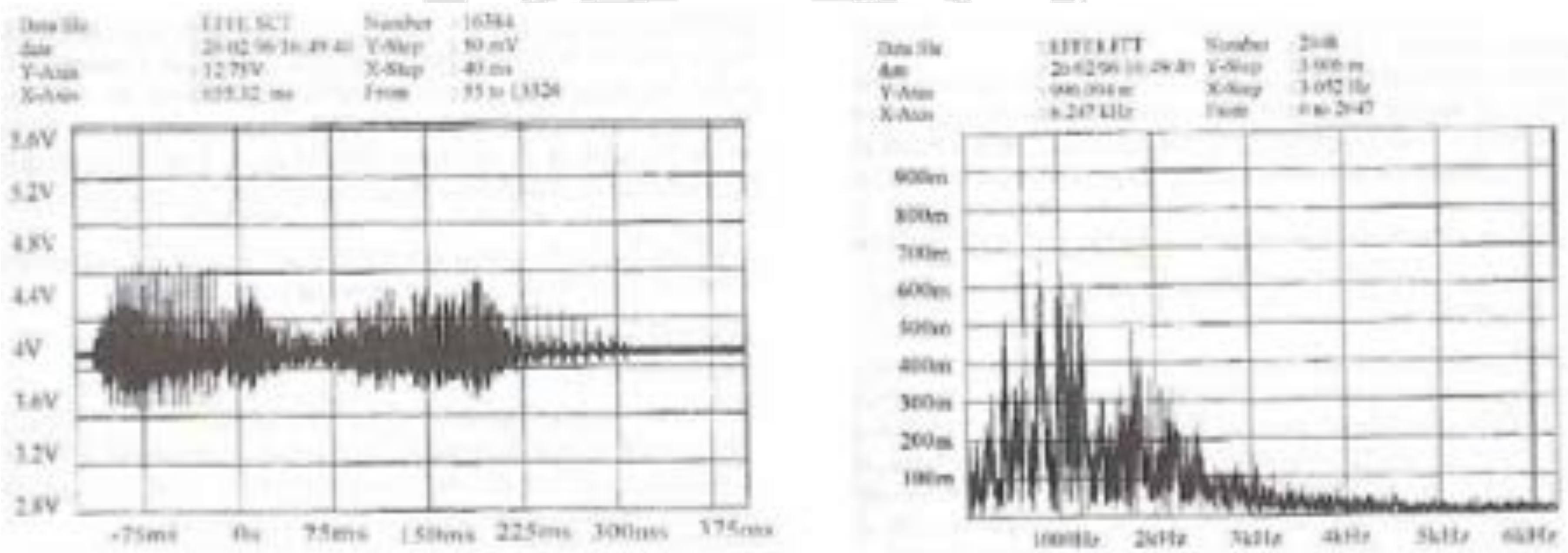


Signal spectrum (frequency components)

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

Voice coding

One word: effe



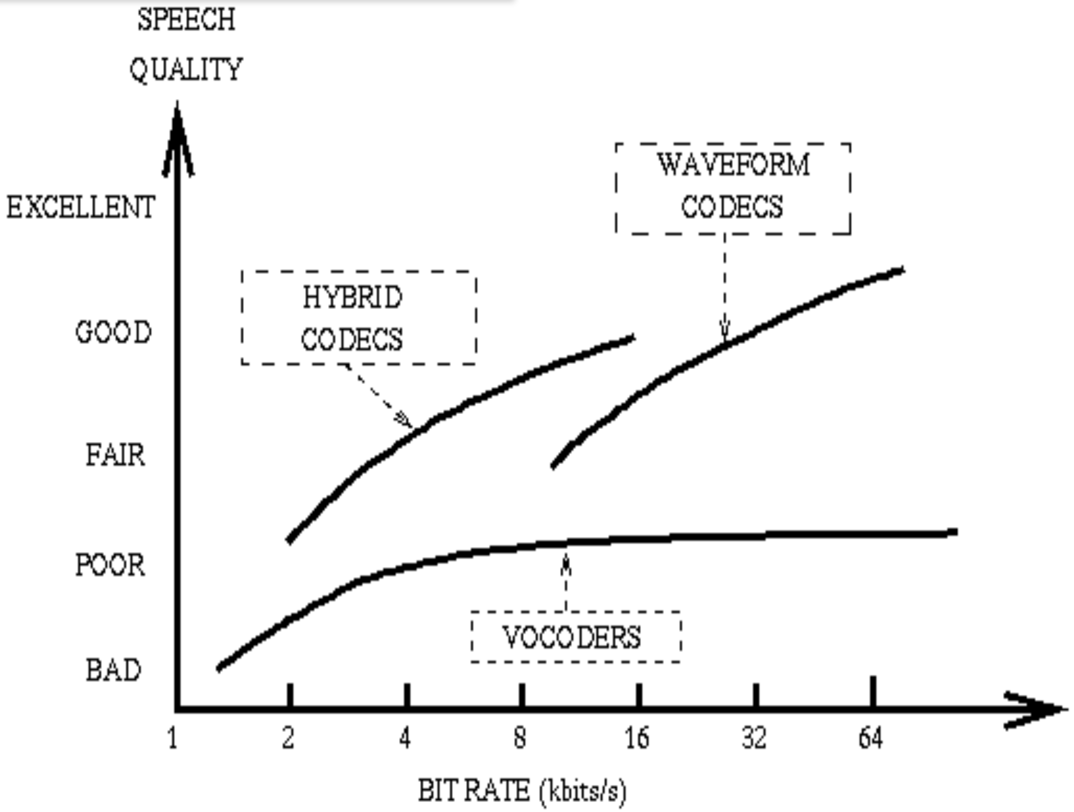
- Vowels and Consonants have different amplitudes.
- The most significant frequency components are located between 300Hz and 3400Hz, with (small) spectral components till 7KHz

Voice coding

Speech signal is translated into a sequence of bits

Mean Opinion Score

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



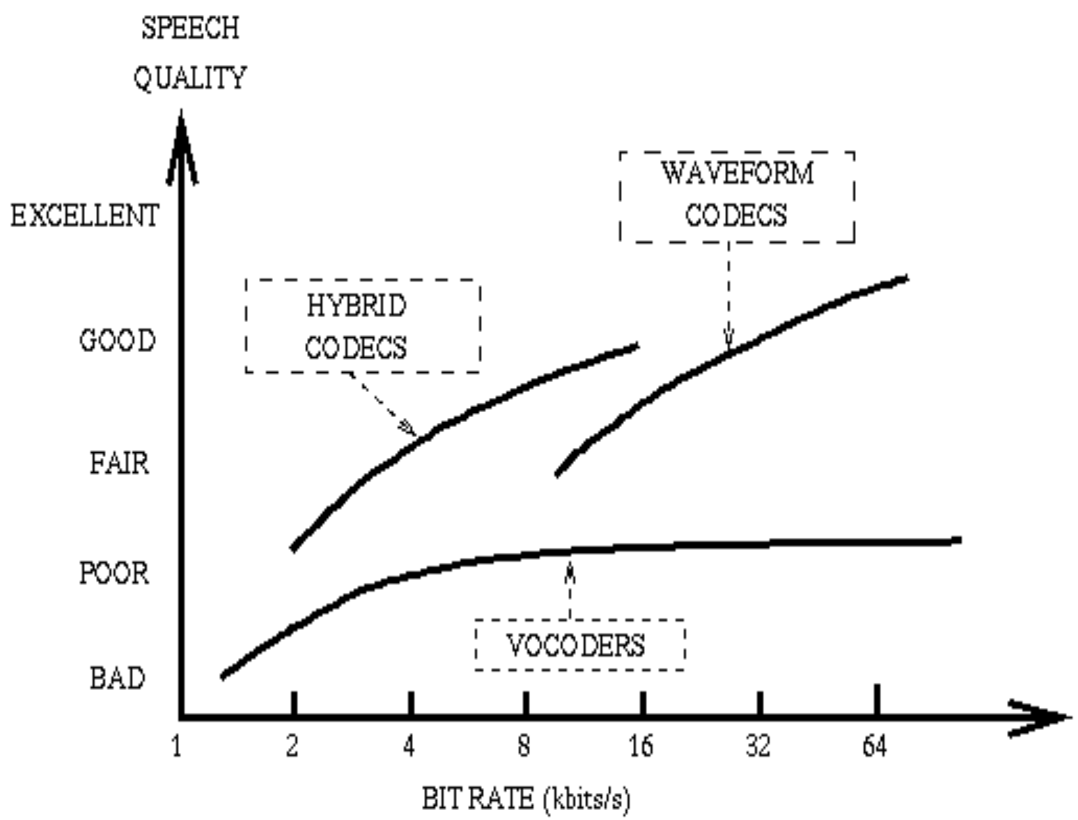
Voice coding

Speech signal is translated into a sequence of bits

Digitization of an analog signal ←

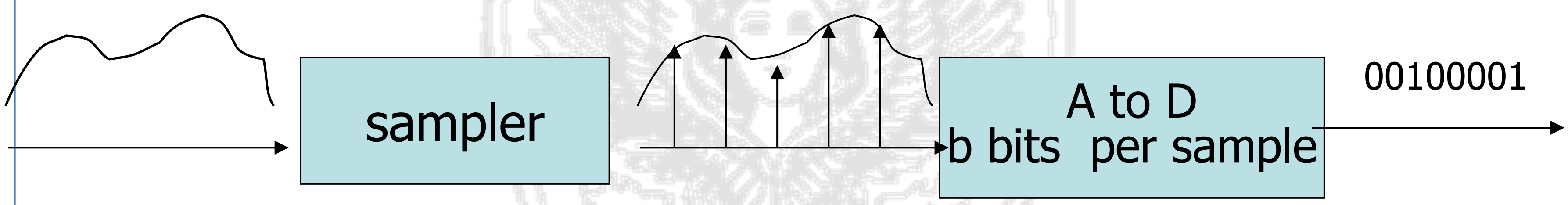
They make an explicit description of the waveform input (es. PCM)

- 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),

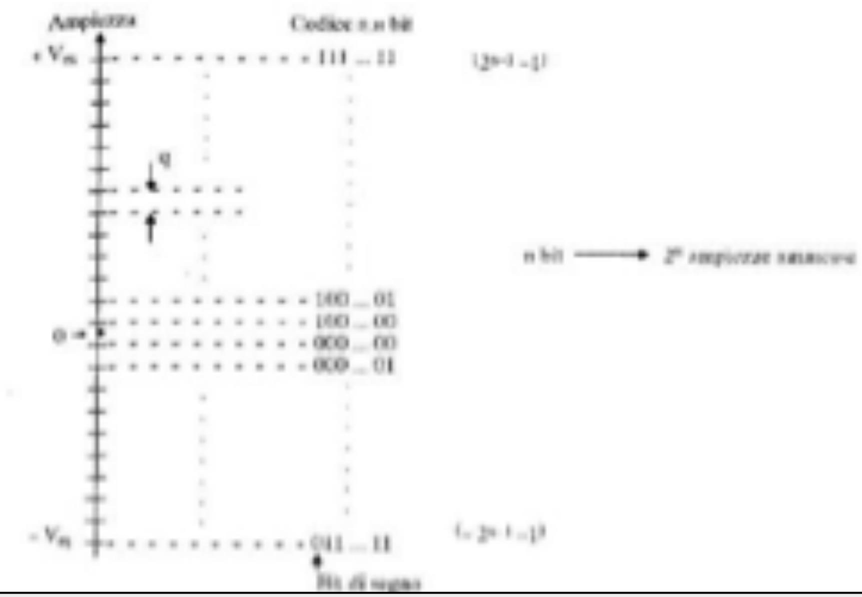
robustness to errors and background noise

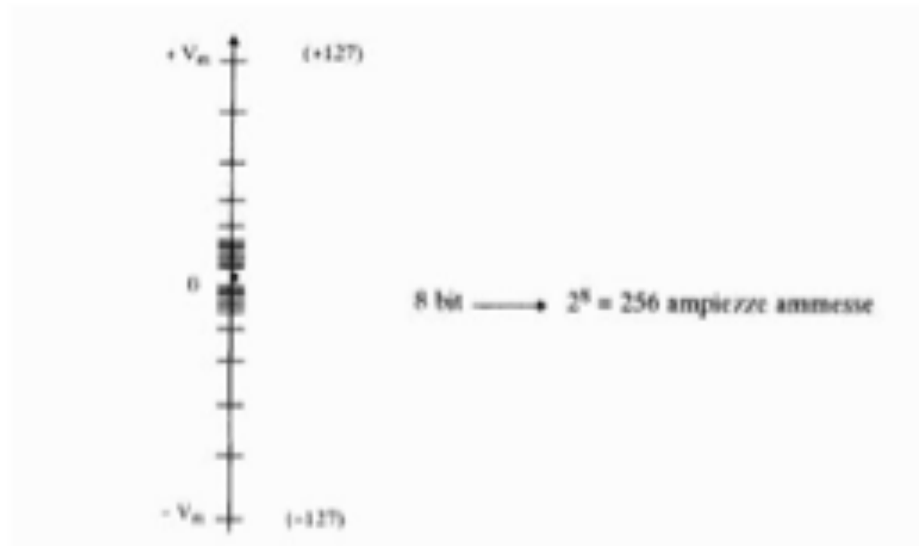
Waveform coding: Pulse Code Modulation (PCM)

- 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

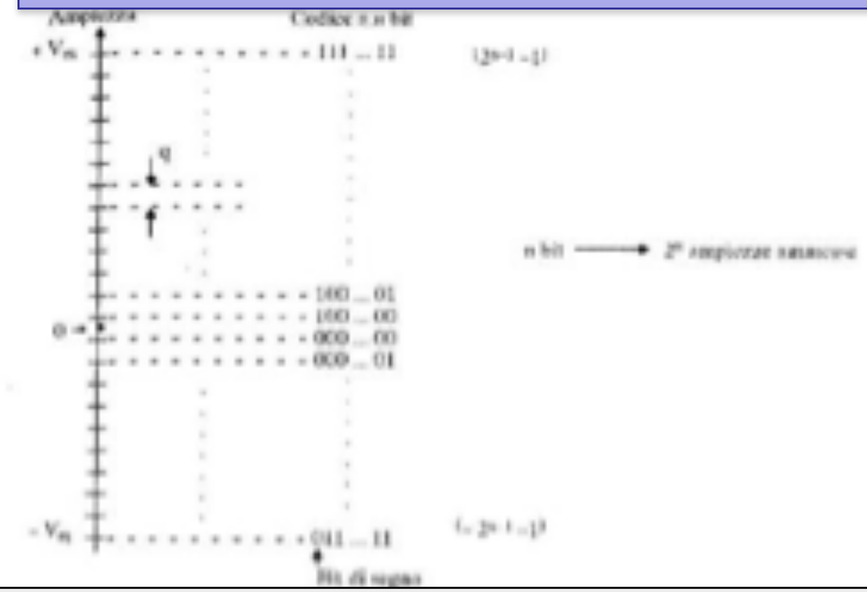




- Quantization error is fixed (<q/2, where q is the quantization step)
- 12 bit per sample are needed to achieve a quantization error low enough also for small values

Quantization

The axis of the amplitudes is divided into equal intervals



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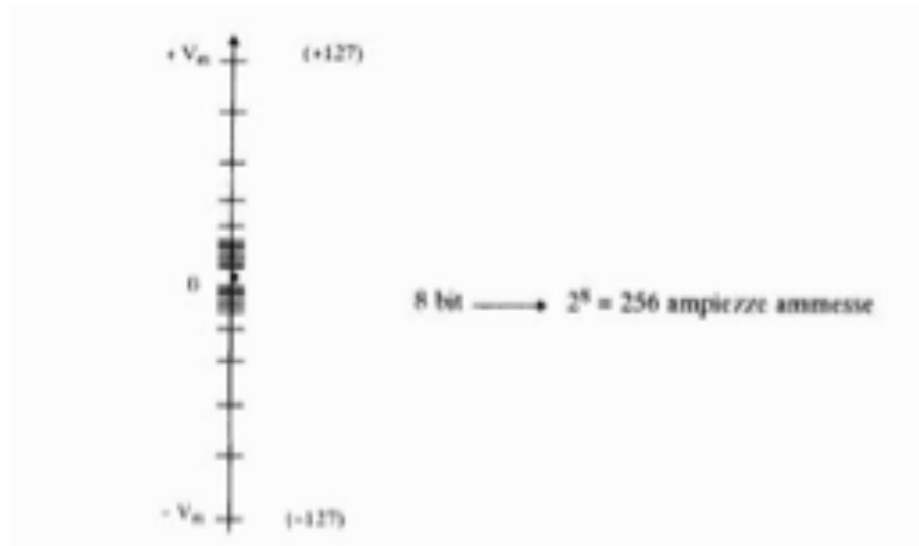
Non - Uniform quantization

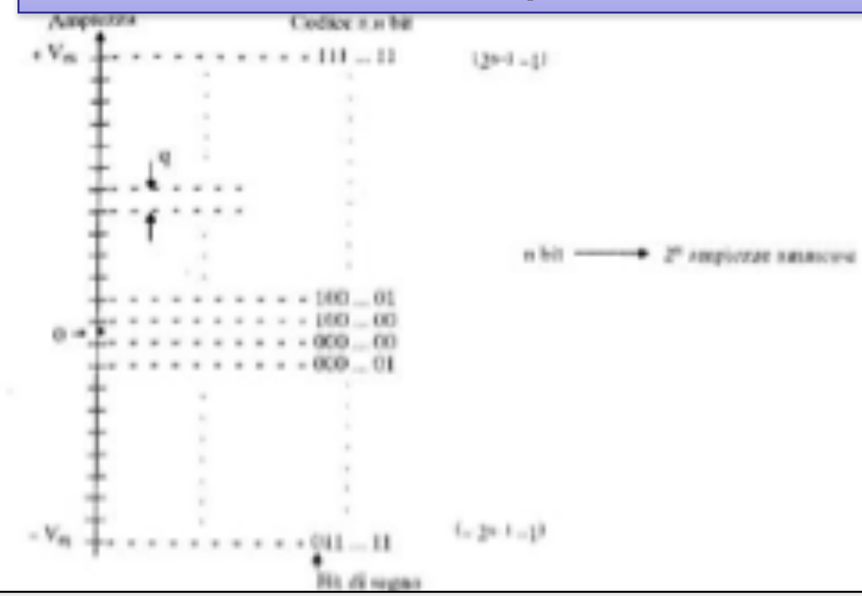


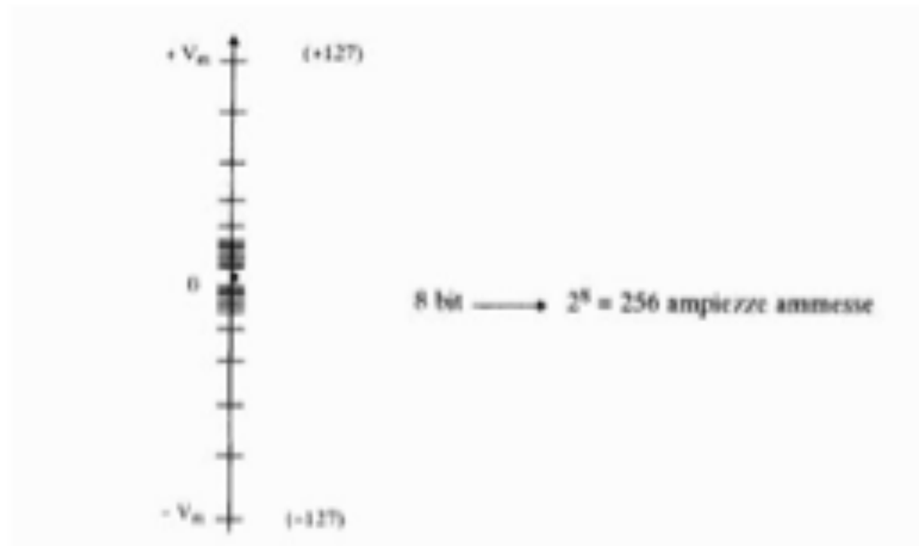
- Many relatively small values
- Higher quantization errors can be tolerated in case of high values
- 8 bits per sample results in excellent perceived quality

Quantization

The axis of the amplitudes is divided into equal intervals

Non - Uniform quantization





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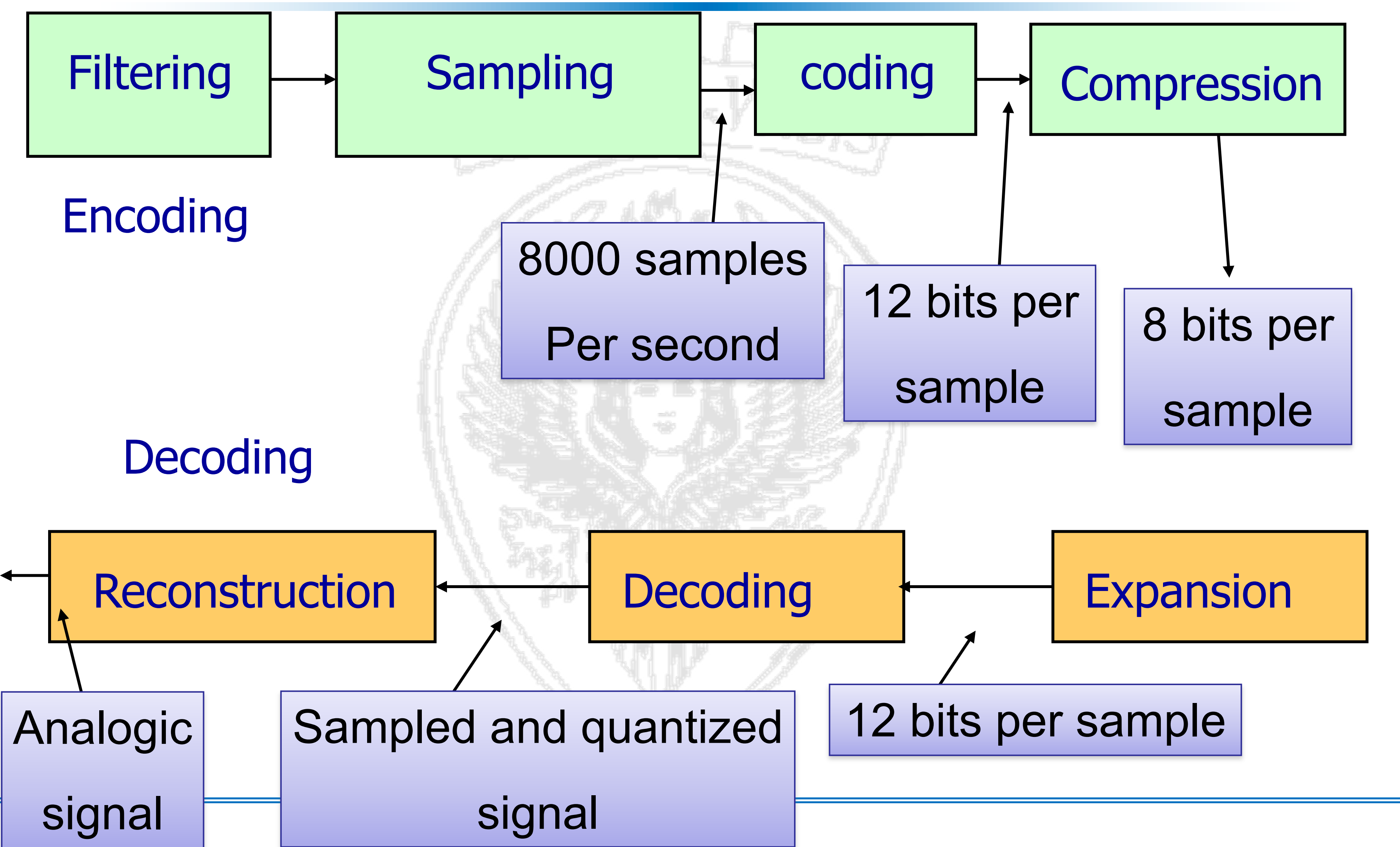
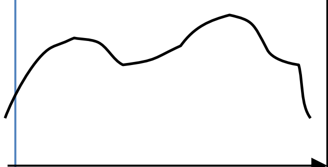
compression

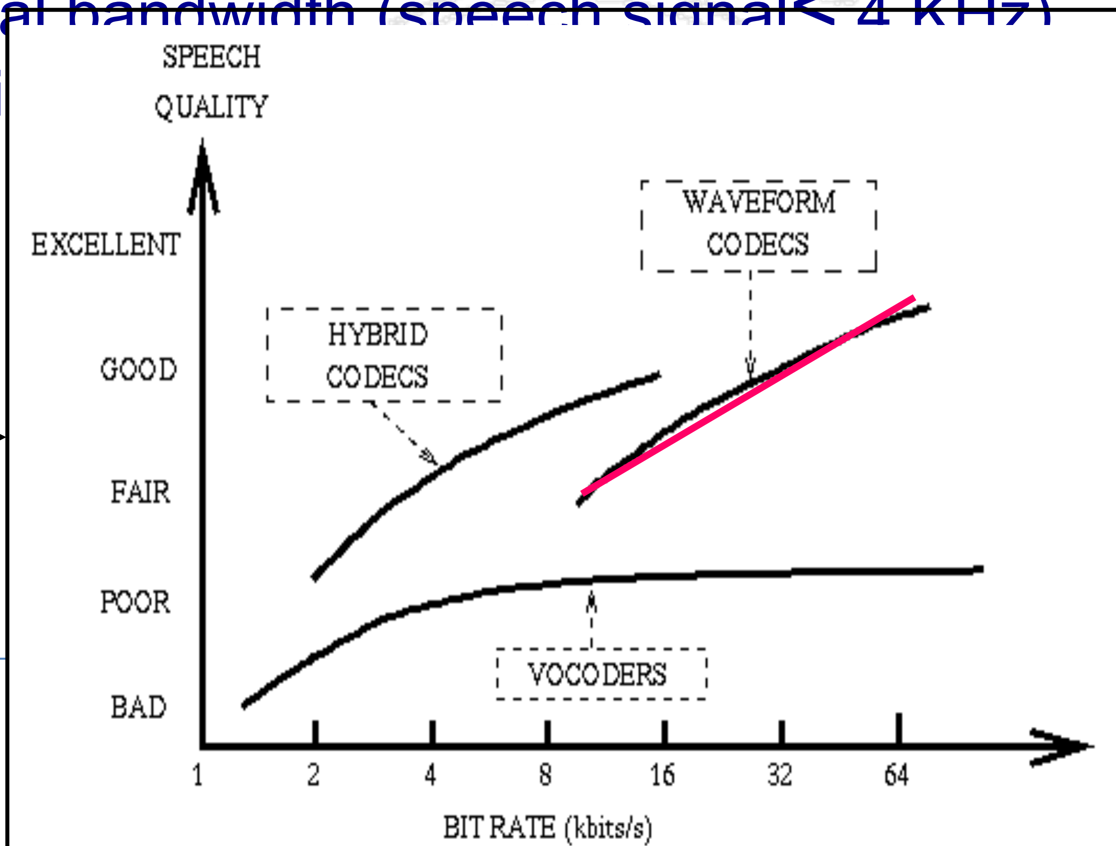
Fasi della codifica/decodifica



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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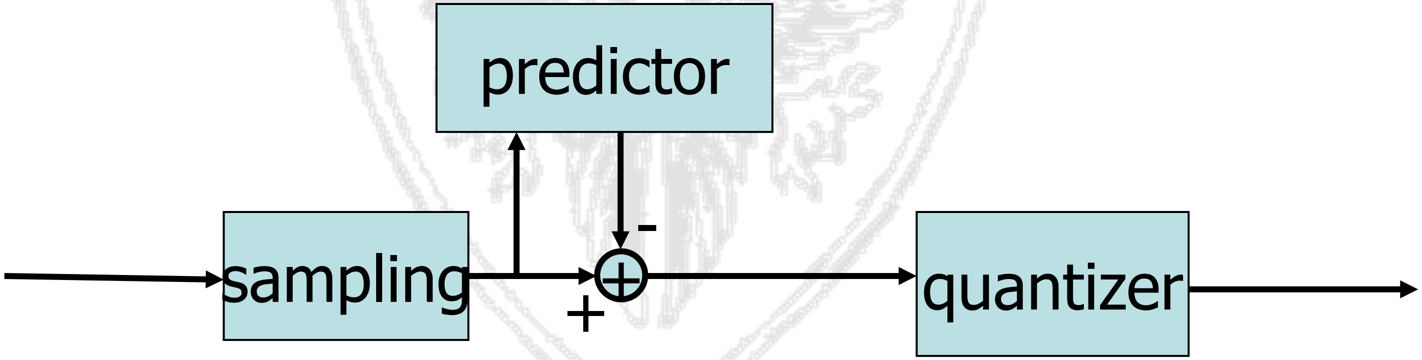
Differential PCM (DPCM)

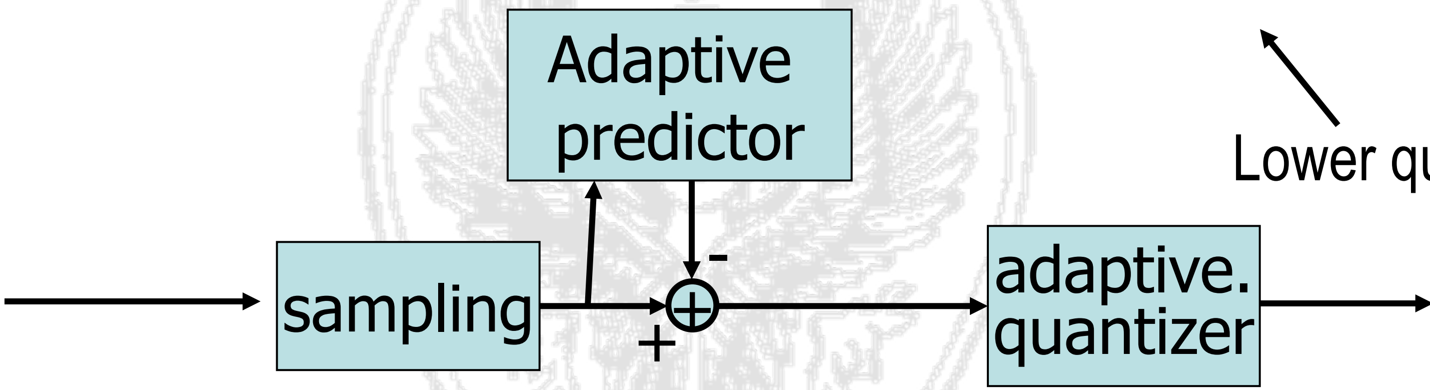
- the subsequent voice samples are correlated
- You can use prediction methods for evaluating the next sample known previous
- transmitting only the difference between the predicted value and the actual value
- because of the correlation the variance of the difference is smaller and it is possible to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

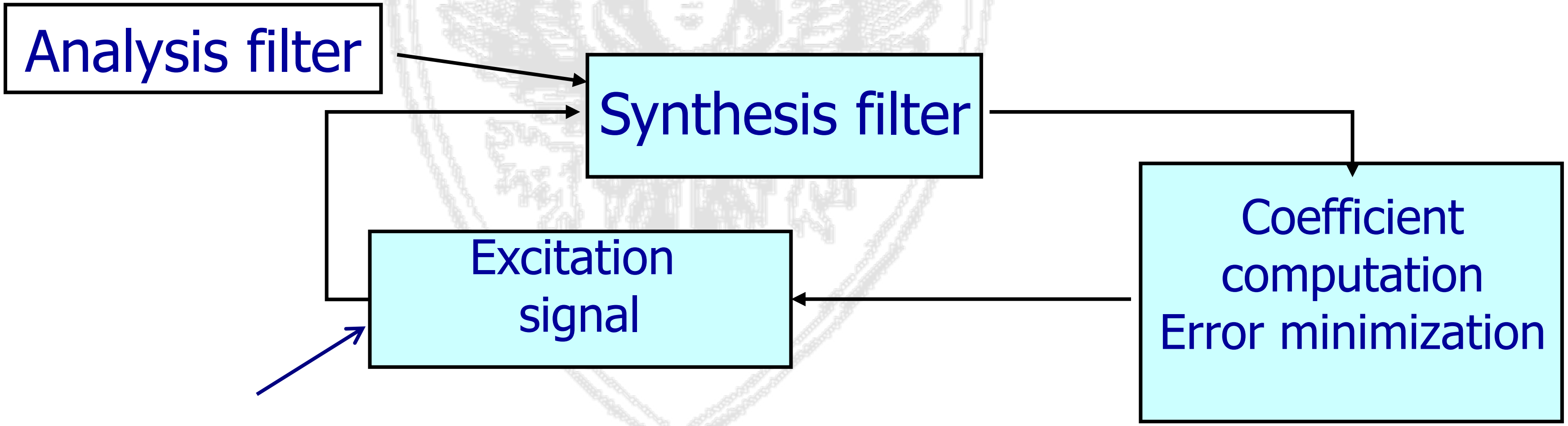


Lower quality

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)

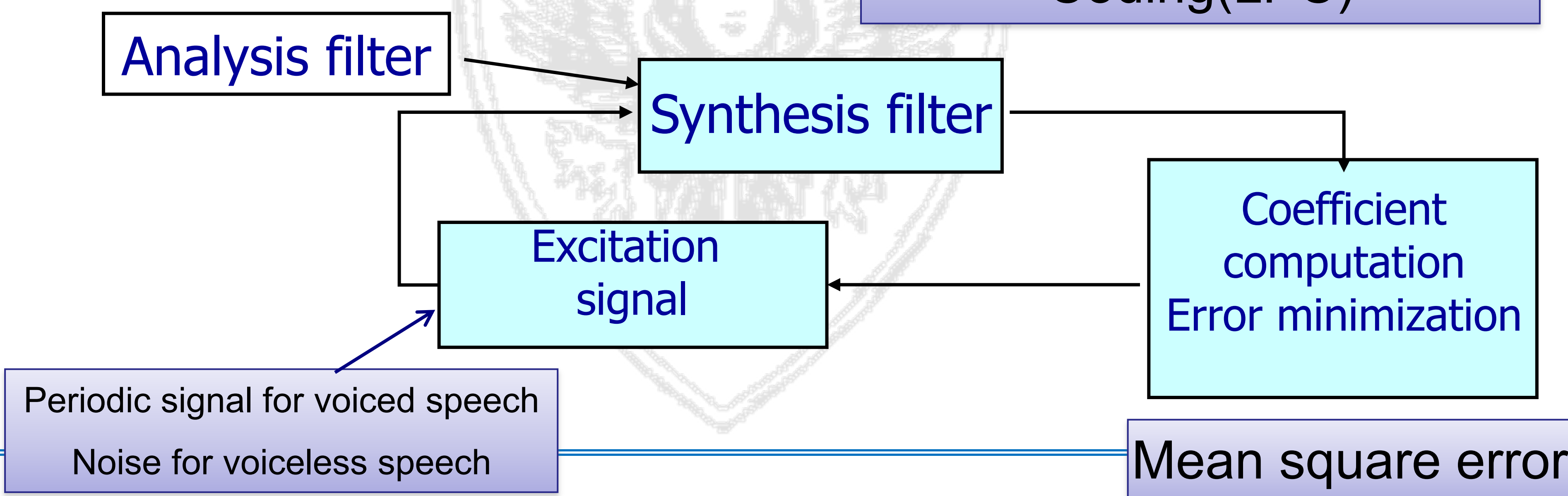
- They are based on models for the generation of the human voice
- models allow us to "remove redundancy" from the vocal segments to obtain an information base sufficient to reproduce the original voice signal (Idea: If we know the structure of the signal little information features will be enough to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

Codebook Excited Linear prediction

LTP= Long term prediction

Hybrid

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

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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3.5 – Procedures

Cellular systems & GSM

Wireless Systems, a.a. 2014/2015

Procedures

☑ O. Bertazioli, L. Favalli, *GSM-GPRS*, Hoepli Informatica 2002

Capitolo 11

GSM procedures

- Network Access
- Mobility
- Call Set Up
- Handover
- Paging

IMSI attach and Location Update

IMSI attach

- When a MS is switched ON:
  - **Cell selection**: the MS selects the BTS to which tune to
  - **Registration** (Location Update procedure): the MS notifies the MSC of its presence in the Location Area

Cell Selection

- The MS scans all RF carriers operating in the cell:
  - Scans c0 carrier over which the BCCH is transmitted
  - Such carriers are transmitted ad higher power than other carriers (dummy bursts are used when necessary), and frequency hopping is disabled
- The MS connects to the RF carrier from which the strongest signal is received
- Through the FCCH channel the MS synchronizes to the BTS carrier
- Through the SCH the MS synchronizes to the slot and frame and receives the BSIC – Base Station Identity Code
- The MS can now decode the BCCH, which includes
  - ✓LAC (Location Area Code)
  - ✓CGI (Cell Glocal Identity)
  - ✓MCC (Mobile Country Code)
  - ✓MNC (Mobile Network Code)

Registration

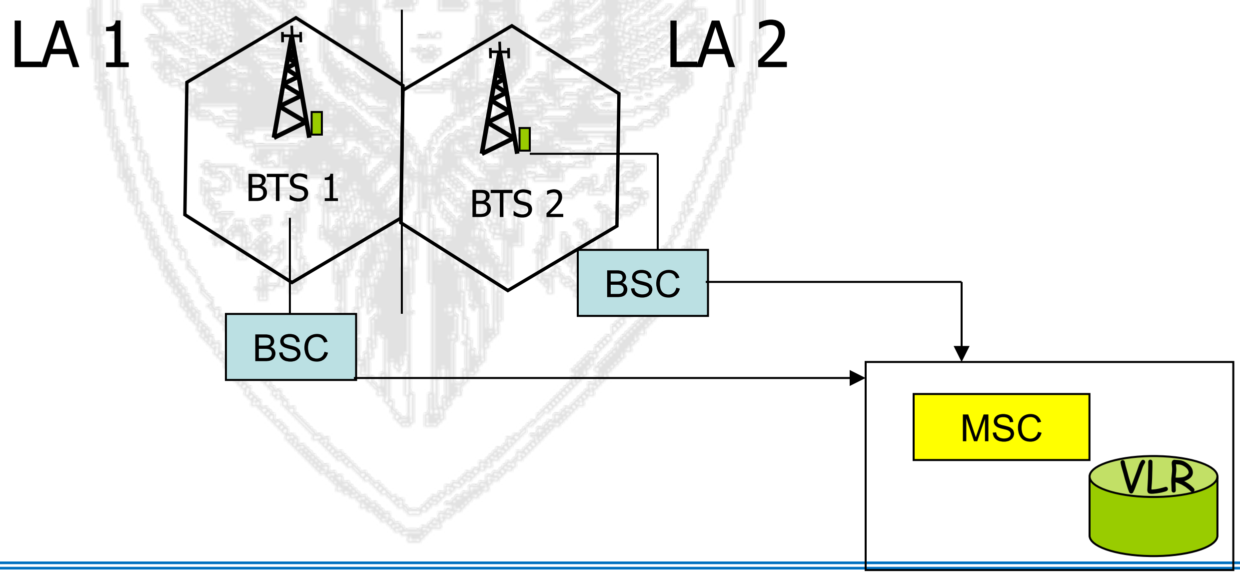
Two cases are possible, based on the **received LAI**:

1)**<u>It is the same of that stored in the SIM</u>** (which happens when the phone is turned off and on in the same LA). The ***IMSI attach*** procedure is invoked, with which the MS activates its IMSI stored in the current VLR (it means the MS was previously registered with the VLR, and that the detached flag was set when the MS was switched off – paging is not performed towards detached users)

1)**<u>No LAI stored, o received LAI different from the stored one</u>** (which happens when the phone is turned off and on in different LAs). The ***Location Update*** procedure is invoked.

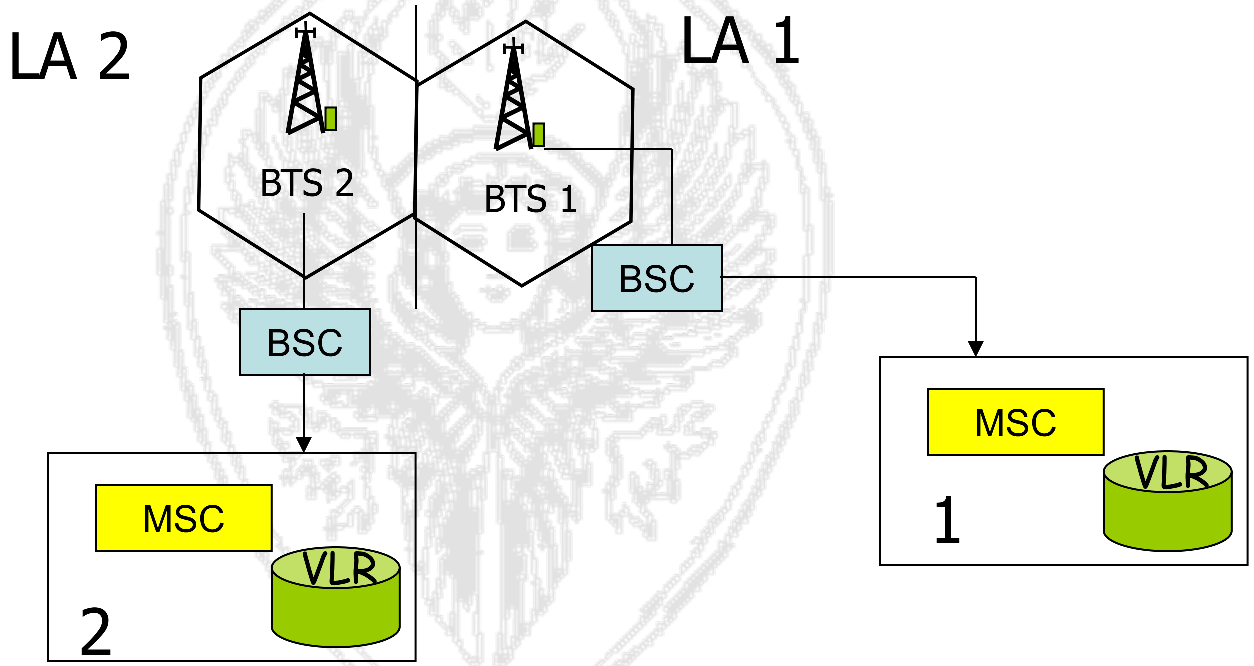
Location Update (1)

- When is it performed?
  - When a MS is switched on (if needed);
  - Periodically (e.g. every 30 min). If the periodic location update is not received, the VLR flags the user as detached -- *implicit detach*;
  - When the Location Area changes due to MS movements (roaming);
- Two types of Location Update:
  - Two LAs of the same MSC/VLR (the simplest case)



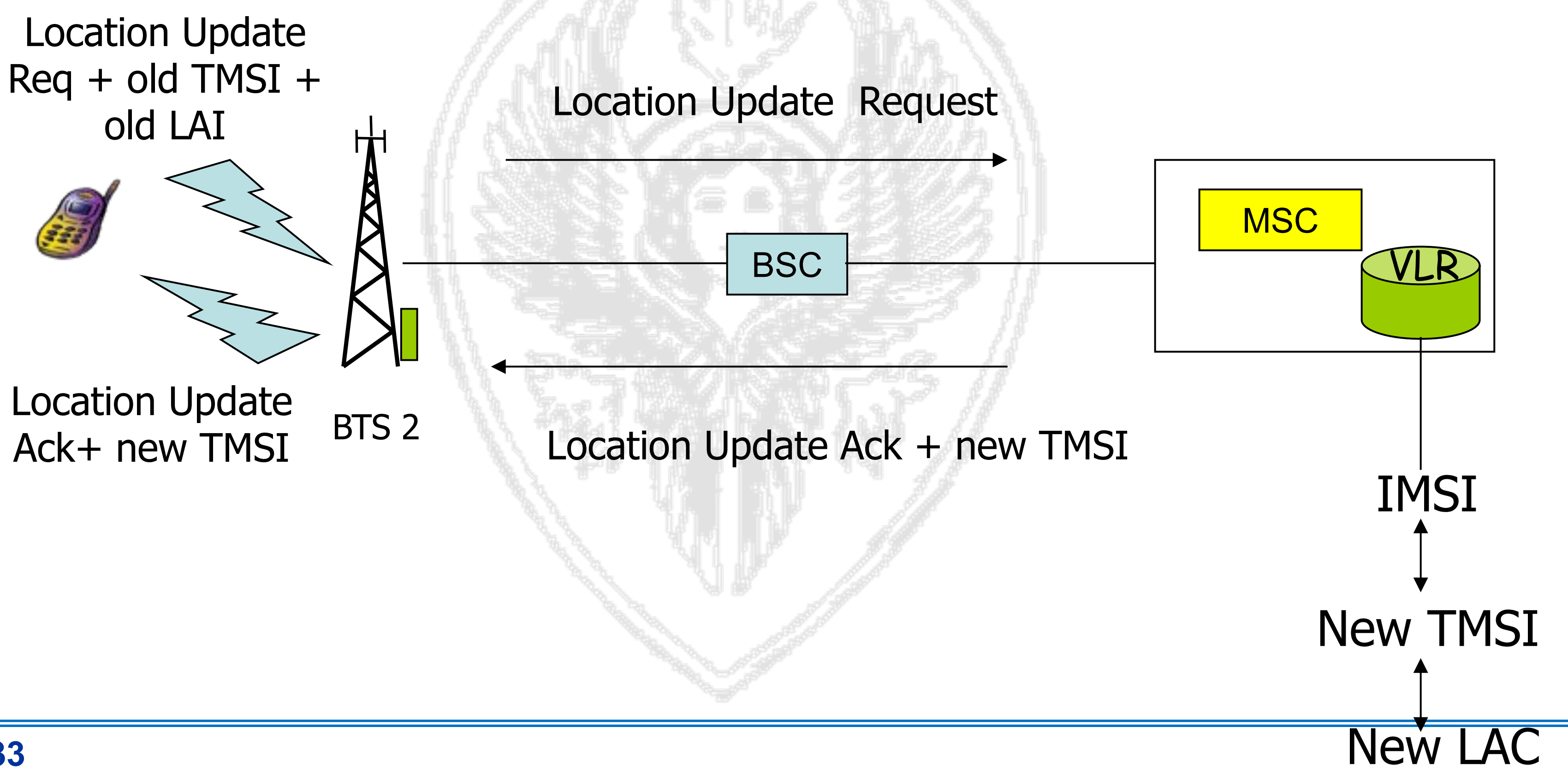
Location Update (2)

- Roaming between LAs of different MSC/VLRs



Location Update - Intra MSC

The System Information Message sent over the BCCH contains the location area identifier (LAI). Once tuned to a new BTS, the MS thus can determine if a location update is needed.



Location Update - Intra MSC

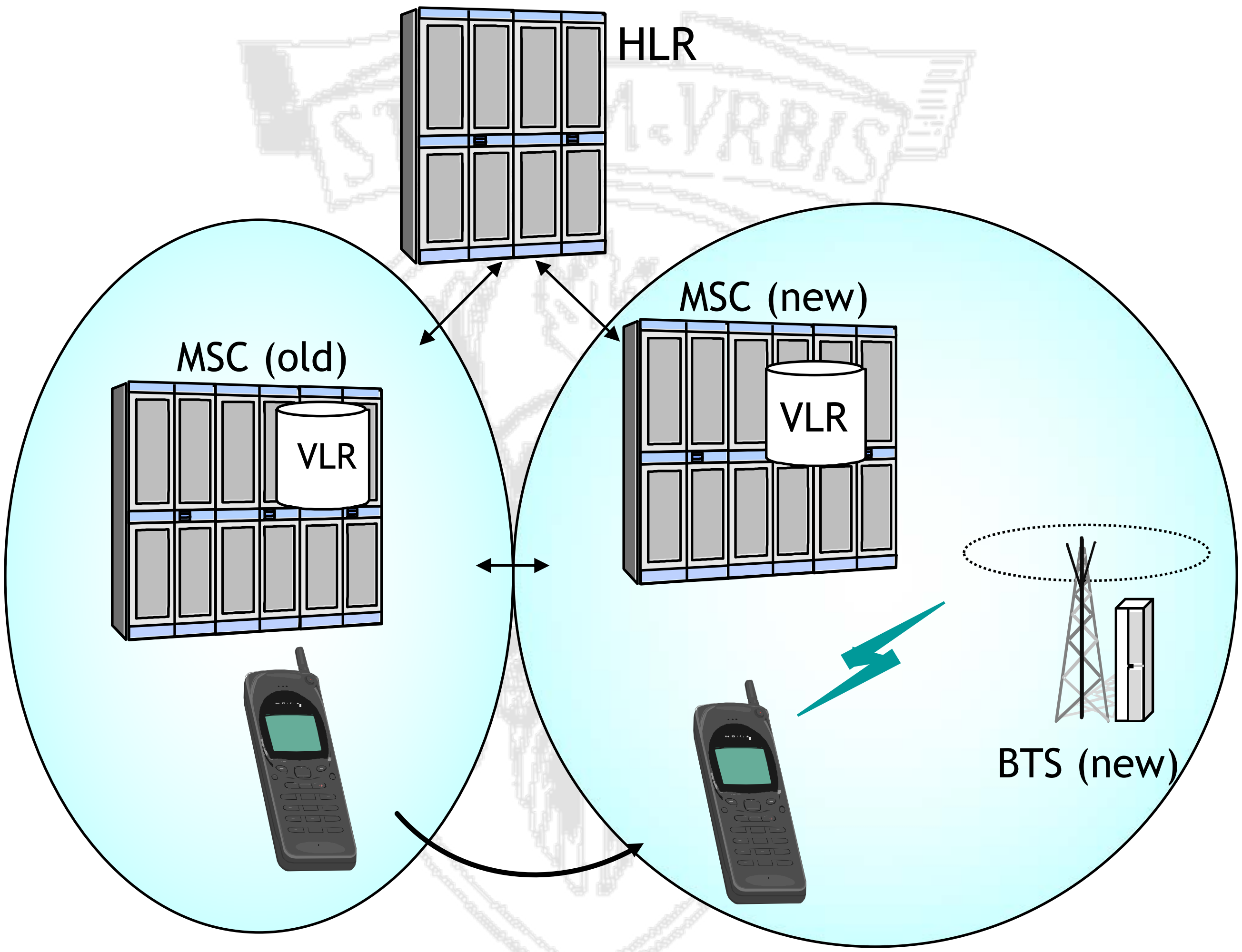
MS

Messages	Channels
CHAN REQ	RACH
IMMEDIATE ASSIGNMENT	AGCH
LOC UPD REQ	SDCCH
AUTH REQ	SDCCH
AUTH RES	SDCCH
CIPH MOD CMD	SDCCH
CIPH MOD COM	SDCCH
LOC UPD ACK + TMSI new	SDCCH
TMSI REALL CMD	SDCCH
CHAN REL	SDCCH

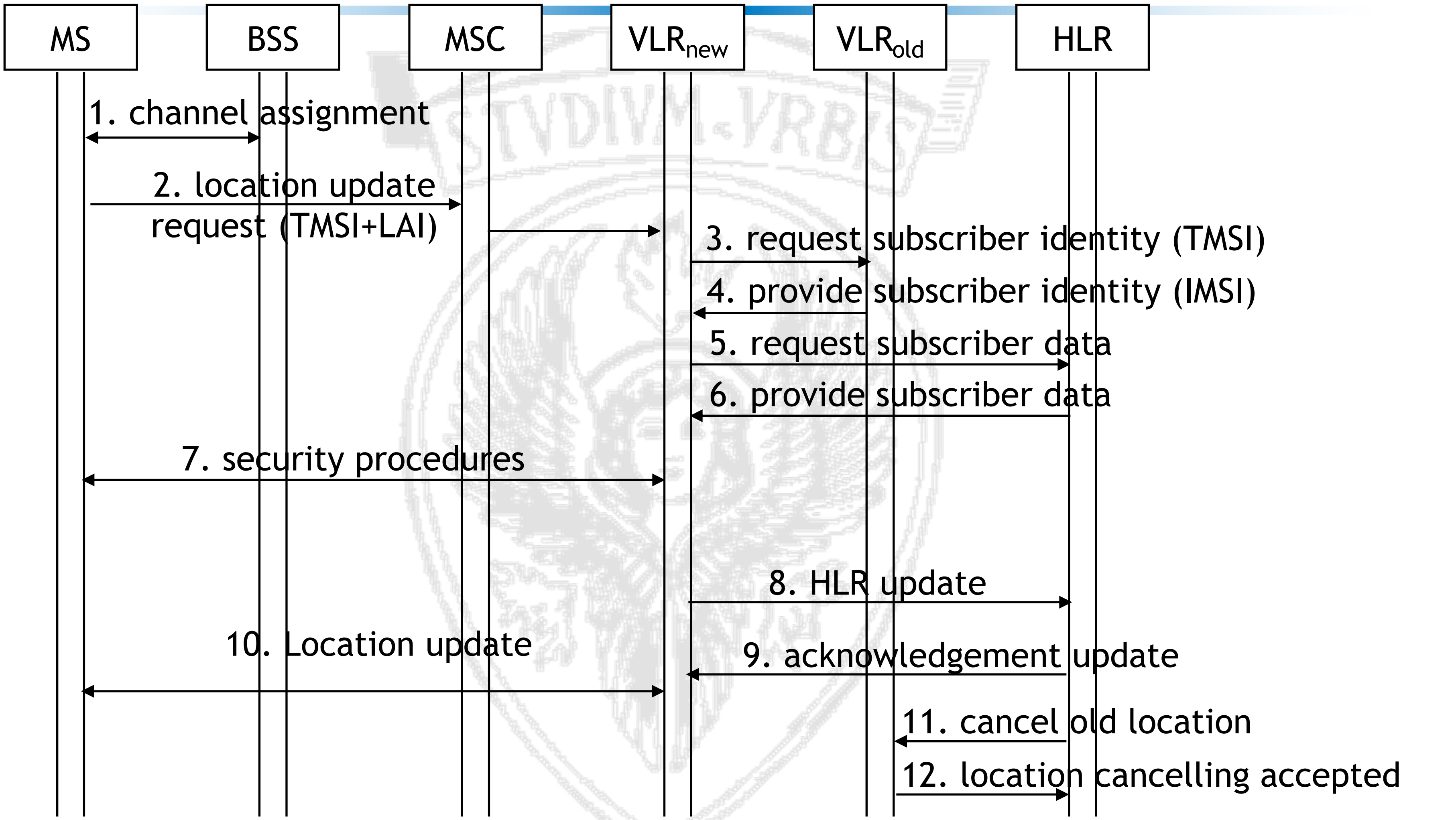
Network

TMSI Reallocation Complete

Location Update inter MSC

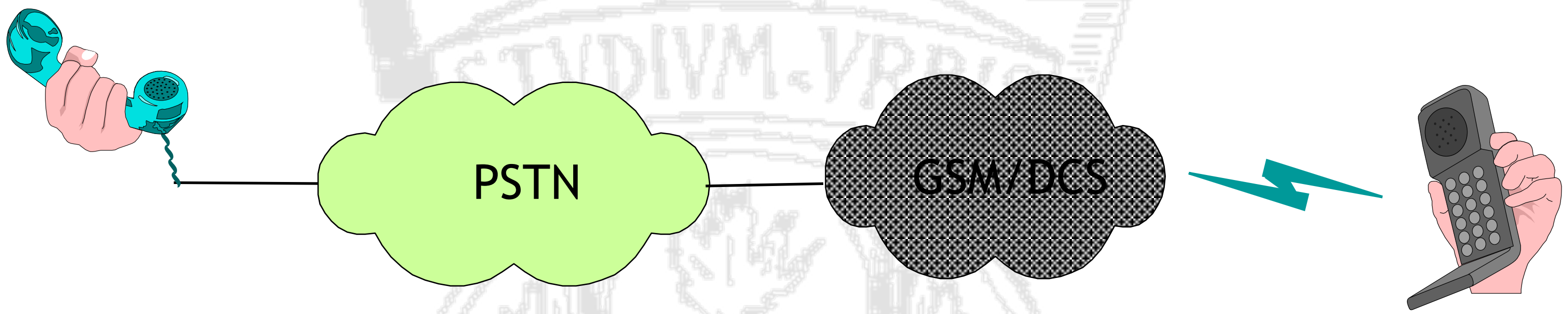


Location Update inter MSC



Call Set Up

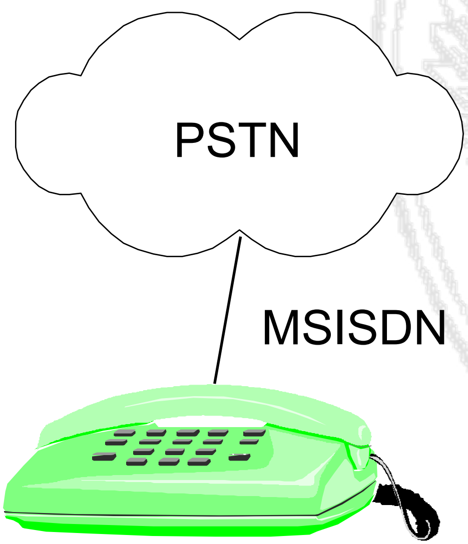
Call originated from PSTN



- Setting up a call terminated on a mobile user is more involved that setting up calls between PSTN users

Call Set-up Step by Step (1)

A The PSTN/ISDN user dials the Mobile Subscriber International ISDN Number (MSISDN) of the user she wants to call



MSISDN: +39 347 6527268

39 = Country Code (Italy)

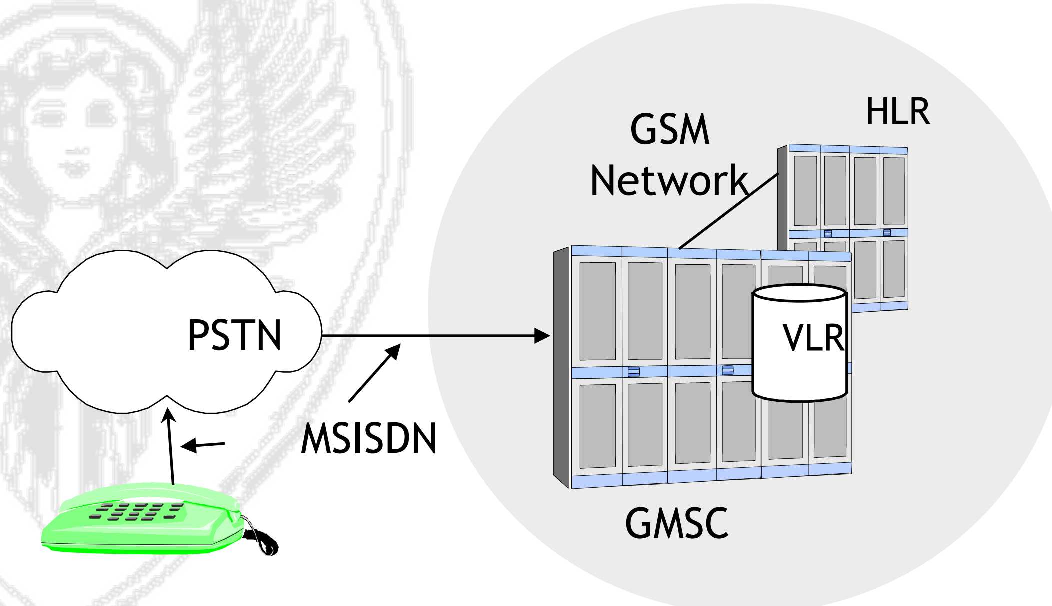
347 = National Destination code

6527268 = Subscriber Number

Call Set-up Step by Step (2)

B The dialled number is analysed by the PSTN/ISDN network, which routes the call to the GMSC of the PLMN of the called user by making use of the National Destination Code (NDC)

C The GMSC receives the message requesting to set-up a call through the SS7 network, which contains the MSISDN of the user called

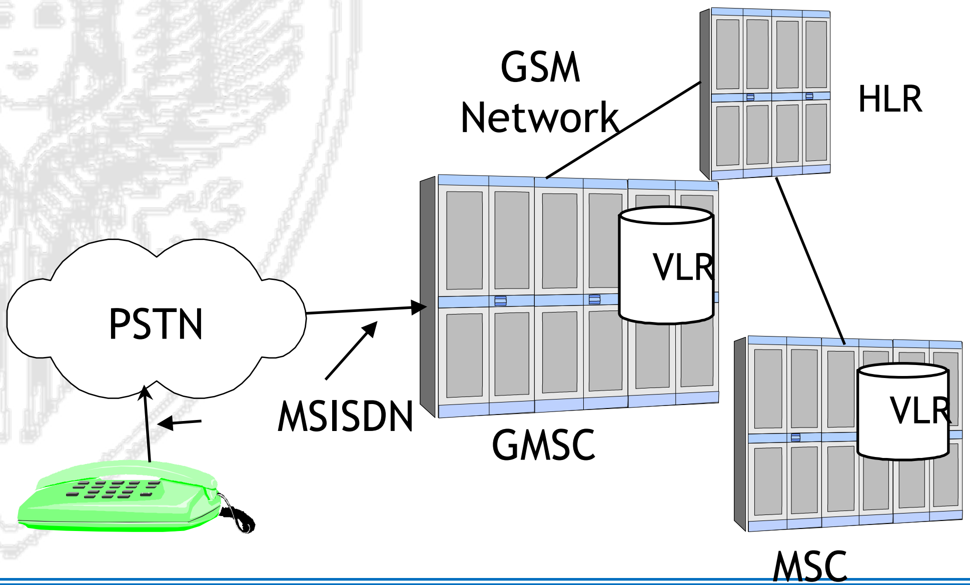


Call Set-up Step by Step (3)

D The GMSC identifies the HLR containing the data of the called user (it is not aware of the position of the MS!!)

E The GMSC sends a message requiring to “send routing information” to the HLR

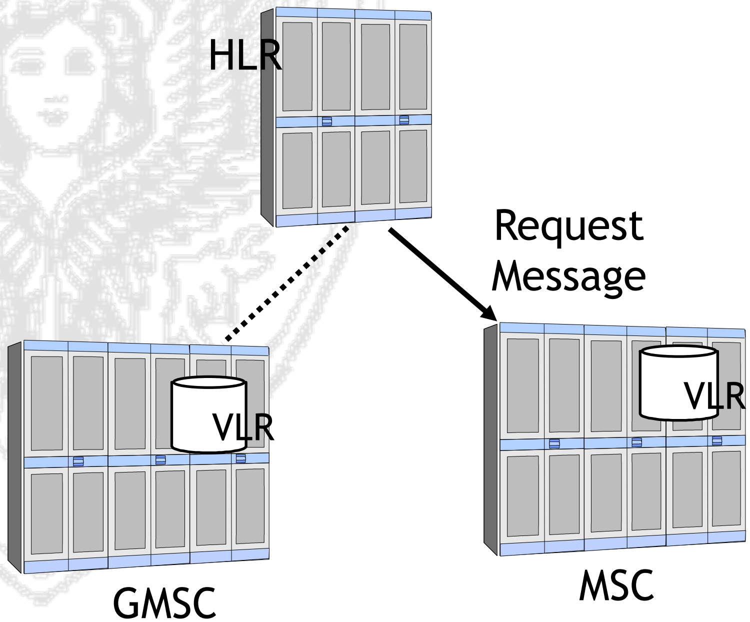
F The HLR identifies the address of the VLR in which the called MS is currently registered



Call Set-up Step by Step (4)

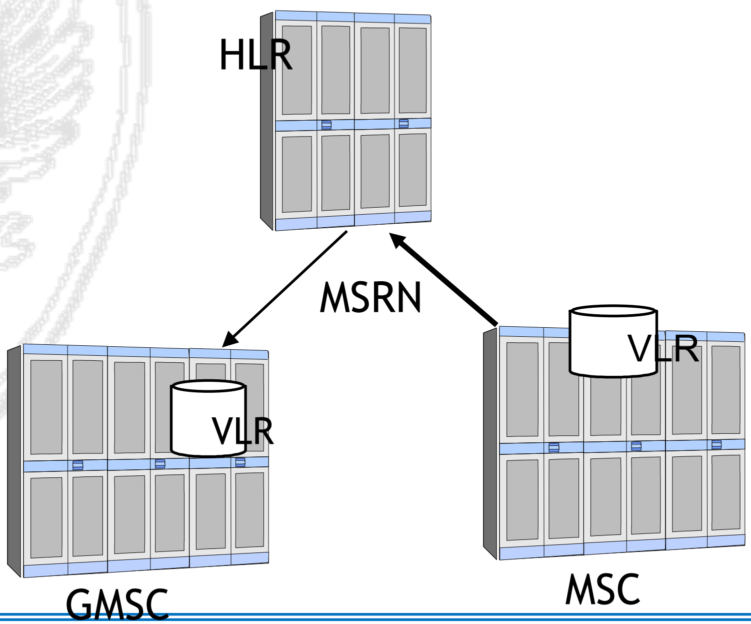
G The HLR sends a "provide roaming number" message to the MSC/VLR

H The MSC/VLR temporarily allocates a Mobile Station Roaming Number (MSRN) to be used for the call



Call Set-up Step by Step (5)

- I The MSRN is forwarded by the MSC to the HLR
- J The GMSC routes the call towards the MSC/VLR of the LA in which the MS is currently located



Call Set-up Step by Step (6)

K The MSC/VLR activates the **paging** procedure:
  – It identifies the currently-visited LA thanks to the IMSI
  – It sends a paging command to all BSC of the location area

L BSC requires the BTSs to send the paging message destined to the MS over the paging channel (PCH) -- this message contains the TMSI assigned to the MS

M The MS replies to the paging message by requiring a Stand alone Dedicated Control CHannel (SDCCH) through the Random Access CHannel (RACH)

Call Set-up Step by Step (7)

N The MSC/VLR activates the authentication and the ciphering procedures

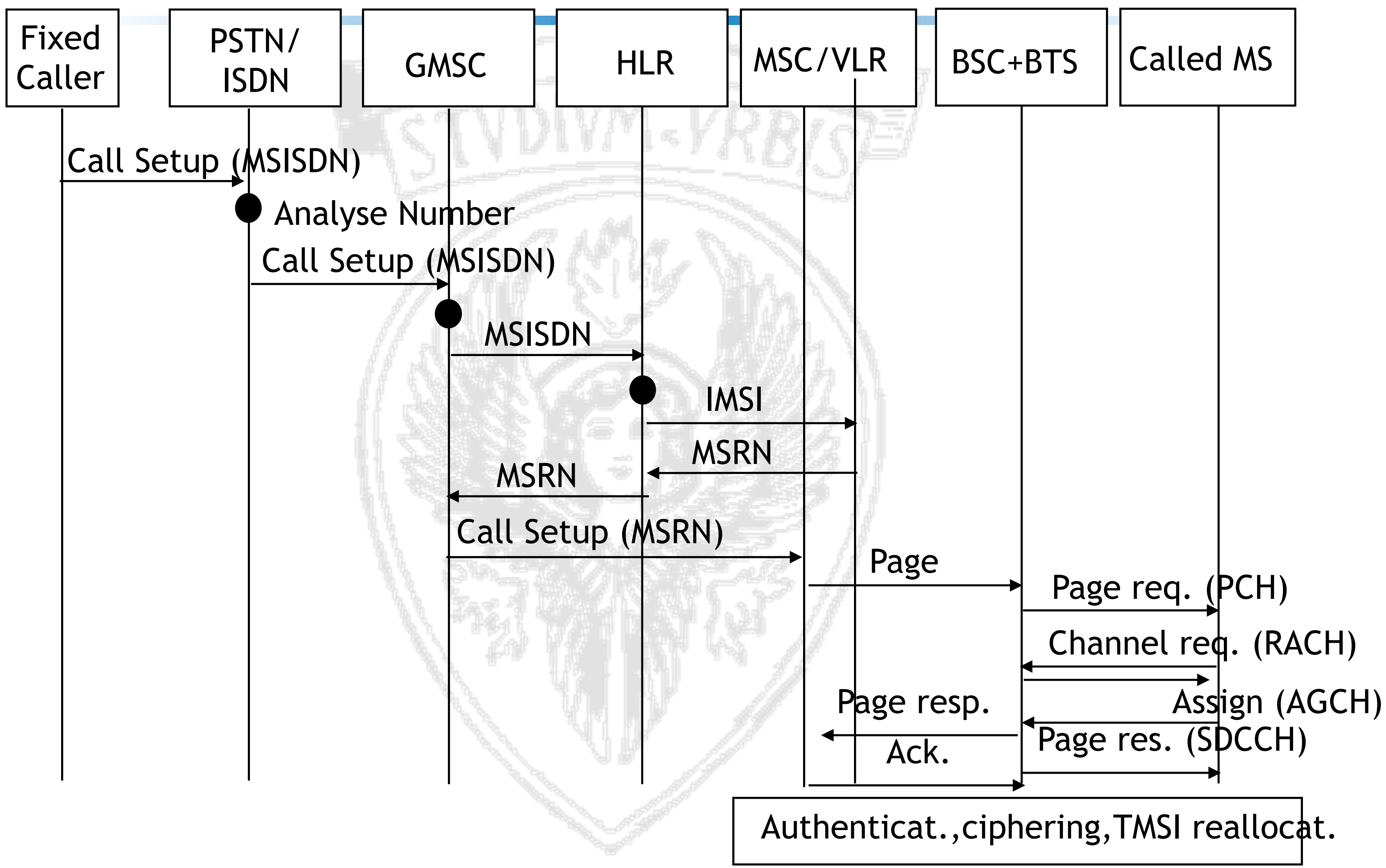
P A traffic channel (TCH) is allocated for the communication

Q The MSC/VLR notifies the caller that the called phone is ringing

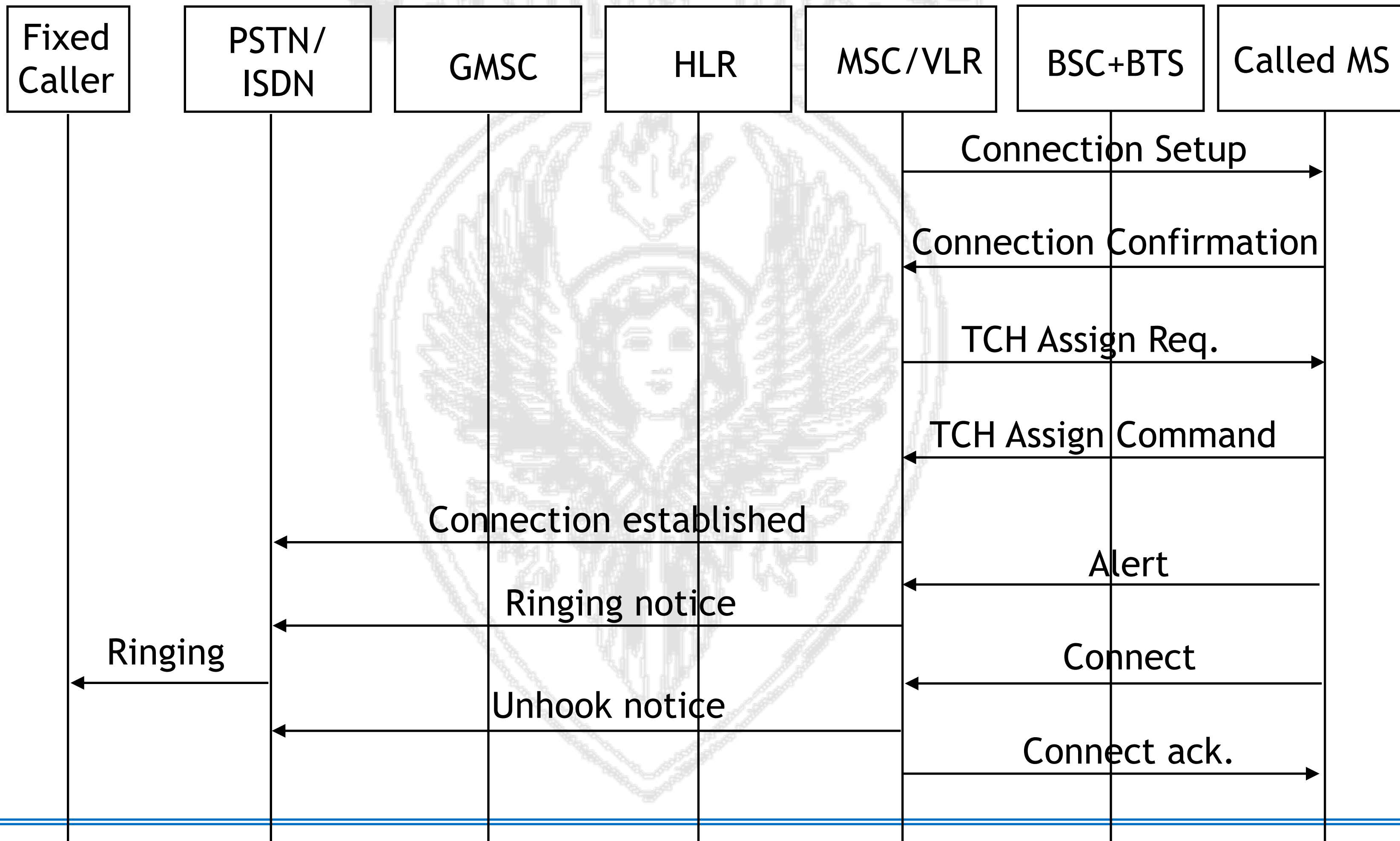
R The called user answers the call

S The connection between the two users is established

Summary of the Call Set-up Steps (1)



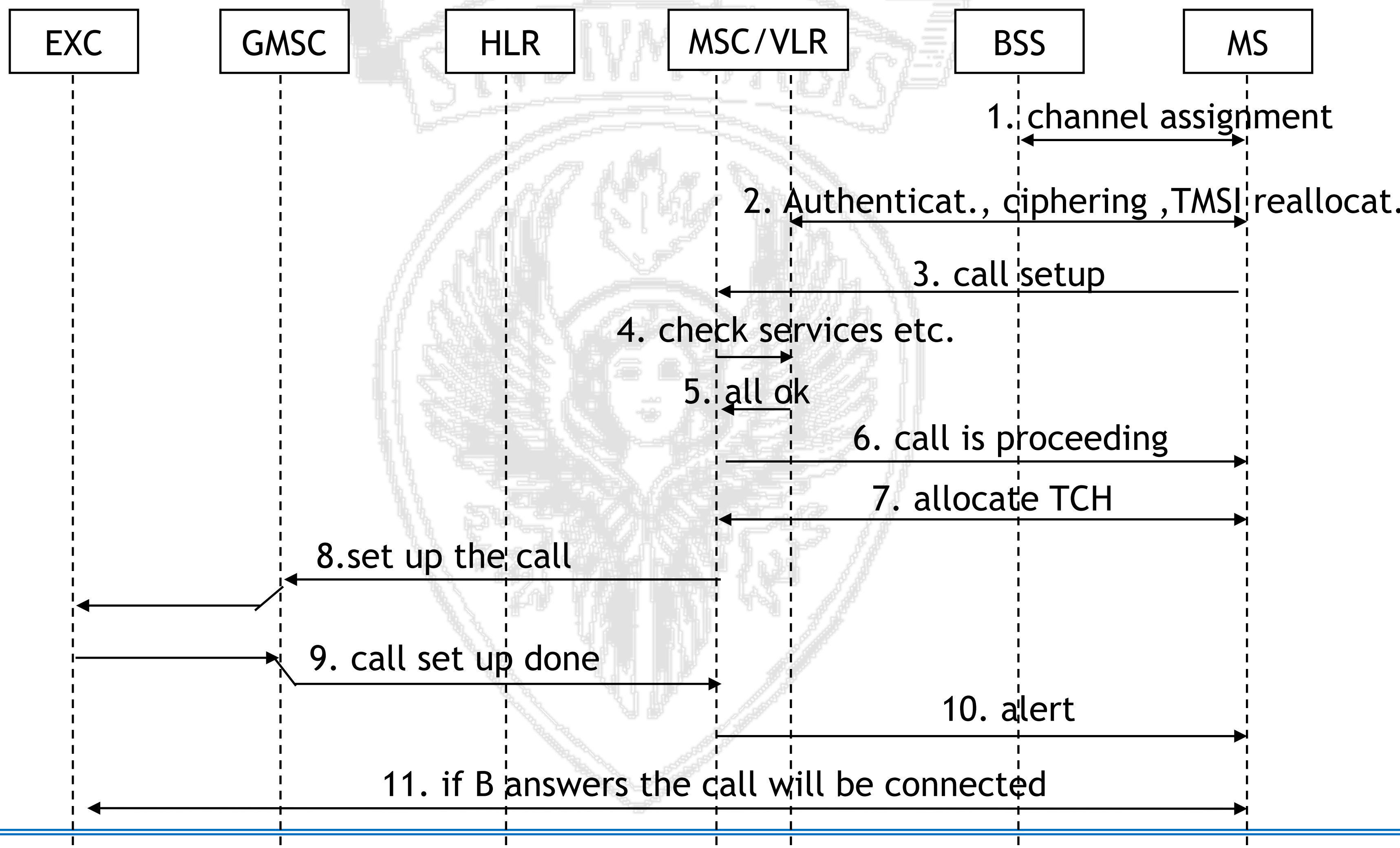
Summary of the Call Set-up Steps (2)



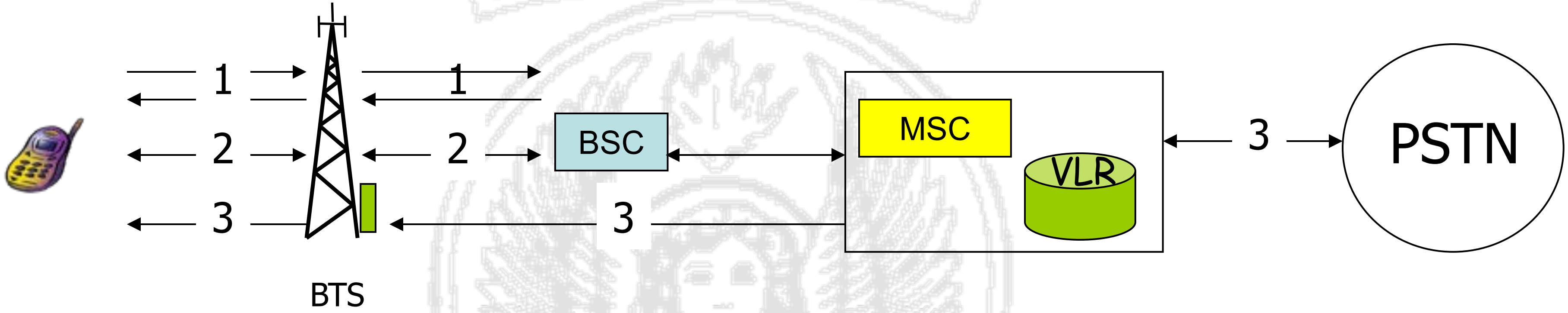
MS-originated call

- The called number is dialled by the MS
- The current MSC analyses the caller data and:
  - It either authorizes or deny the call
  - The call routing procedure is started
- If the called number is in the same GSM network, a “send routing info” procedure is started to obtain the MSRN
  - Same procedure as PSTN-originated calls
- If the called number is in another GSM network, the call is routed to the GMSC.

Summary of the Call Set-up Steps



Mobile-originated call (1)



1- Access request, resource allocation for signaling

2 – Authentication and ciphering, caller id is transmitted, traffic channel is allocated

3 –Call routing

Mobile-originated calls (2)

Canali		Procedura di rete
RACH	CHAN REQ → ← IMM ASSIGN	SDCCH assignment
SDCCH	SABM → ← UA	Service Request
SDCCH	→ ←	Authentication Ciphering
SDCCH	SET UP → ← ASSIGNMENT CONNECT ACK →	TCH/SACCHAssignment
	TCH	
SACCH	DISCONNECT → ← RELEASE RELEASE COMPLTE →	TCH/SACCHRelease

MS

RETE