UCL DEPARTMENT OF ELECTRONIC AND ELECRICAL ENGINEERING
INTEGRATED GRADUATE DEVELOPMENT PROGRAMME
MSC TELECOMMUNICATIONS

Coding Techniques for Wireless Systems

Only a brief introduction!

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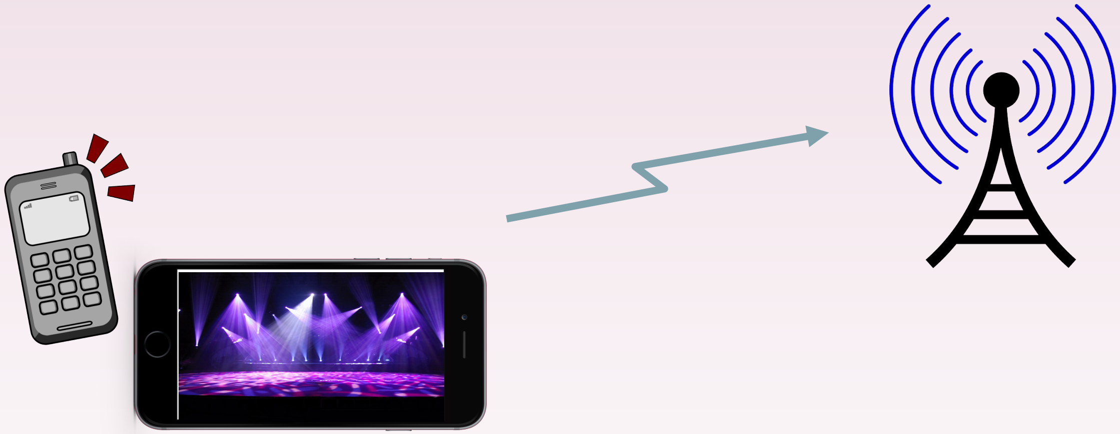
Why do we need coding?



- Speech and video needs to be compressed (limited network resources for large volume of data)
- Compressed bitstream needs to be protected against errors (wireless channel might introduce error in the transmitted bitstream)

Requirements for Speech/Video Transcoding

- Speech: many options available to convert analogue signals into a digital representation
- For a wireless channel there are extra considerations
- Minimisation of the channel bandwidth
- Good quality under ‘good’ radio conditions

How can we limit the bandwidth and maintain quality?

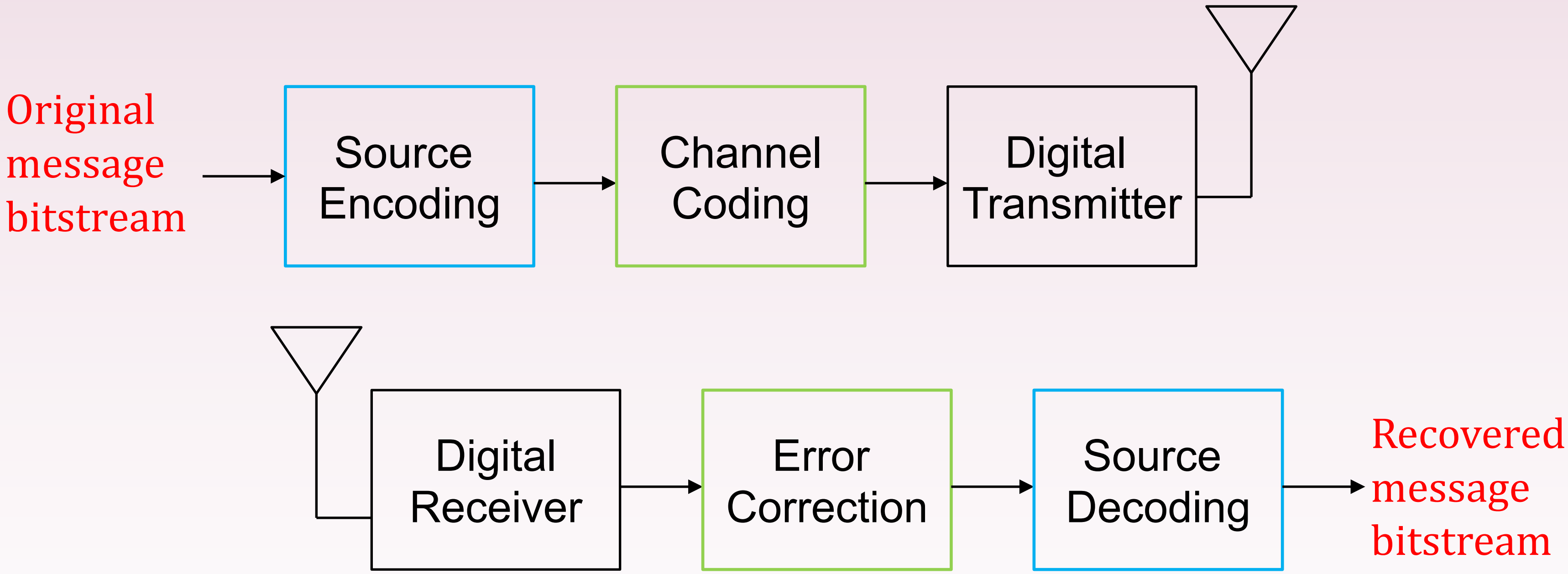
- To limit the bandwidth we can:
  - Use the redundancy inherent in human speech to compress the data
  - Recognise that although a full-duplex type of link is required, each channel will have periods of inactivity

source encoding

- To maintain quality we can:
  - Include error checking and error correction

channel coding

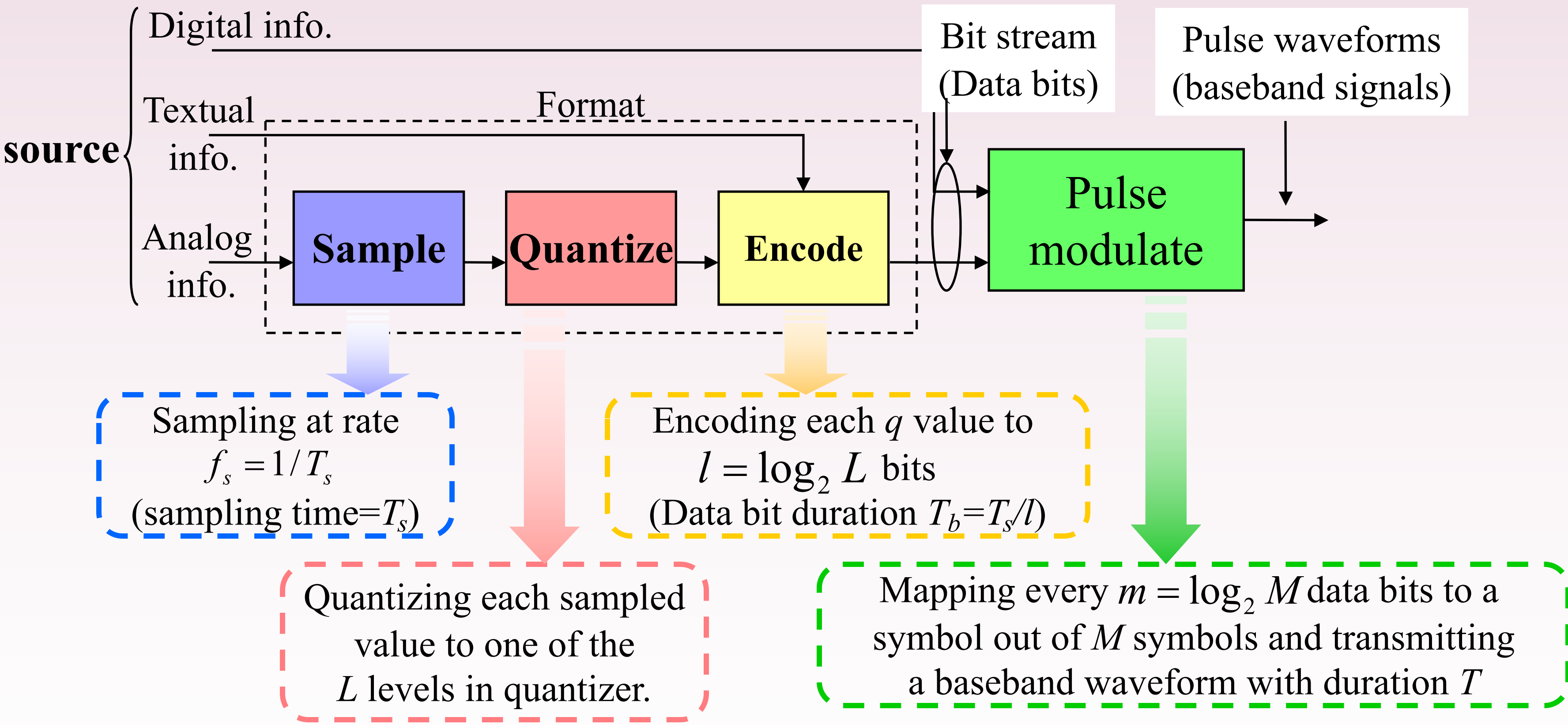
Source Encoding vs Channel Coding



Source encoding: reduce the redundancy → represent the source with fewest bits such that best source recovery from compressed data is possible

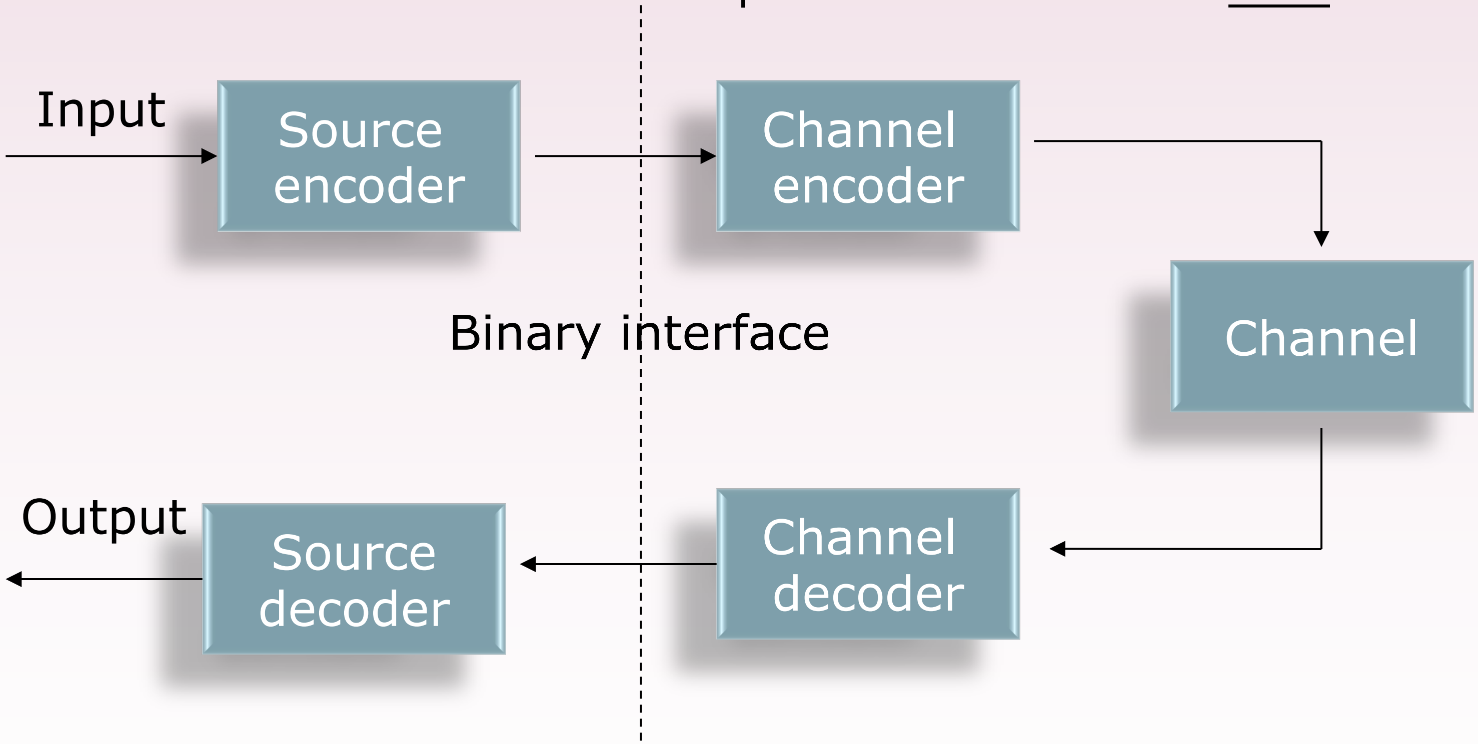
Channel coding: increase the redundancy → represent the compressed message with more bits to increase such that reliability against error

Formatting and transmission of baseband signal



- Information (data) rate: $R_b = 1/T_b$ [bits/sec]
- Symbol rate : $R = 1/T$ [symbols/sec]
  - For real time transmission: $R_b = mR$

- **Basic communication system layering**
  Separation of source and channel coding
  The source encoder needs to understand the probabilistic structure of the source
  The channel encoder needs to understand the probabilistic nature of the noise



Introduction

- **Part 1: Source Coding**
  - Requirement for Speech Coding
  - Digital Speech
  - GSM RPE-LPC arrangement
  - Image/Video Coding

- **Part 2: Channel Coding**
  - Error Control coding requirements
  - Error Control principles and techniques (overview)
  - Application to GSM networks

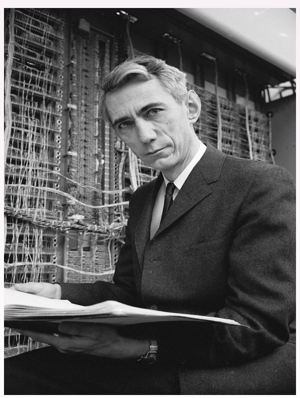
Source Encoding

Source Encoding

Lossless

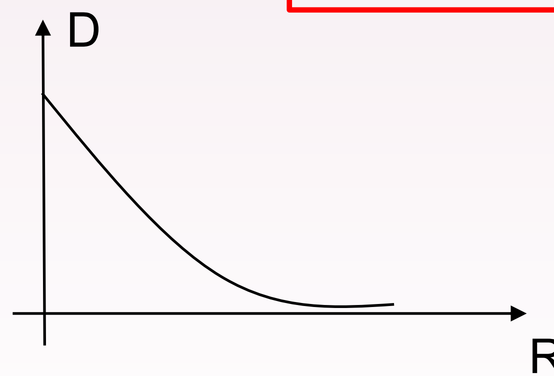
Lossy

Speech and Video coding



Shannon's first theorem (1948)

Coding rate R greater than the source entropy



Source recovered with a distortion D (function of the coding rate)

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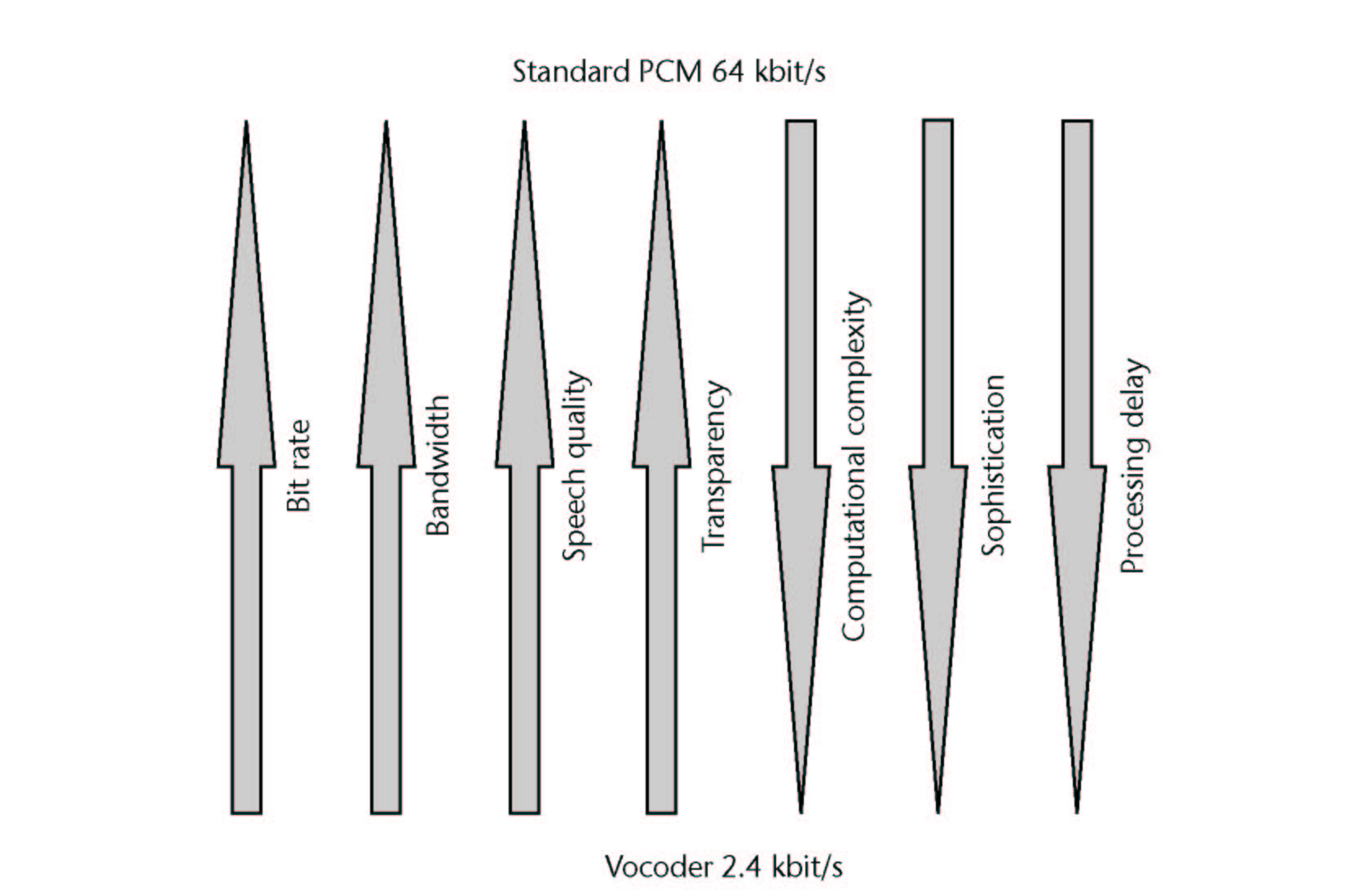
Digital Speech

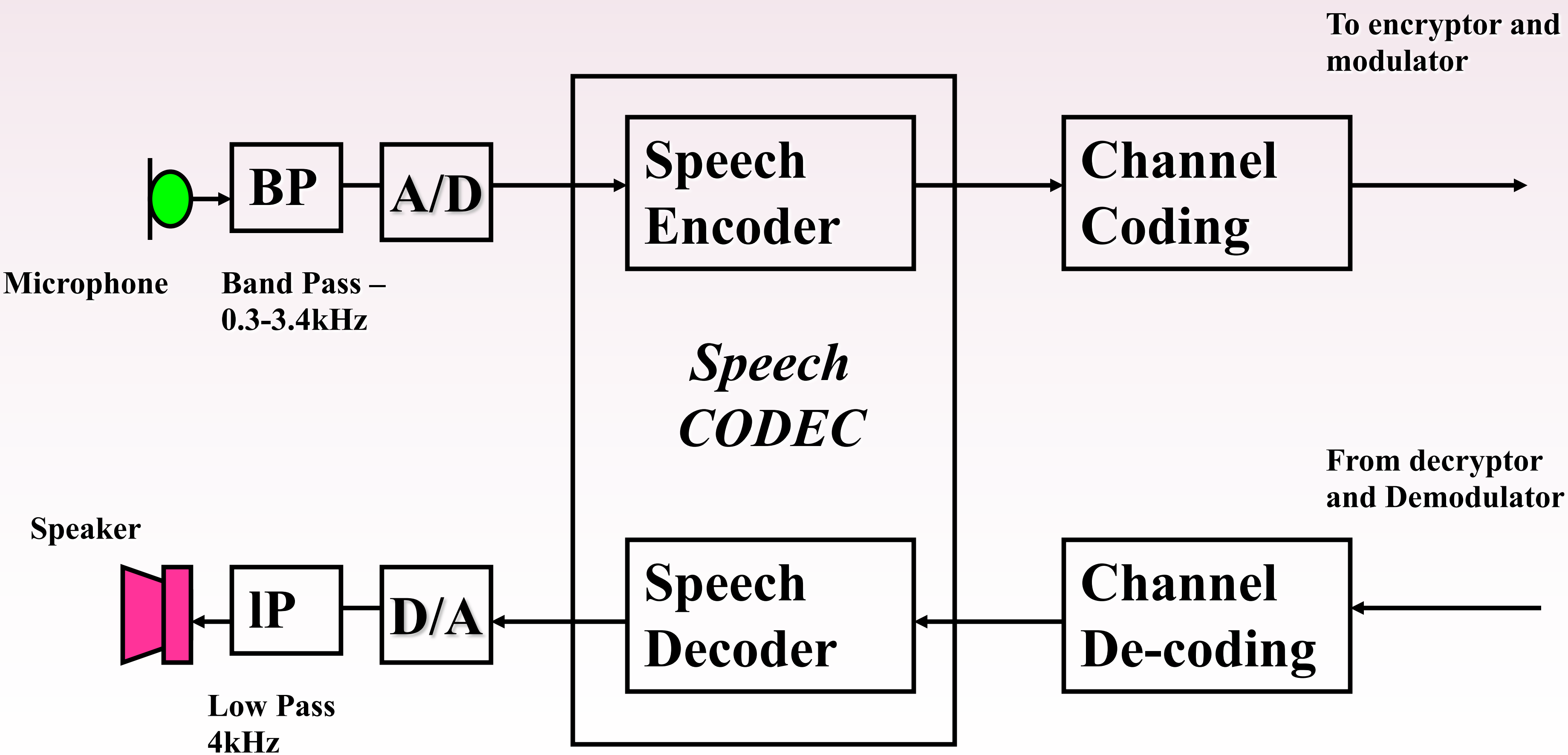
- Pulse Code Modulation (PCM)
  - 64 kb/s companded for telephony (A-law and μ-law)
- Adaptive Differential Pulse Code Modulation (ADPCM)
  - 32 kb/s ITU-T standard for telephony
- Various 'compressed' speech arrangements:
  - Sub-band, pitch predictive ADPCM, Adaptive Predictive Coder, Adaptive Transform Coder, Vocoder (voice excited, channel, formant), Linear Predictive Coefficient
  - The GSM speech coder employs regular pulse excitation (RPE), long term prediction(LTP) and linear prediction coding (LPC) and is commonly referred to a RPE-LPC
- A Transcoder provides for conversion and rate adaptation between GSM speech and digital PCM speech in the main (fixed) network

Trade-offs in speech coding

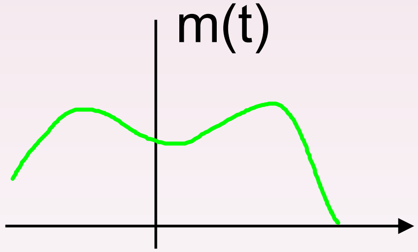


Taken from *Communication Engineering Principles*, © Ifiok Otung, published 2001 by Palgrave

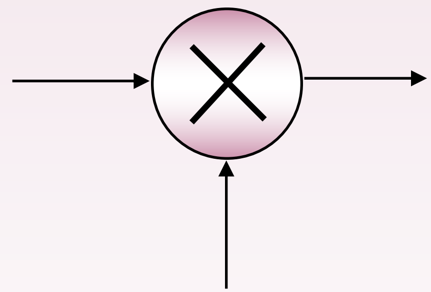
How does the whole process work?



Sampling - the time domain

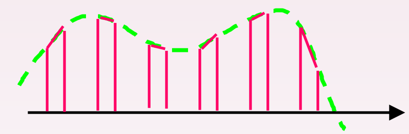


Message m(t) with spectrum M(f); M(f)=0 for |f|>W

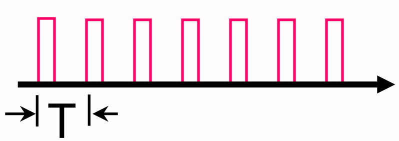


Sampling pulse train

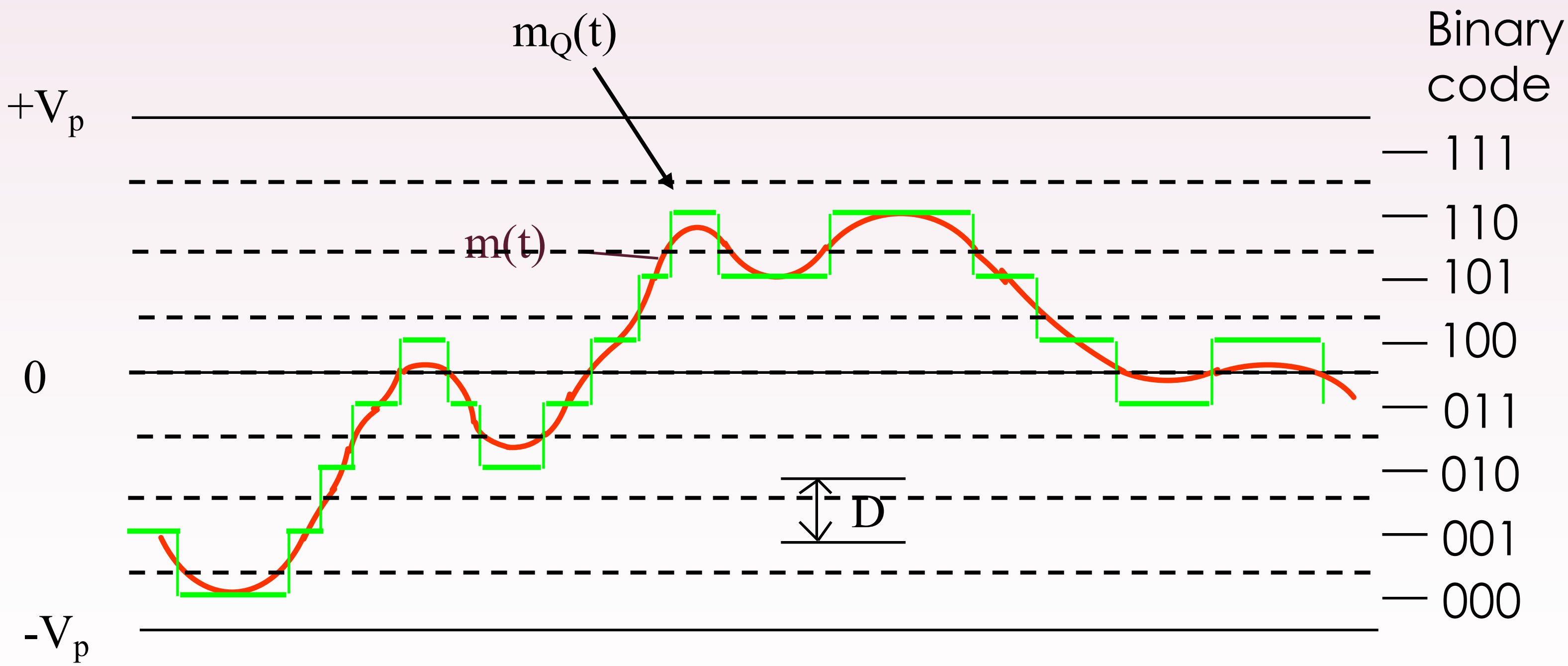
PAM signal



$x_p(t)=\Sigma_k p(t-kT);$
$F_s=1/T$



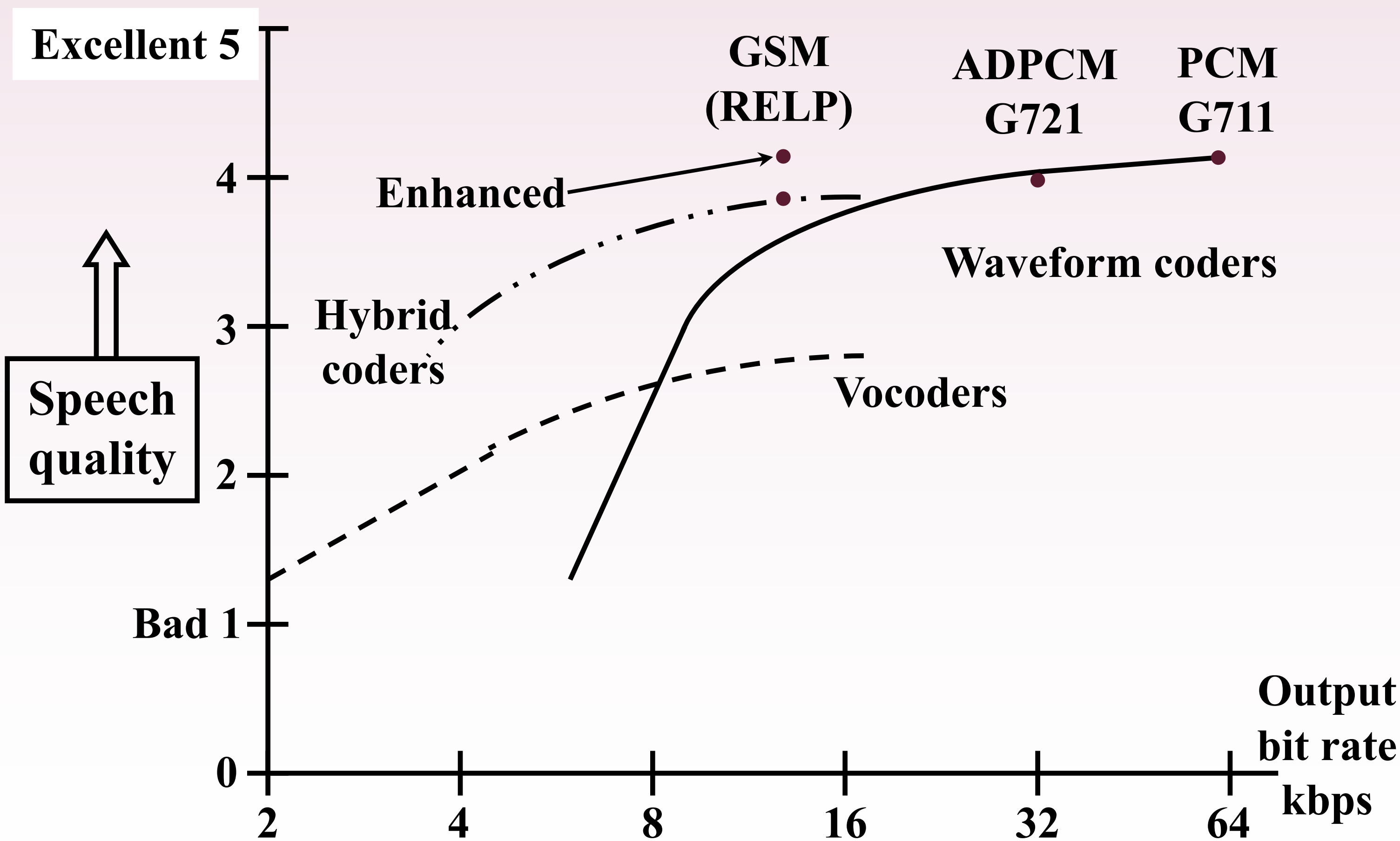
Quantisation and Coding



GSM Speech coding

- GSM is slightly different to normal ‘wired’ PCM
- The PAM signal is quantised by 13 bits $=2^{13}=8,192$ quantisation levels
- This gives a data rate of 8,000 x 13=104 kb/s
- This is too high, the coder must do something to significantly reduce this rate
- The GSM speech coder employs regular pulse excitation (RPE), long term prediction(LTP) and linear prediction coding (LPC) and is commonly referred to a RPE-LPC

Speech Quality versus Bit Rate



Required Bandwidth

- Typically the range of human hearing is from 20Hz up to a maximum of 20kHz
- However, different types of signals have different requirements
- Music

22kHz – 16bit 12kHz – 16bit 5.5kHz – 16bit 4kHz – 16bit

- Telephony

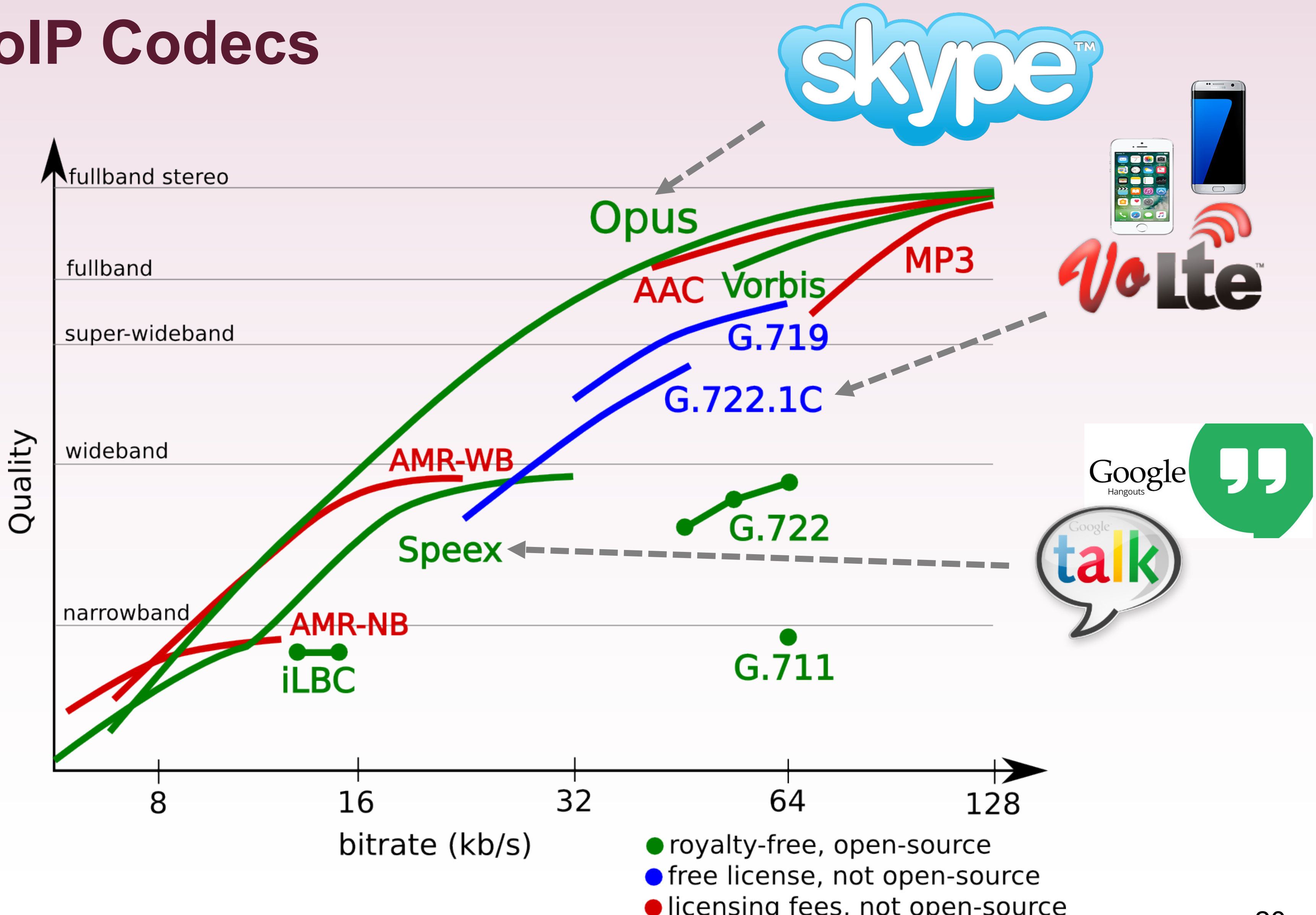
16kHz – 8bit 4kHz – 8bit

VoIP Codecs

Codec	Algorithm	Bit Rate (Kbps)
ITU G.711	PCM (Pulse Code Modulation)	64
ITU G.723	Multi-rate Coder	5.3 and **6.4**
ITU G.726	ADPCM (Adaptive Differential Pulse Code Modulation)	16, 24, 32, and 40
ITU G.727	Variable-Rate ADPCM	16-40
ITU G.728	LD-CELP (Low-Delay Code Excited Linear Prediction)	16
ITU G.729	CS-ACELP (Conjugate Structure Algebraic-Code Excited Linear Prediction)	8 and **6.3**
ILBC	Internet Low Bitrate Codec	13.33 and 15.20
GSM - Full Rate	RPE-LTP (Regular Pulse Excitation Long-Term Prediction)	13
GSM - Enhanced Full Rate	ACELP (Algebraic Code Excited Linear Prediction)	12.2
GSM - Half Rate	CELP-VSELP (Code Excited Linear Prediction - Vector Sum Excited Linear Prediction)	11.4
ITU G.722	SBADPCM (Sub-Band Adaptive Differential Pulse Code Modulation)	48, 56 and 64
Speex	CELP (Code Excited Linear Prediction)	2.15-44.2
AMR-WB (ITU G.722.1)	ACELP (Adaptove Code Excited Linear Prediction)	6.6-23.85

Broadband codecs

VoIP Codecs


fullband stereo
fullband
super-wideband
wideband
narrowband
Quality
Opus
AAC
Vorbis
MP3
G.719
G.722.1C
AMR-WB
Speex
G.722
AMR-NB
iLBC
G.711
8
16
32
64
128
bitrate (kb/s)
royalty-free, open-source
free license, not open-source
licensing fees, not open-source
skype
Google Hangouts
Google talk

Speech Coding

- But if we use this on music which does not follow the same patterns as speech then the result is slightly different.

Music Clip CD quality 44.1kHz sampling

2035kbytes total file

Music Clip PCM 8kHz sampling

210kbytes total file

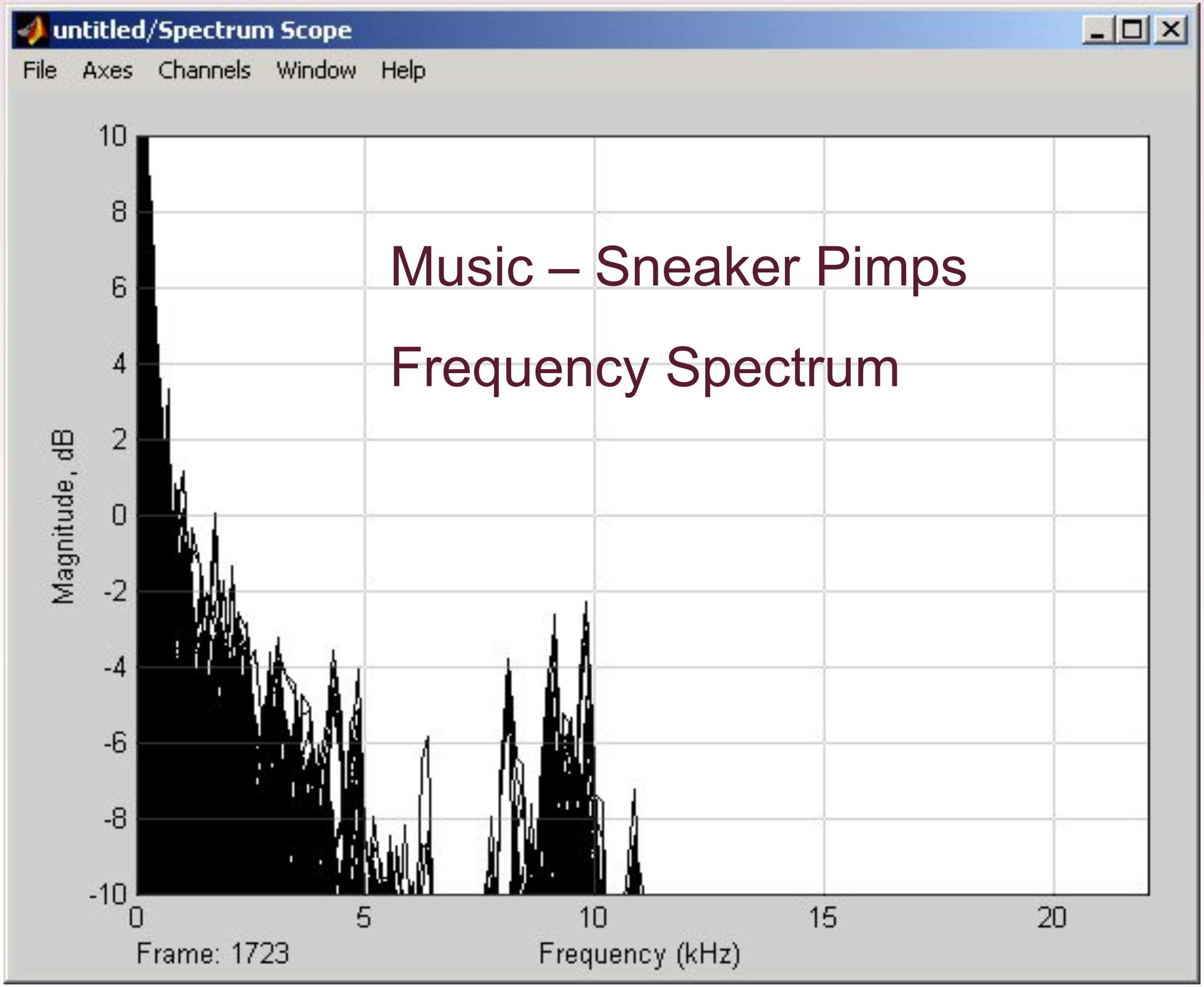
Music Clip ADPCM 8kHz sampling

56kbytes total file

Music Clip GSM 6.1 Encoding 8kHz sampling

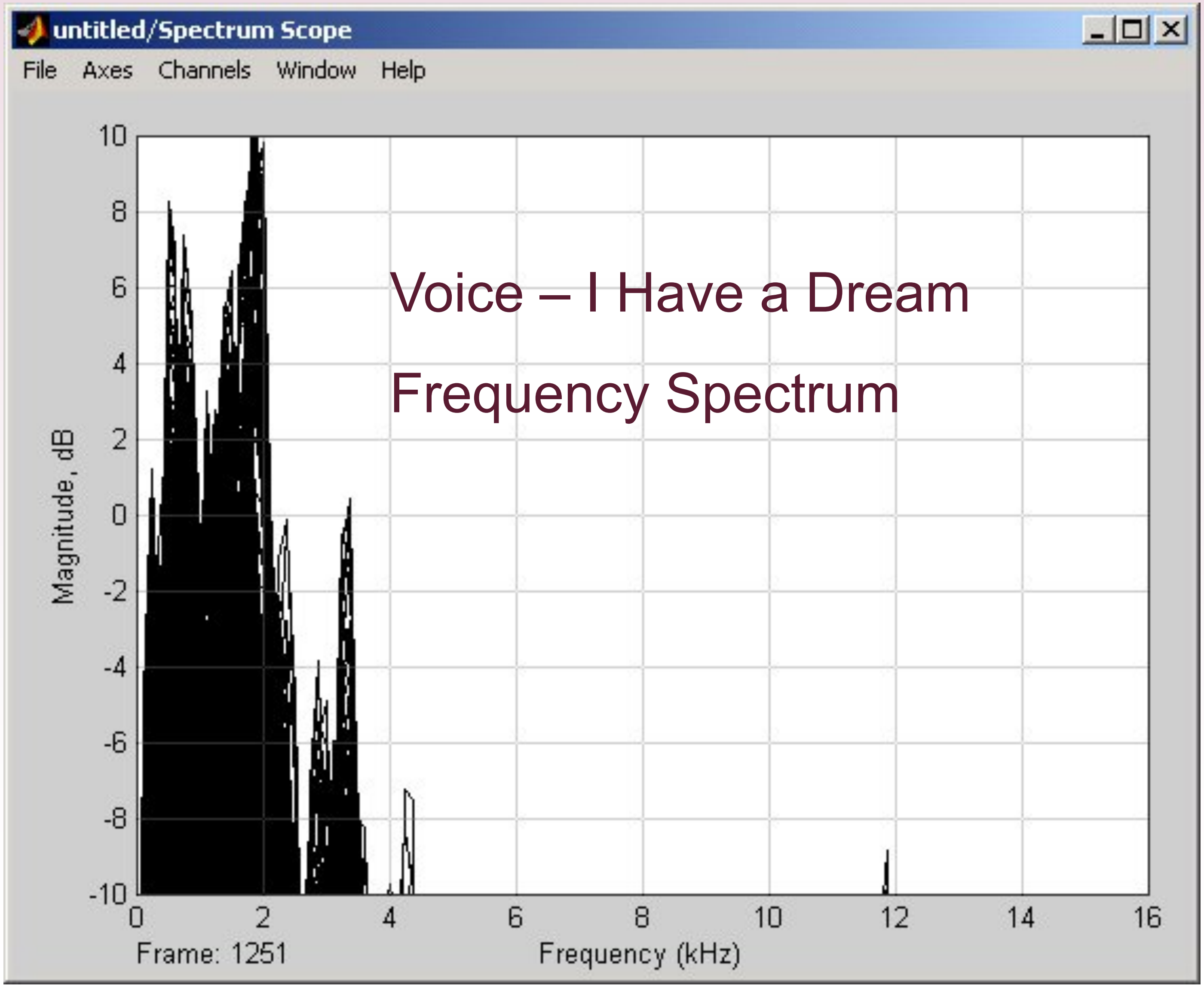
1kb/s - 22kbytes total file

Why?

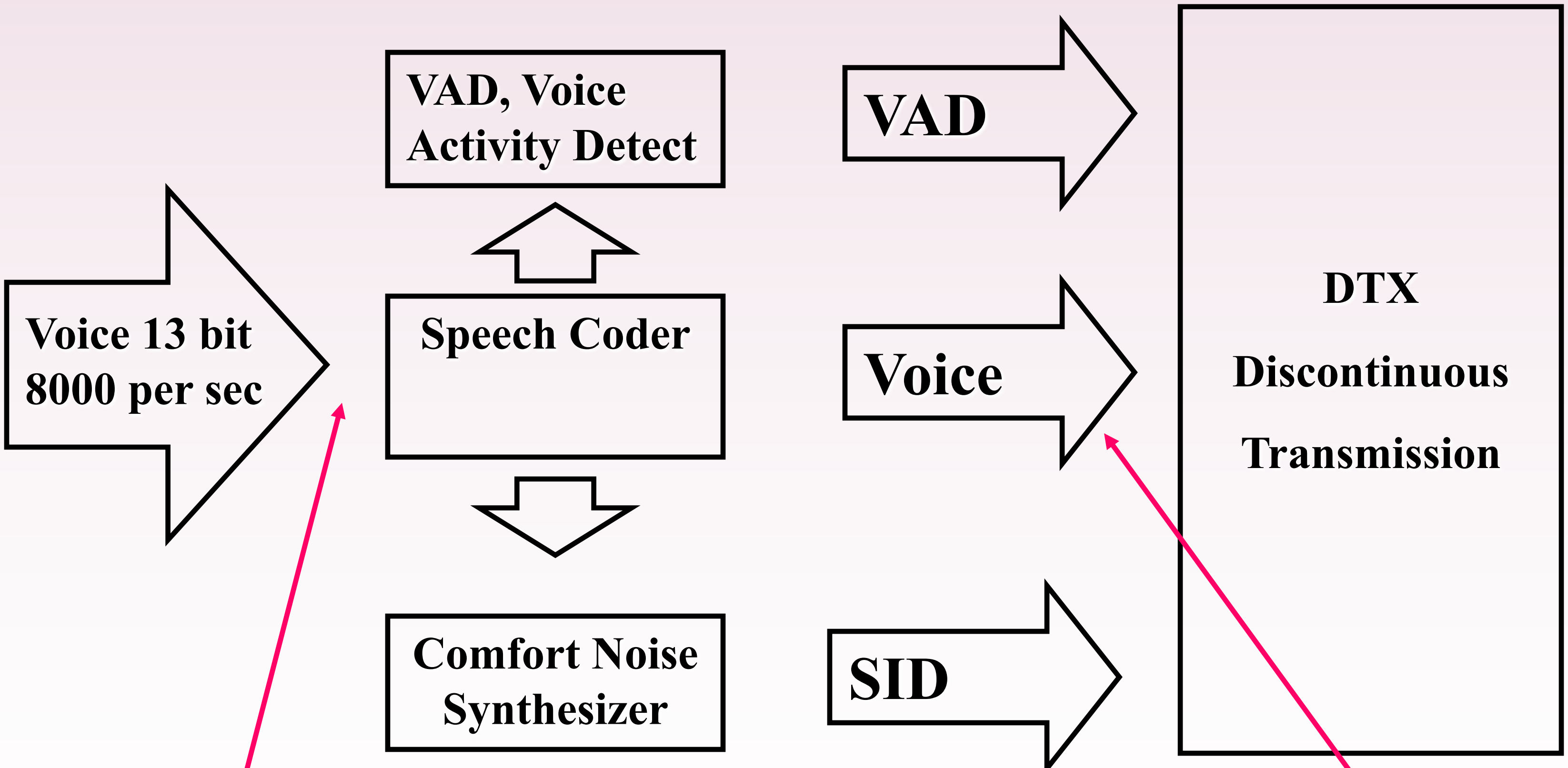


untitled/Spectrum Scope
File Axes Channels Window Help
Magnitude, dB
10
8
6
4
2
0
-2
-4
-6
-8
-10
0
5
10
15
20
Frame: 1723
Frequency (kHz)
Music – Sneaker Pimps
Frequency Spectrum

Why?


untitled/Spectrum Scope
File Axes Channels Window Help
Voice – I Have a Dream
Frequency Spectrum
Magnitude, dB
10
8
6
4
2
0
-2
-4
-6
-8
-10
0
2
4
6
8
10
12
14
16
Frame: 1251
Frequency (kHz)

GSM Speech Codec



Blocks of 160 Samples of 13 bits every 20ms

260 bit blocks at rate of 13 kb/s, i.e. 1:8 compression

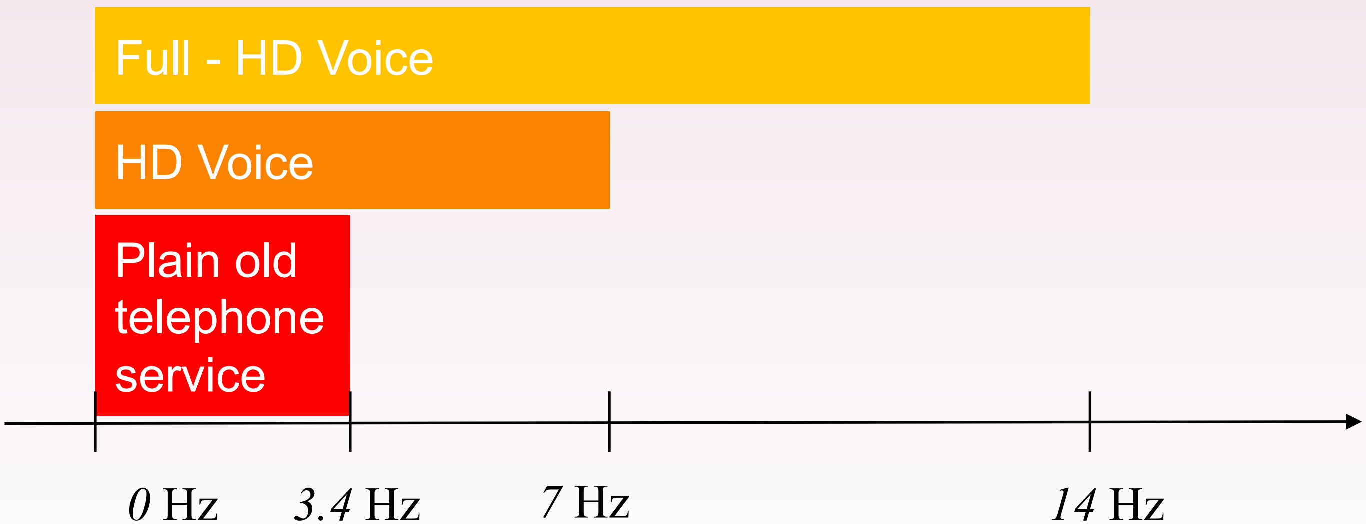
Discontinuous Transmission

- GSM provides for discontinuous transmission - the DXT mode (In the US literature the term variable bit rate is sometimes used)
- It aims to increase system efficiency by decreasing the interference level as a result of inhibiting radio transmission when not required
- DXT is an optional alternative
- It involves some degradation of quality, especially when used twice for a call between two GSM users
- It can be effected by the network on a per-call basis

DTX - cont'd

- The goal is to encode at 13kbit/s when speech is present and otherwise at a much reduced rate of around 500bit/s
- This low rate is sufficient to encode the background (comfort noise), regenerated for the listener to give reassurance that the connection remains
- It corresponds to decreased effective radio transmission since active speech flow is one 260 bit frame every 20 ms and inactive speech flow is one 260 bit frame every 480 ms

What's Next? Full-HD Voice



Human voice transmits audio between 75 Hz and 14 kHz.
Full-HD will cover the entire range of human hearing.

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Image/Video compression

"Since it's digital, there's no generation loss"

WRONG!

– All image/video compressions are lossy
– If generations are made by compress/decompress, then losses can be ugly

Image Compression - JPEG

Compress (avoid redundancy) in the spatial domain

Compression level “low”
Compression ratio 1:16
6 % of original file size
No visible image quality degradation

Compression level “high”
Compression ratio 1:96
1 % of original file size
Image quality clearly degraded

Video Compression

Spatial vs. temporal compression

- spatial → intra-frame, just image compression
- temporal → inter-frame (based on motion compensation)

↓

Two kinds of frames

- key frames: spatial compression only
- difference frames: spatial and temporal (relative to some other frame)

Temporal Compression

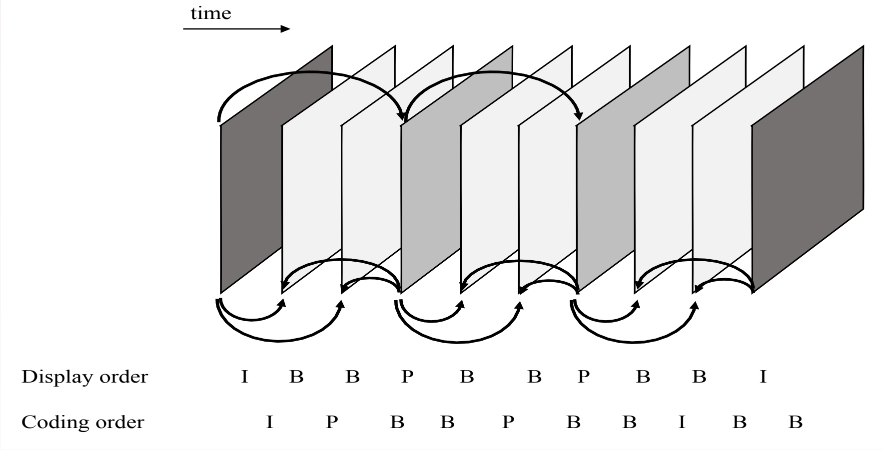
- Two ways to deal with moving material
  - reuse the background in multiple frames
  - move the foreground actor across the scene
  - in practice, these are the same thing (why?)
- MPEG (Moving Picture Experts Group) uses motion compensation
  - attempts to find nearby areas with similar pixel patterns
  - 16x16 “macroblocks” plus motion vectors describing how they shift
  - not trying for perfect representation

MPEG sequencing

- Three types of pictures (frames)
  - **I-picture**: intra-frame compression only
  - **P-picture**: predicted, difference from earlier I
    - 3:1 compression
  - **B-picture**: bidirectional prediction
    - based on earlier I/P, later I/P
    - 4.5:1 compression, but reconstruction is complex

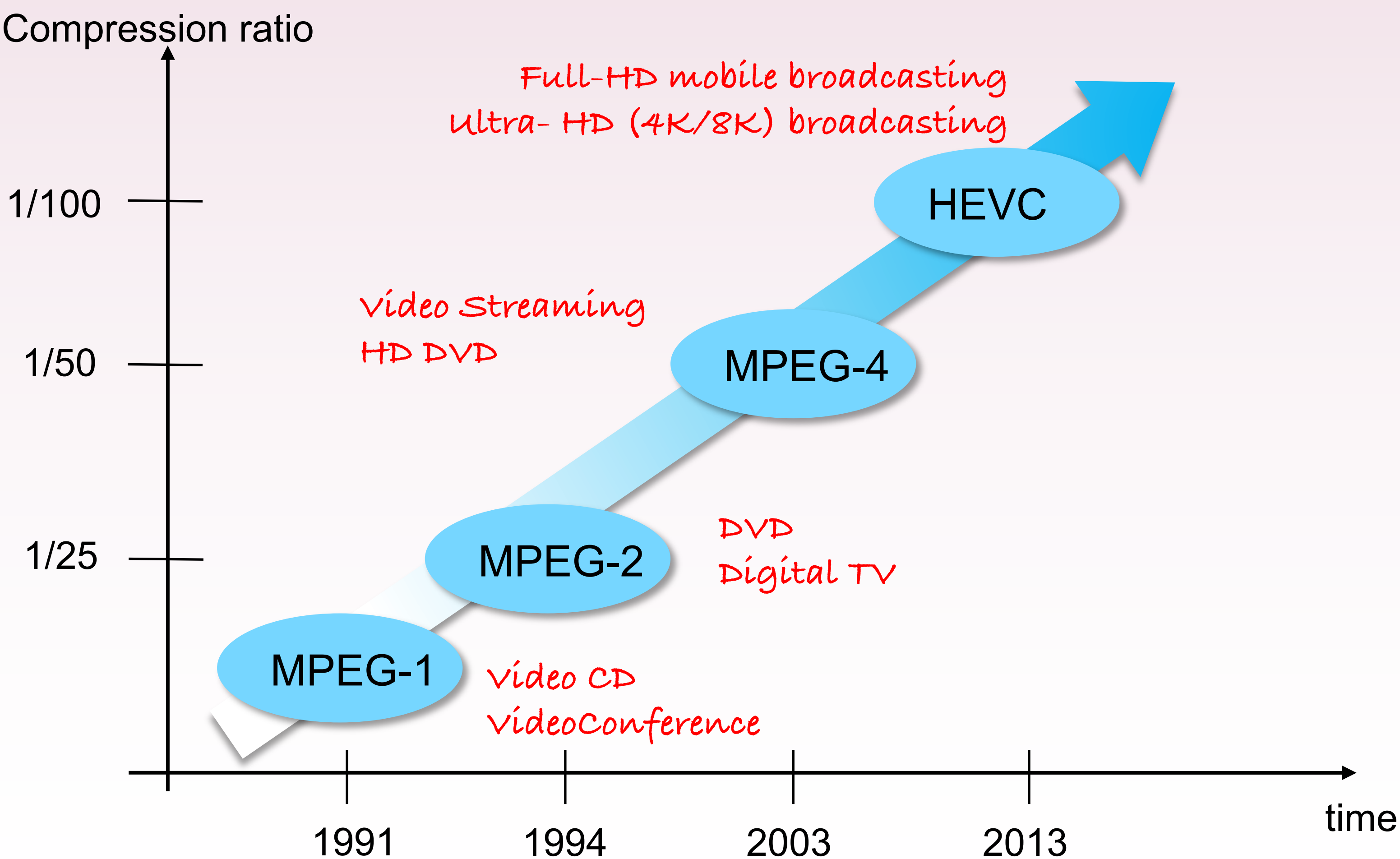
- Group of pictures
  - begins with I-picture
  - IBBPBBPBB is a common pattern

Display vs. Coding ordering



- Display order is the order in which the images should be shown
  - requires decoder to buffer B pictures
- Coding order
  - requires buffering of 2 I/P images
  - first series has unusual order to bootstrap process

History of Video Compression Standards



Competing Codecs

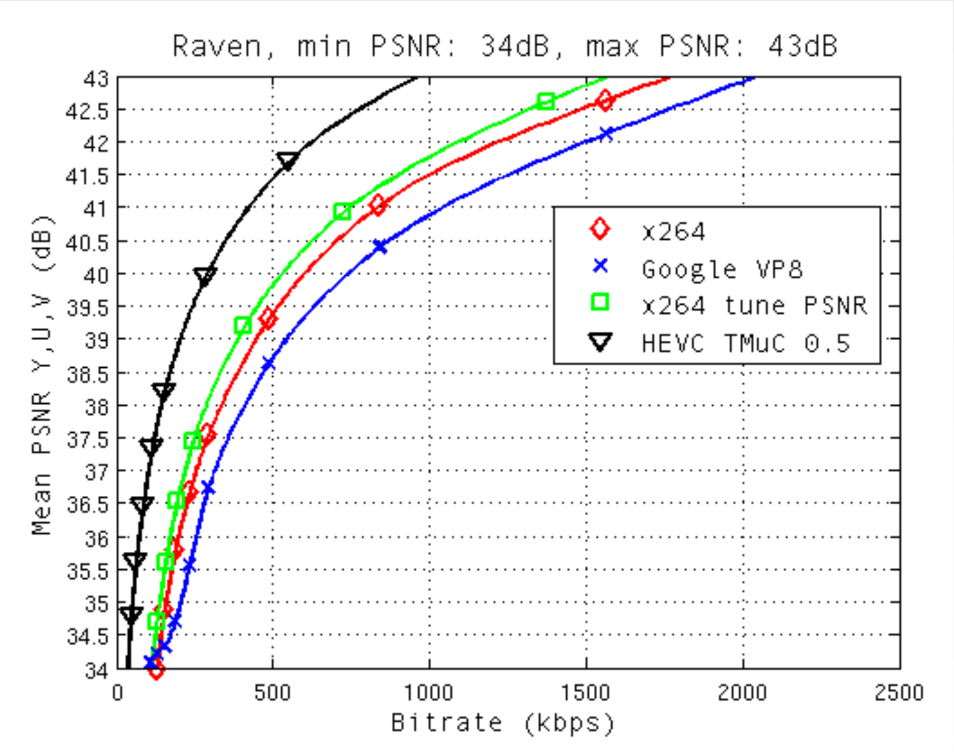
- VP8/VP9,
- AV1
- MPEG/ITU HEVC/H.265

- use scalar or vector quantization
- use temporal compression
- highly asymmetric
- 4K full-HD videos
- Compression up to 50% compared to previous codecs

Competing Codecs

- VP8/VP9,
- AV1
- MPEG/ITU HEVC/H.265

Performance comparison



"Rate-distortion performance of contemporary video codecs: Comparison of Google/WebM VP8, AVC/H. 264, and HEVC TMuC", E Ohwovoriole and Y Andreopoulos - LENS Symp., London, 2010.
http://www.ee.ucl.ac.uk/~iandreop/OHWOVORIOLE_LCS_2010_H264_VP8_HEVC_comparison.pdf

Raw Bandwidth

Raw Bandwidth = color depths * W * L * refresh frequency

Resolution (WxL)	24bit@25fps	8bit@25fps	8bit@15fps	8bit@5fps
4K (3840x2160)	5 Gbit/s	1.6 Gbit/s	995 Mbit/s	331 Mbit/s
HDTV (1920x1080)	1.3 Gbit/s	414 Mbit/s	248 Mbit/s	83 Mbit/s
VGA (640x480)	184 Mbit/s	61 Mbit/s	36 Mbit/s	12 Mbit/s
SCIF (704x576)	240 Mbit/s	80 Mbit/s	48 Mbit/s	16 Mbit/s
CIF (352x288)	60 Mbit/s	60 Mbit/s	12 Mbit/s	4 Mbit/s
QCIF (176x144)	12 Mbit/s	12 Mbit/s	3 Mbit/s	1 Mbit/s

- Uncompressed video is **BIG**
- A DVD would hold max 5 secs of uncompressed video at 1920x1080 resolution
- 6 MHz channel for transmission therefore maximum possible bitrate is 18 Mb/sec
- Requires compression of 83:1

Future Video Applications

Immersive Communications
(holoportation, from Microsoft)

Virtual Reality,
360 Videos

Telepresence
(from Cisco)

Speech/Audio/Video Coding Summary

- Requirements for Coding of each Source
- Speech Coding Principle
- Discontinuous Transmission
- GSM Speech Coding
  - RPE-LPC
  - Enhanced Full Rate
- Different Video Coding Formats, motion compensation

Wireless Channel Coding

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Need for error control

- Mobile radio channel characterised by marked variability due to propagation effects and interference
  - Propagation in outdoor environments: urban, suburban, rural
  - Propagation into buildings; Propagation within buildings
- Path loss and fading considerations
  - Even over small distances radio signals can experience large variations in level of 30 to 40 dB
  - Interference limits scope for increasing signal power to overcome these effects
- Error control provides for reliable, power efficient operation of the radio systems

Need for error control

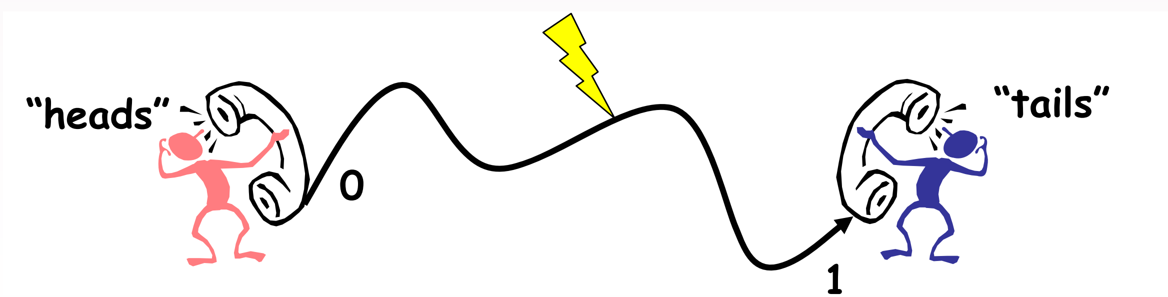
0 - heads

1 - tails

1 bit perfectly encode the information..

… but is it robust enough?

message "0" sent → **during transmission a single-bit error occurs** → message "1" received (different information conveyed)



The information needs to be protected!

Error control coding

- Add redundant bits to a message so that errors can be detected and corrected
- Rate of a code is a measure of redundancy = ratio of message bits to total bits
- May be effected on fixed length data blocks (block codes) or on a continuous data stream (convolutional codes)
- Error control power related to redundancy and block length

Error control techniques

- Automatic Repeat reQuest (ARQ)
  - Full-duplex connection, error detection codes
  - The receiver sends a feedback to the transmitter, notifying if any error is detected in the received packet or not (Not-Acknowledgement (NACK) and Acknowledgement (ACK), respectively).
  - The transmitter retransmits the previously sent packet if it receives NACK.
- Forward Error Correction (FEC)
  - Simplex connection, error correction codes
  - The receiver tries to correct some errors
- Hybrid ARQ (ARQ+FEC)
  - Full-duplex, error detection and correction codes

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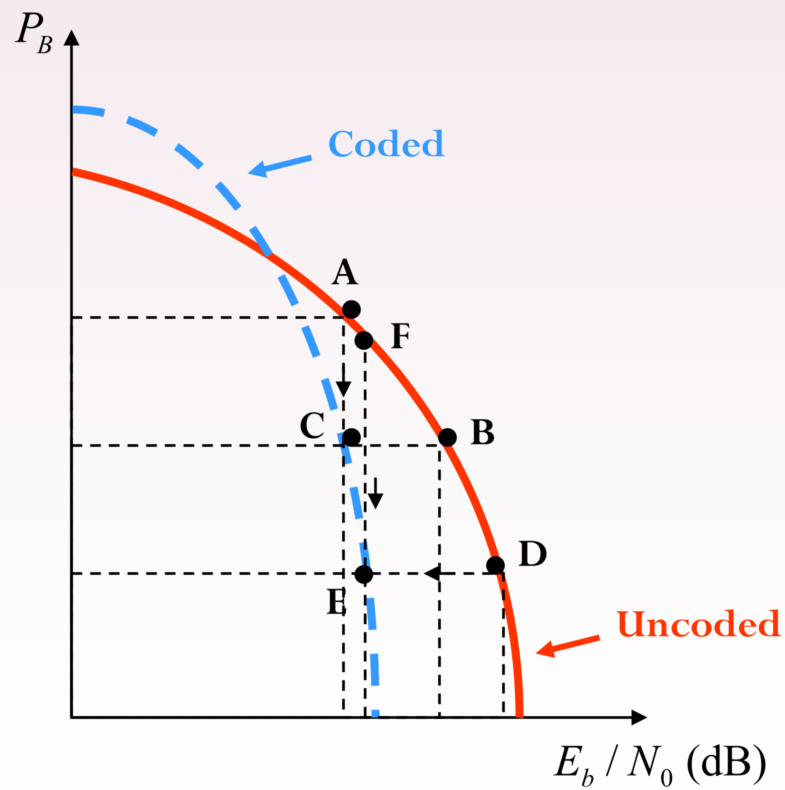
Why using error correction coding?

- Error performance vs. bandwidth
- Power vs. bandwidth
- Data rate vs. bandwidth
- Capacity vs. bandwidth

Coding gain:
For a given bit-error probability, the reduction in the E_b/N_0 that can be realized through the use of code:

$$G\,[\text{dB}] = \left(\frac{E_b}{N_0}\right)_u [\text{dB}] - \left(\frac{E_b}{N_0}\right)_c [\text{dB}]$$



A=(8, 1e-2), C=(8, 1e-4), D=(14, 1e-6), E=(9, 1e-6)

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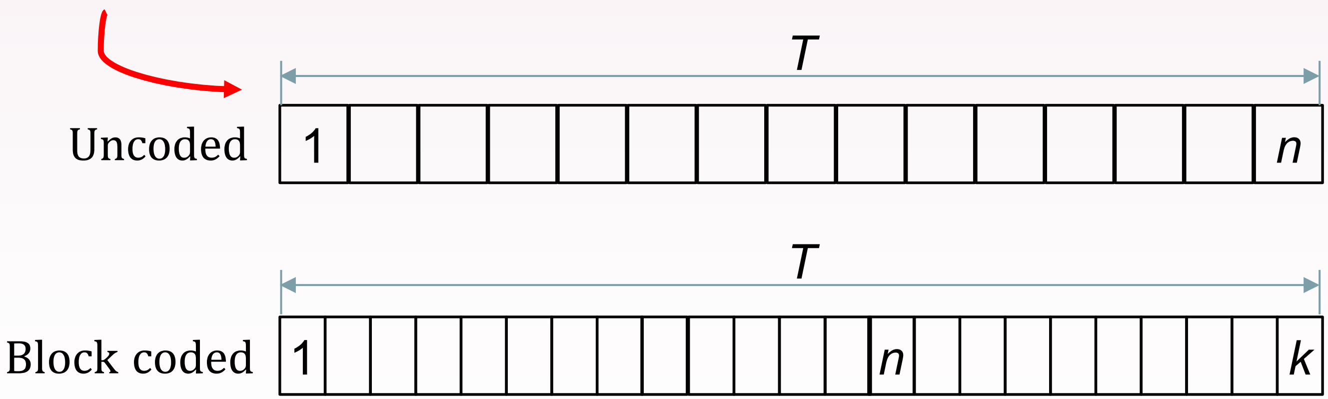
Coding Implementation

- Most systems use a combination of techniques, with the combination depending on the type of data being transmitted
- **Block Codes** provide parity bits for error detection
- **Convolution Codes** generate redundancy needed for error correction in GSM
- UMTS allows for the use of **Turbo codes** as well

We will now review Error Coding Principles

- Block Codes:
  - Referred to as (*n*, *k*) codes
  - Data is divided into blocks of length = *k bits*
  - *n* information bits are coded to form a block of *k* bits
  - The coding rate *R* = *k*/*n*

Block 1	Block 2	Block 3	Block 4	Block 5



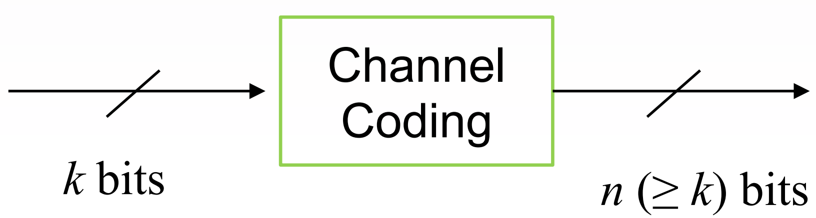
  - This means a higher *R* is required to maintain bit rate

Block code examples

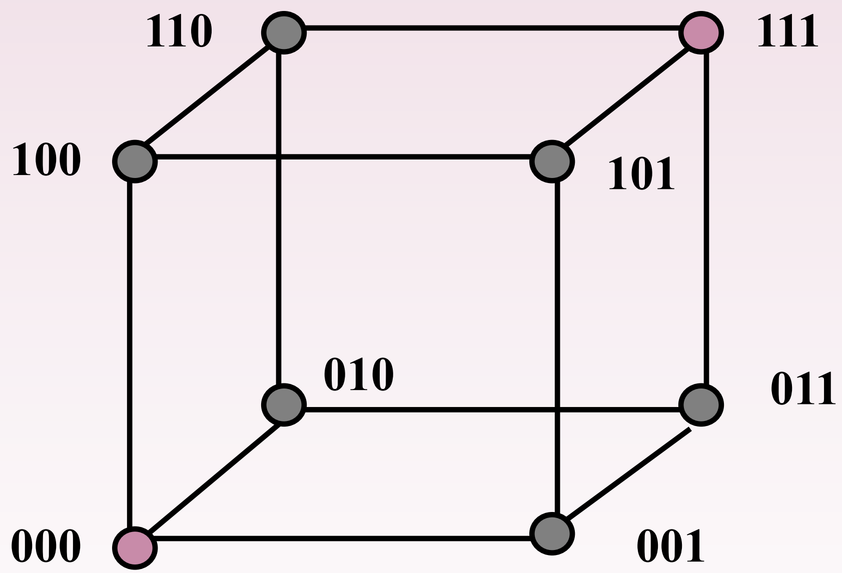
- Hamming (7,4) single error correcting code has block length= 7; k=4 data bits per word; 3 check bits per word; rate R=4/7
- BCH (15,7) is a double error correcting code; n=15, k=7, R=7/15
- RS(255, 239) has 255 symbols per block with each symbol represented by 8 bits (finite field of degree 256); Block length is this 2040 bits with 1912 data bits; R=239/255

Hamming Distance

- Hamming distance
  - the number of symbol positions in which two code words vary is denoted the Hamming distance, d
  - e.g 1001 and 1000 are separated by Hamming distance 1, while 1001 and 0110 are separated by distance 4
- Redundancy
  - binary words of length *n* carrying only *k* information bits contain redundancy which, if properly structured can provide for error detection and correction in an (*n*,*k*) code
- Rate
  - The ratio *k*/*n* is the code rate



Hamming Distance



- 3-bit binary words arranged in 3-dimensional space on a cube
- minimum Hamming distance between words is 1
- 000 and 111 have Hamming distance 3 and represent an ECC
- Code rate is 1/3, single errors can be corrected

Parity Checks

Information bits	Parity bits

- We can add a parity bit to a word to ensure it has a defined parity - 1110 has odd parity (odd number of 1's)
- A single error will then disturb the parity and can thus be detected, this is the basis of error detection coding
- By introducing parity bits applied across subsets of the information bit positions it is possible to both detect and correct errors - the basis of error correction codes (ECCs)
- The parity bits are sometimes mixed in with the information bits rather than attached at the end of the codeword

Array Parity EDC code

row checks

0	1	0	0	0
1	1	0	0	1
0	1	0	1	1
1	1	1	0	0
1	1	0	0	1

column checks

check on row and column checks

- Odd parity is applied to rows and columns
- A single error may be detected and corrected (EDC)

Hamming (7,4) code

- 7-bit codewords: 4 information bits plus 3 check (parity) bits
- With 4 information bits there are $2^4=16$ possible codewords
- 3 equations define the check bits
- $d1,d2,d3,d4$:$c1,c2,c3$ define data and check bits
- Non-zero syndrome [S] defines location of error

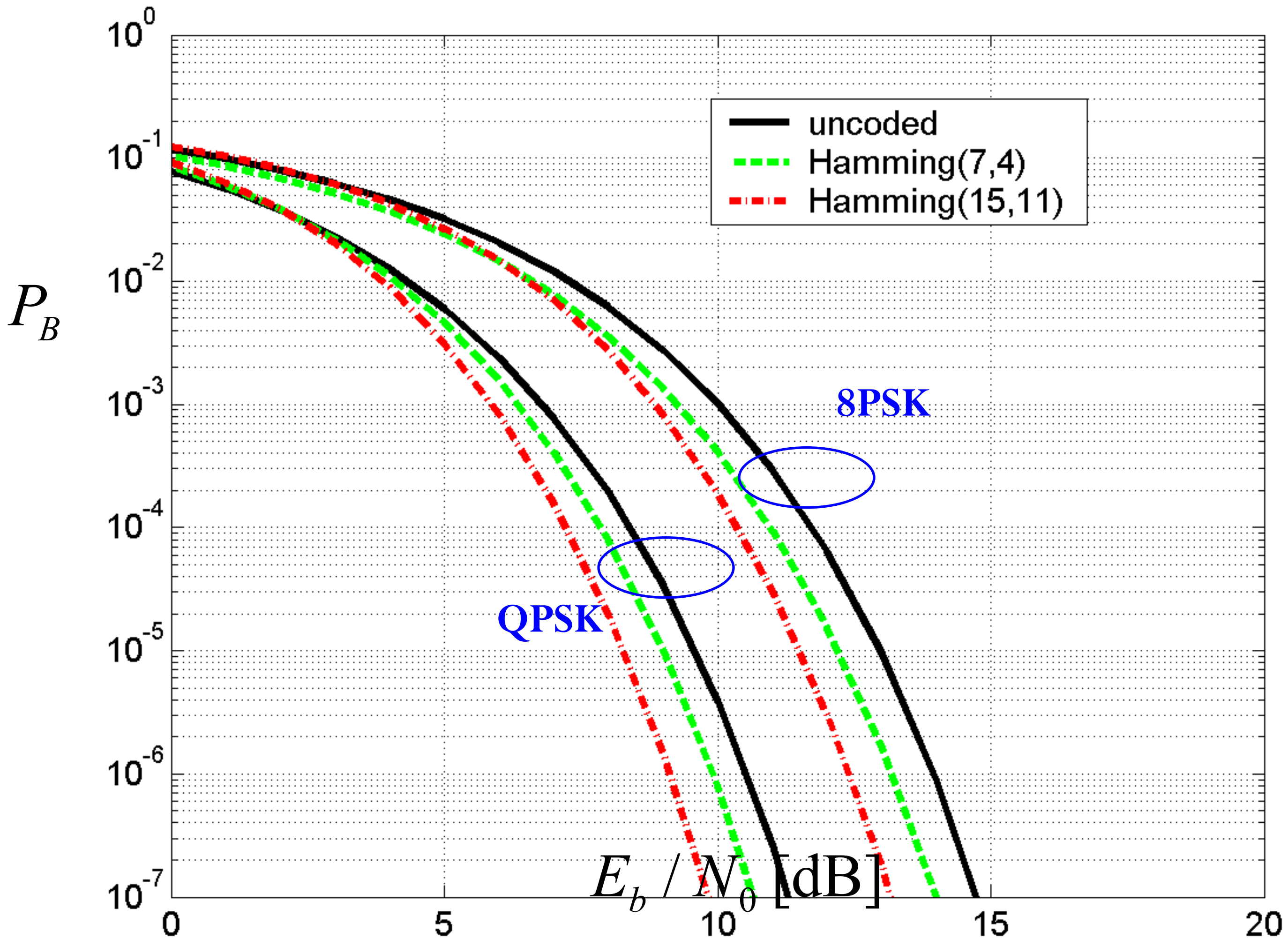
$$[c1,c2,d1,c3,d2,d3,d4]$$
$$c1 = d1 \oplus d2 \oplus d4$$
$$c2 = d1 \oplus d3 \oplus d4$$
$$c3 = d2 \oplus d3 \oplus d4$$

$$[S] = \begin{cases} s1 = c3 \oplus d2 \oplus d3 \oplus d4 \\ s2 = c2 \oplus d1 \oplus d3 \oplus d4 \\ s3 = c1 \oplus d1 \oplus d2 \oplus d4 \end{cases}$$

Multiple error correction

- Hamming distance determines the error correcting properties
- Distance of two allows single error detection, distance three allows single error correction
- Distance to 2t+1 allows up to t errors to be corrected - a t-error correcting code
- An (n,k) code has length n implying 2^n possible binary words of length n and 2^k codewords, giving distance up to (n-k) and 2^{n-k} binary words which are not allowed code words
- Hamming codes have n=2^c-1 and k=2^c-1-c for any integer c

Example of Hamming codes



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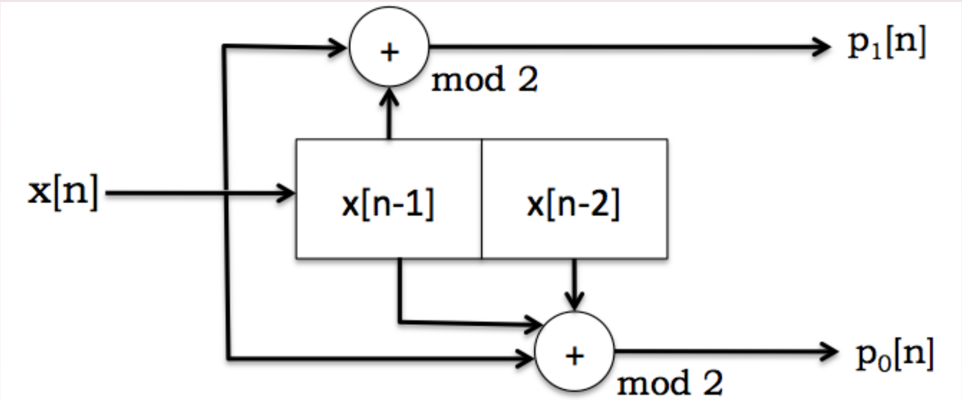
Convolutional Codes



- Like the block codes, they involve the transmission of parity bits that are computed from message bits
- Unlike block codes, the sender does not send the message bits followed by the parity bits; in a convolutional code, the sender *sends only the parity bits*.
- The encoder uses a *sliding window* to calculate r > 1 parity bits by mixing or current and preceding digits on a continuous basis

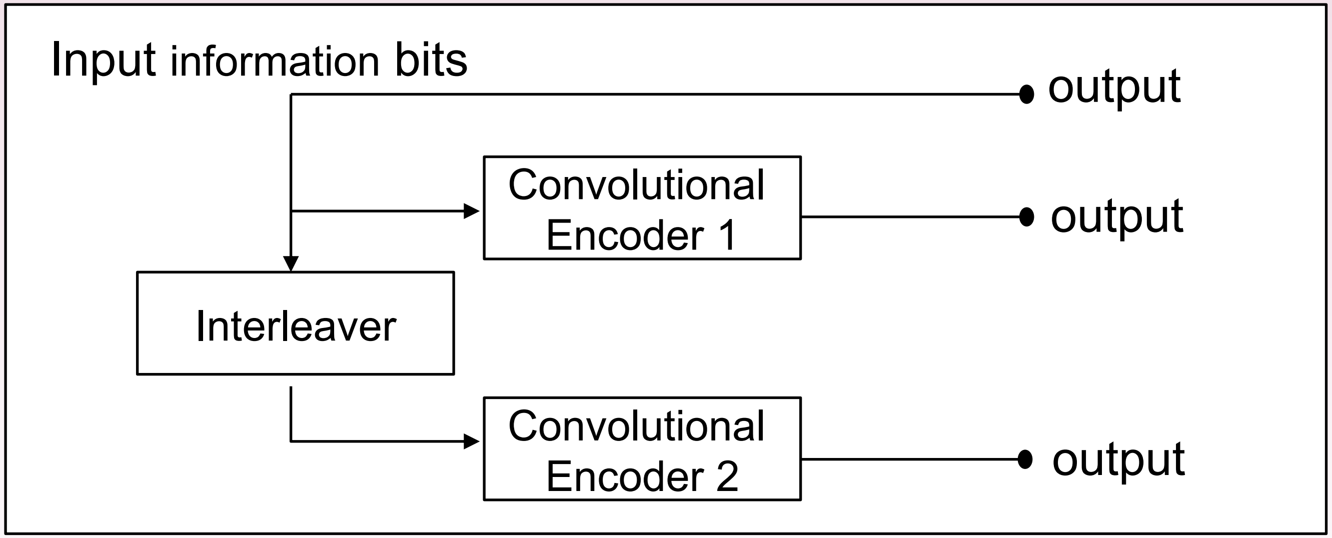
Convolutional Codes

- Used in a variety of systems including today's popular wireless standards (such as 802.11) and in satellite communications.

- Tend to be relatively low rate, such as 1/2, 3/4

- Punctured convolutional codes can have higher rates: 7/8 would be a high rate (although some very high rate codes have been devised recently - at the cost of implementation complexity)

Turbo Codes



- Formed as parallel concatenated convolutional codes
- Turbo decoders use a pair of SISO (Soft Input Soft Output) decoders

Turbo Codes

- Relatively new invention (circa 1993)
- Can perform very closely to Shannon's theoretical channel limit
- Adopted in 3G/4G mobile communications and in satellite communications
- UTRAN uses a parallel concatenated convolutional code (PCCC), which is two convolutional encoders in parallel separated by an interleaver

Error Control in GSM

- Channel coding
  - Improves transmission quality when interference, multipath fading and Doppler shift are encountered
  - The bit/frame/word error rates are reduced, but throughput is also reduced
- Interleaving
  - Interleaving scrambles and 'spreads in time' a sequence of bits prior to transmitting the.
  - Bursts of errors occurring in transmission thus affect adjacent 'channel' bits, corresponding to widely separated 'message' bits
  - De-interleaving at the receiver puts back the message bits back again and separates

Interleaving

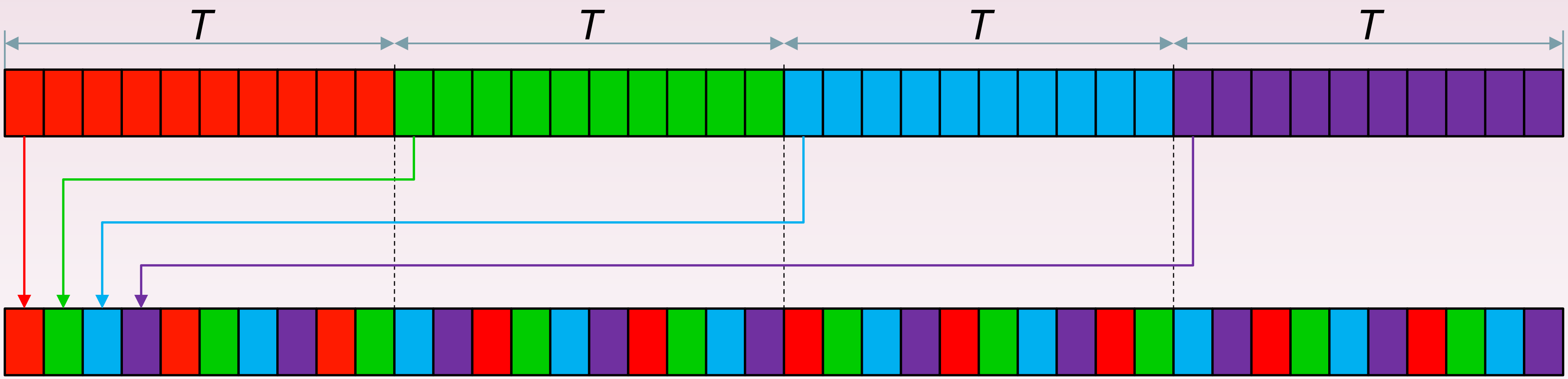
- Interleaving is intended to de-correlate the relative positions of the bits respectively in the code words and in the modulated radio bursts
- Useful because bit errors tend to occur consecutively, both due to statistics of radio transmission and because of intersymbol interference introduced by modulation processes and circuits
- Interleaving involves spreading *b* bits of a code into *n* bursts so as to change the proximity relations between bits
- The larger the value of *n* the better the transmission performance but the longer the transmission delay - hence a compromise is required
- Several interleaving schemes are used in GSM, depending on channel usage

Interleaving - continued

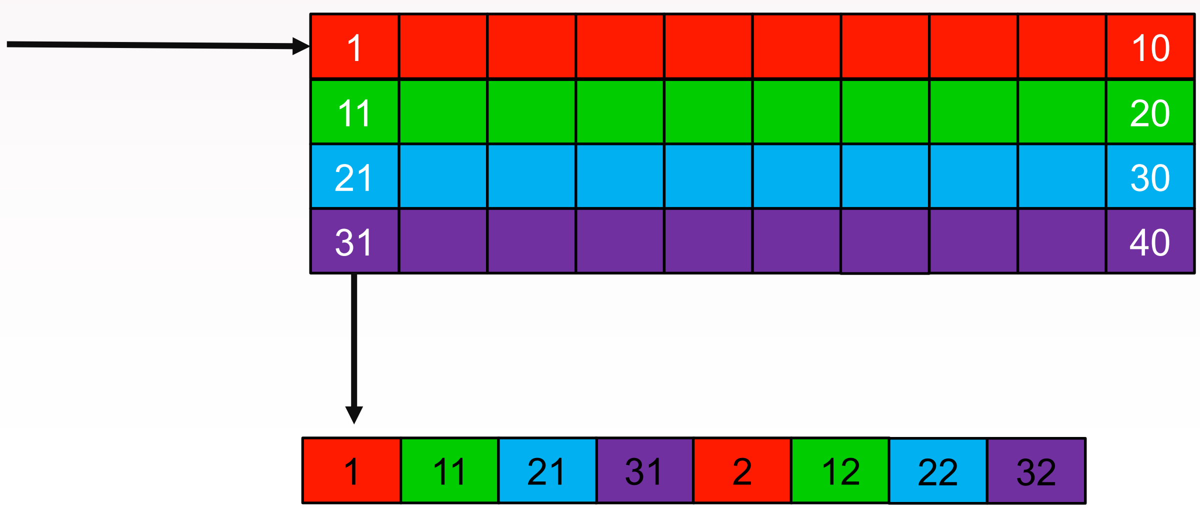
- Simple arithmetic relations between *b* and *n* and the number of bits per burst (4x114 in GSM) reduces complexity
- Example - for a code word of *b*=456 = 4x114 bits we could adopt:
- 4 parts of 114 bits, each filling up a whole burst
- 8 parts of 57 bits, each filling half a burst
- GSM also uses a markedly more complex interleaving scheme for 9.6kbit/s data based on an arrangement involving 76 parts of 6 bits each using one 19th of a burst, referred to as "19 bursts interleaving" (although since 19 does not divide 456 it is actually effected on 22 bursts!)

Channel Coding

- Interleaving:

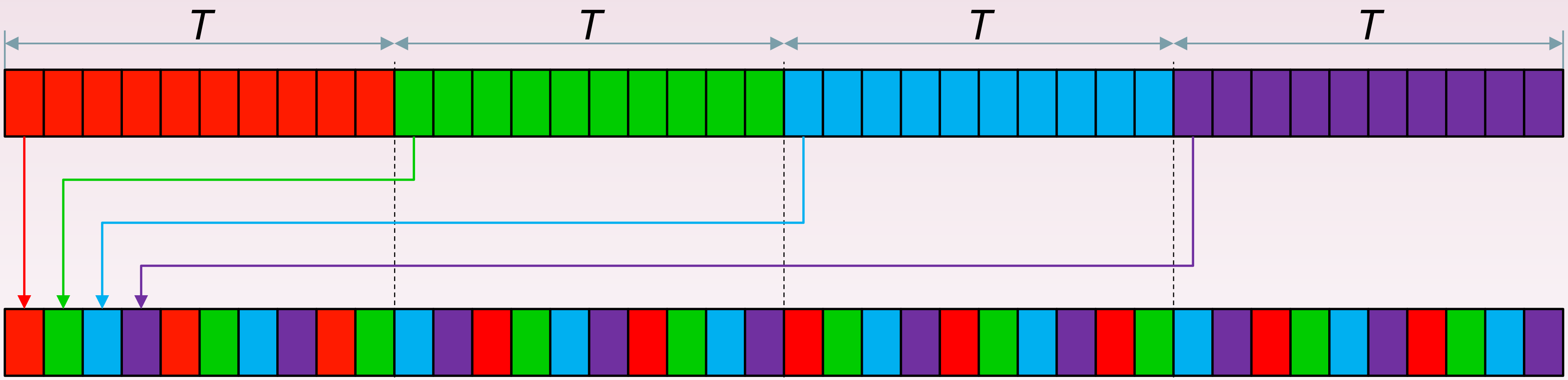


- How?

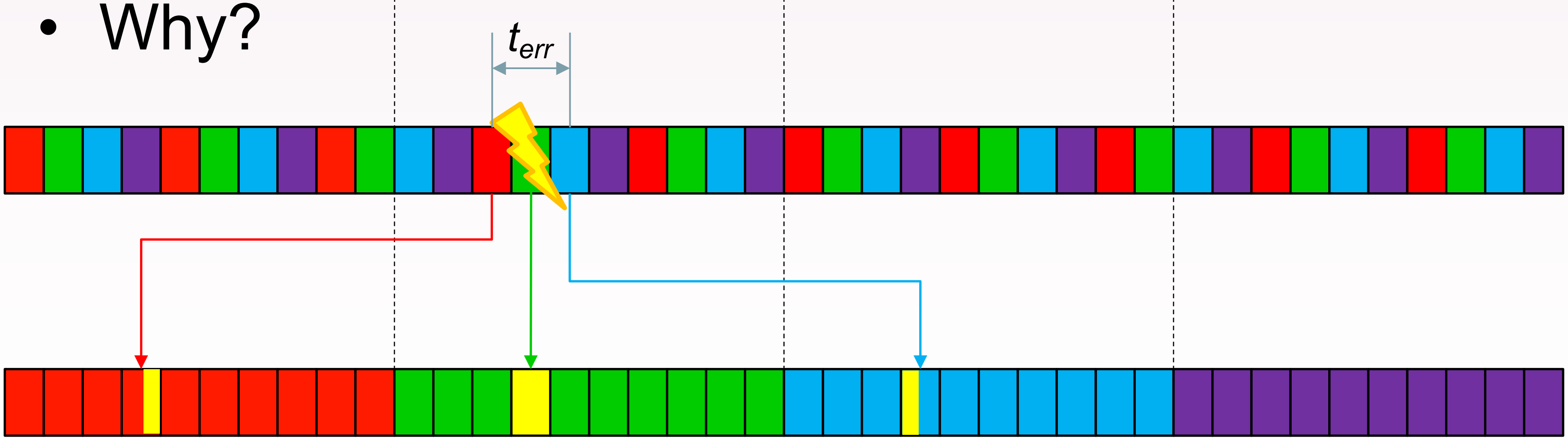


Channel Coding

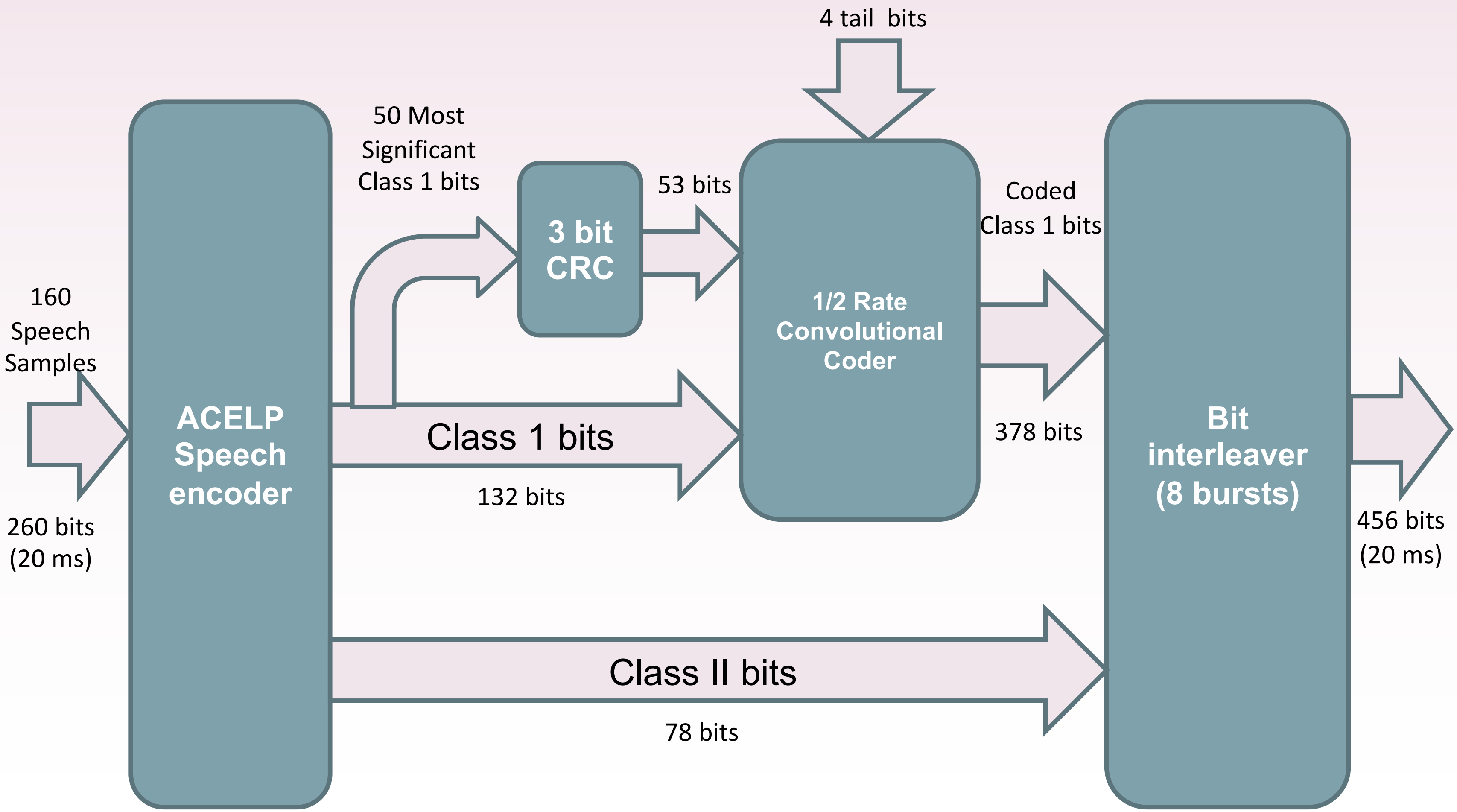
- Interleaving:



- Why?



- Example of coding in GSM:
  - Combined source & channel coding



Most Popular Channel Coding Techniques

- Convolutional coding (Elias 1955): GSM, WiFi IEEE 802.11 ac
- Turbo coding (Claude Berrou 1993): UMTS, LTE.
- Low density parity check (LDPC) coding (Gallager, 1963): 5G, WiFi 802.11 ax
- Polar coding: 5G

Error control techniques

- Automatic Repeat reQuest (ARQ)
  - Full-duplex connection, error detection codes
  - The receiver sends a feedback to the transmitter, notifying if any error is detected in the received packet or not (Not-Acknowledgement (NACK) and Acknowledgement (ACK), respectively).
  - The transmitter retransmits the previously sent packet if it receives NACK.
- Forward Error Correction (FEC)
  - Simplex connection, error correction codes
  - The receiver tries to correct some errors
- Hybrid ARQ (ARQ+FEC)
  - Full-duplex, error detection and correction codes

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Why Channel Coding?

Professor Robert Fano MIT, interview in 2001:

"I remember John Pierce at Bell Labs… He was Shannon's boss. He was playing down the importance of the noisy channel theorem, saying: 'just use more bandwidth, more power'… there was no limitation then – you could do whatever you needed in terms of reliable communication without long encoding. And besides, even if you wanted to, you were very badly limited by equipment complexity and cost…"

6.933 Project History, Massachusetts Institute of Technology
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