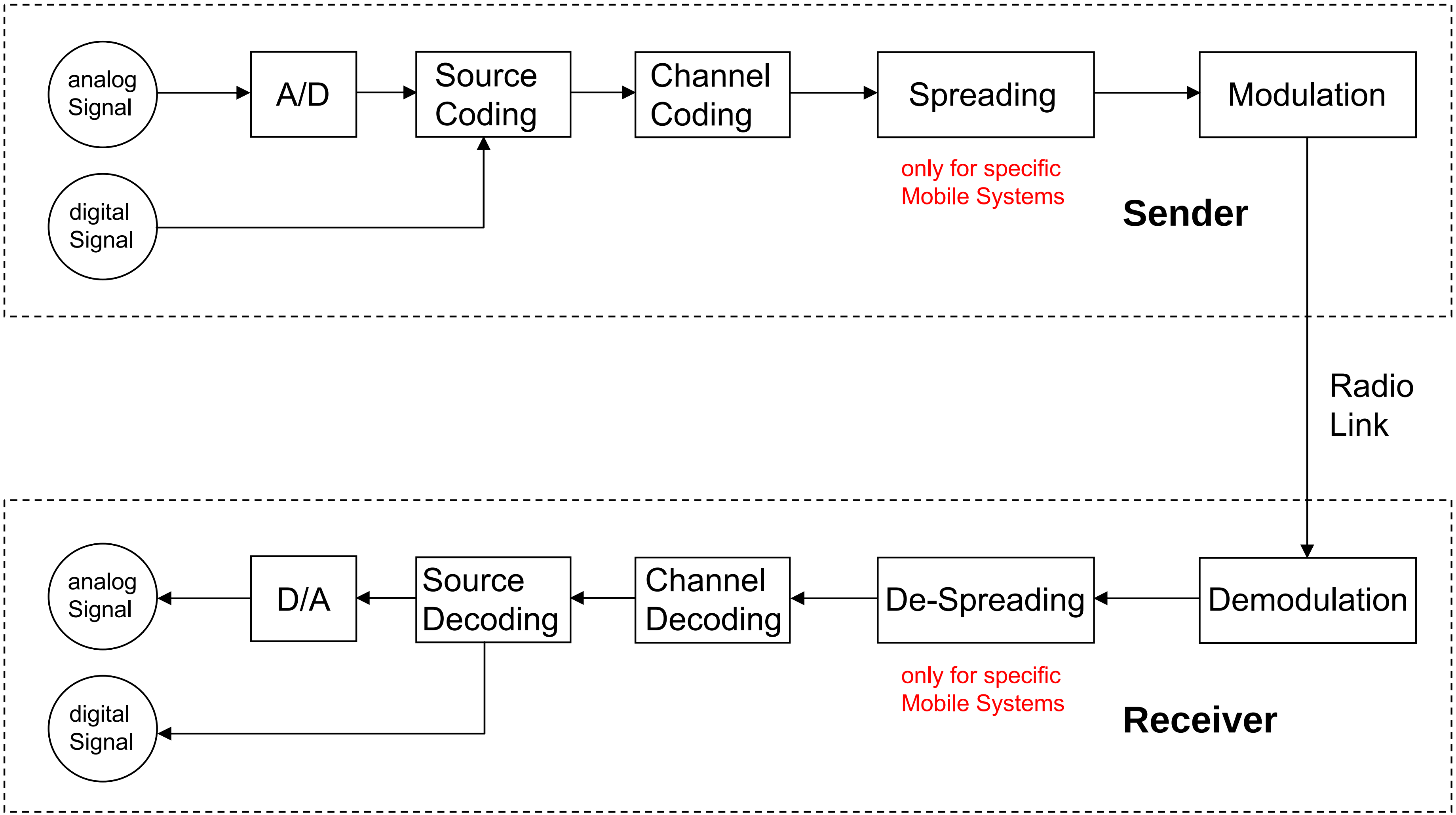
Source and Channel Coding

Overview



analog Signal
A/D
Source Coding
Channel Coding
Spreading
Modulation
digital Signal
only for specific Mobile Systems
Sender
Radio Link
analog Signal
D/A
Source Decoding
Channel Decoding
De-Spreading
Demodulation
digital Signal
only for specific Mobile Systems
Receiver

Source Coding

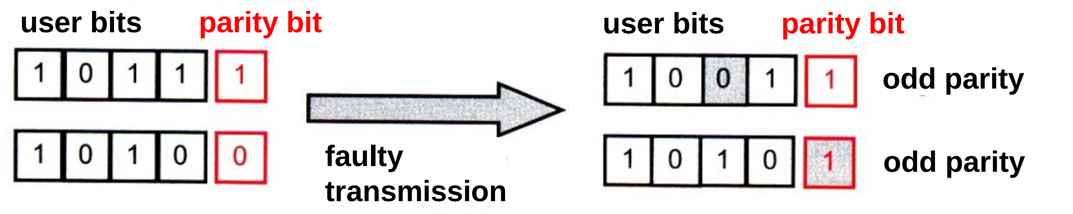
- Purpose of source coding:
  - the analog signal (i.e. the signal that has to be encoded) is converted into a binary sequence; the focus of source coding is on efficient data transmission (compression); source information should not be distorted
- Example: voice codecs in cellular mobile networks
  - Full Rate Codec (RPE-LTP)
    - original GSM voice codec
    - 260 bit per 20ms → 13 kbit/s (compression factor 8)
  - Half Rate Codec
    - voice quality comparable with Full Rate Codec, but the quality of non-voice signals (music, …) is considerably poorer
    - 6,5 kbit/s (compression factor 16)
  - Enhanced Full Rate Codec (ACELP)
    - better voice quality compared to Full Rate Codec
    - 244 bit per 20 ms → 12,2 kbit/s (compression factor 8,5)
  - Adaptive Multi-Rate Codec (AMR)
    - different bitrates possible: 12,2 kbit/s - 4,75 kbit/s
    - in-band signaling to adjust the rate

Channel Coding

- Purpose of channel coding:
  - (bit) error detection
  - (forward) (bit) error correction
- Basic principle of channel coding:
  - additional redundancy bits (= overhead) are added to user bits, to allow error detection or correction
- Definition of "code rate":
  - code rate = ratio of user bits / (user bits + redundancy bits)
- Channel coding methods:
  - block codes
  - convolution codes
  - turbo codes
  - ....

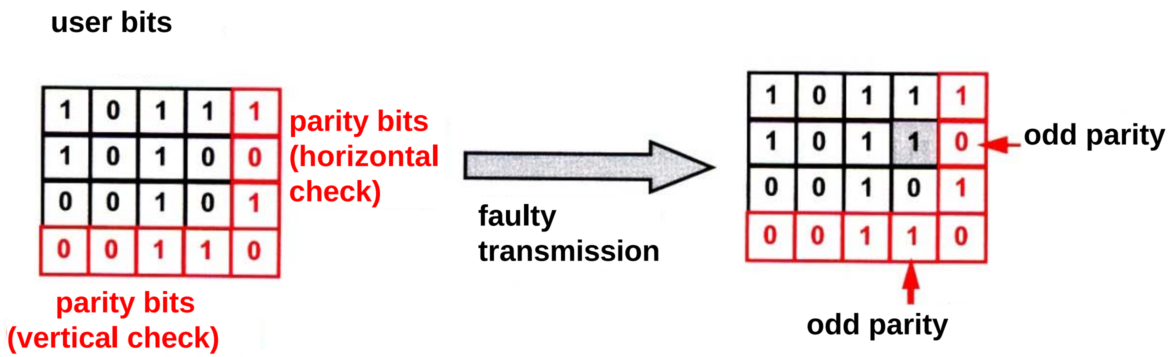
Channel Coding - Block Codes

- Example: block coding to detect a single bit error
  – a parity bit is added to the user bits, so as to achieve always an even number of “1”



one single bit error (grey field) can be detected, but cannot be corrected

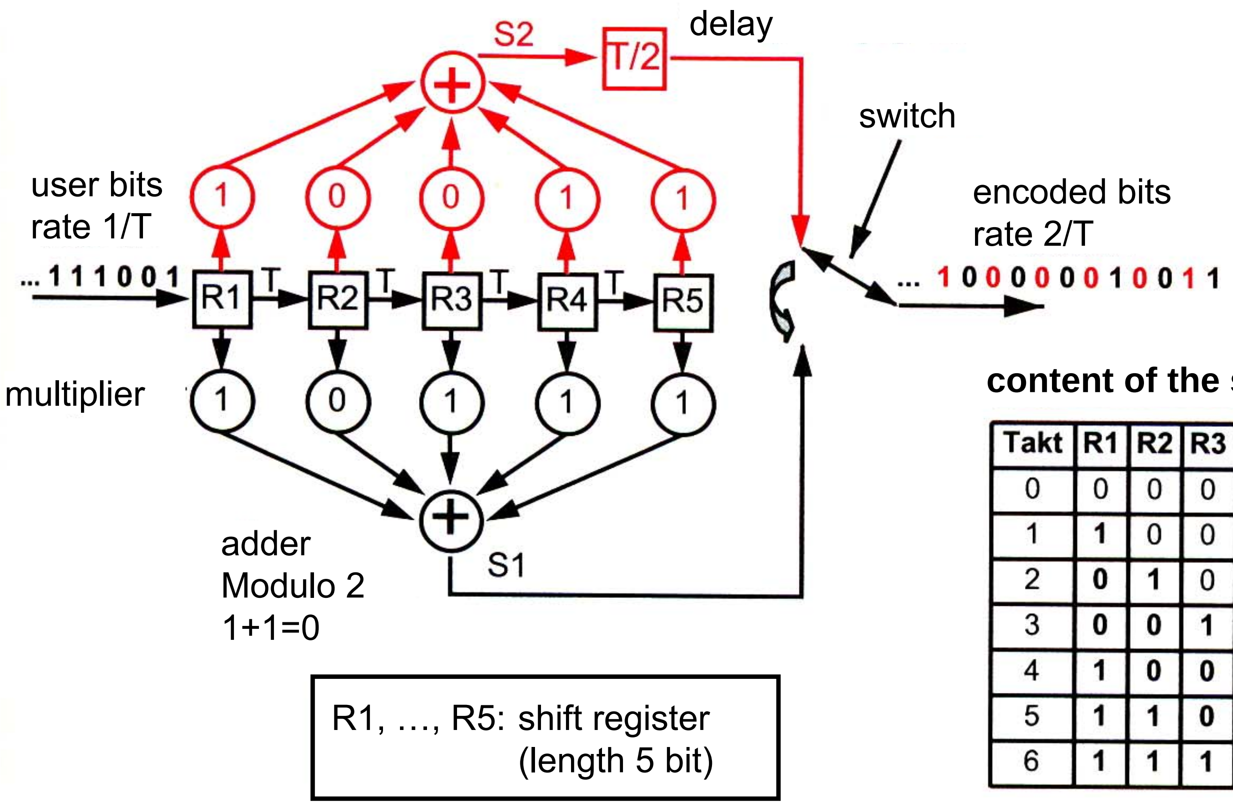
- Example: block coding to correct a bit error
  – the user bits are aligned to a rectangular **block** and parity bits are added for horizontal and vertical parity check



the bit error in row 2, column 4 (grey field) can be corrected

Channel Coding - Convolution Codes

- Example: convolution coding (with code rate 1/2)
  - direct coding of a bit stream "on the fly"
  - implemented by using shift register, multiplier and adder

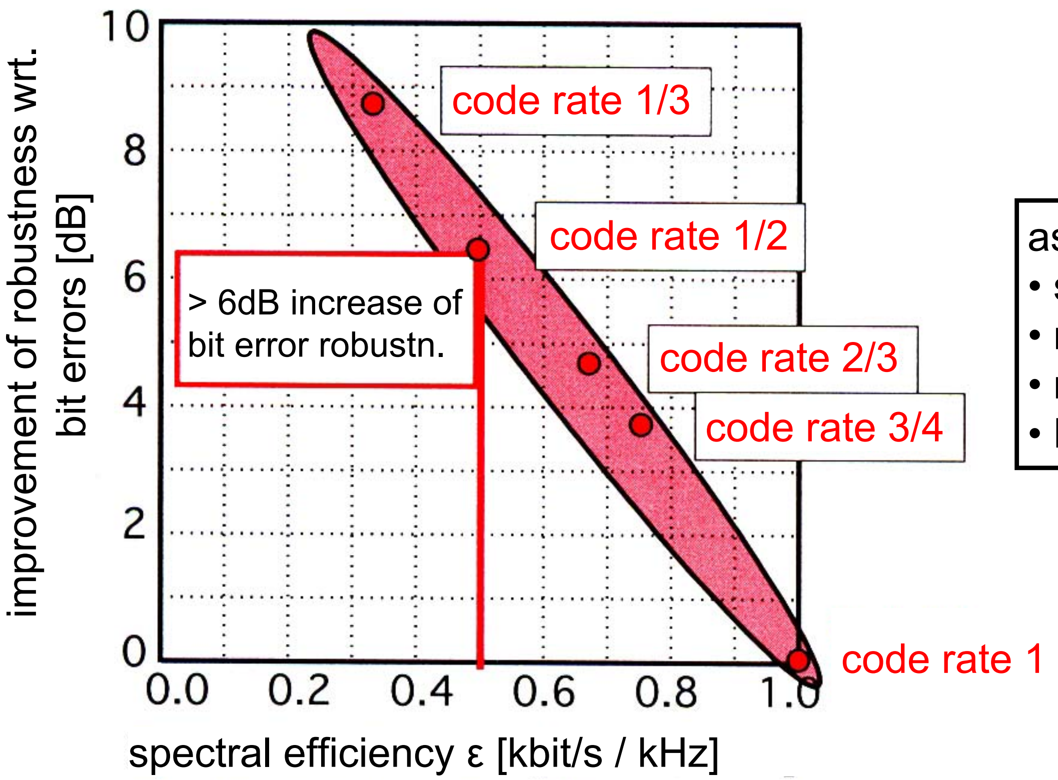


content of the shift register:

Takt	R1	R2	R3	R4	R5	S1	S2
0	0	0	0	0	0	0	0
1	1	0	0	0	0	1	1
2	0	1	0	0	0	0	0
3	0	0	1	0	0	1	0
4	1	0	0	1	0	0	0
5	1	1	0	0	1	0	0
6	1	1	1	0	0	0	1

Channel Coding - Coding Gain

- user data rate (spectral efficiency) vs. robustness wrt. bit errors (SNR)
- remark: code rate = ratio of user bits / (user bits + redundancy bits)



assumptions:
- shift register length: 6 bit
- no Rayleigh fading
- required BER: 0.1%
- BPSK modulation

Error Protection

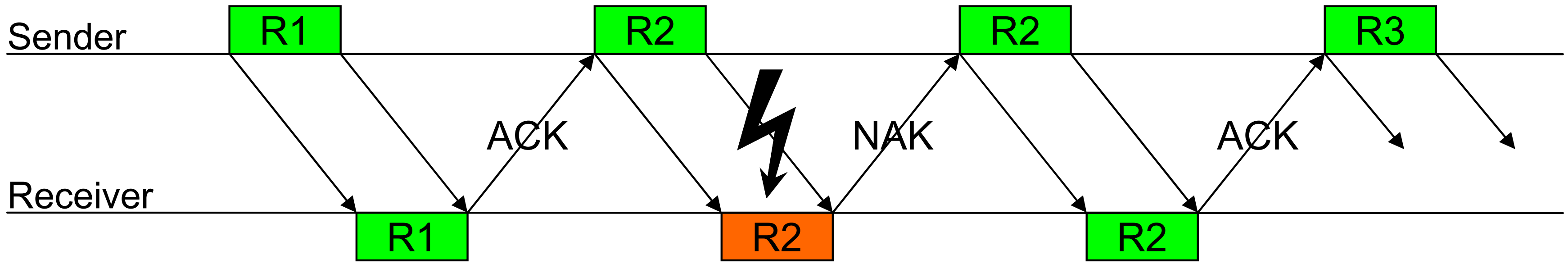
Motivation

- Error protection methods (applied in mobile communication):
  - Forward Error Correction (FEC) via channel coding
    - especially useful in case of single bit errors

    already addressed in the section about "Channel Coding"
  - Automatic Repeat Request (ARQ)
    - useful, if whole data blocks are lost
    - advantage: saves channel coding overhead
    - disadvantage: additional delay, therefore not suited for real time transmission
    - usually works at layer 2
  - Combinations of FEC and ARQ (Hybrid ARQ schemes)
    - optimizes the transmission overhead
  - Interleaving
    - re-sorting of data blocks
    - useful in the case of burst errors
- Problem: tradeoff overhead ↔ robustness wrt. bit errors

ARQ Schemes

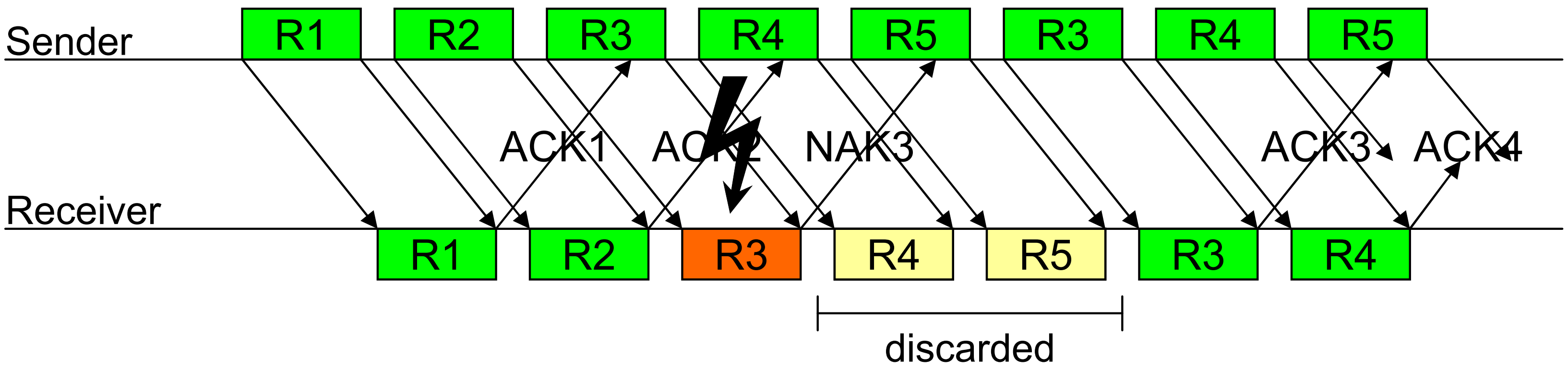
- Stop-and-Wait ARQ
  - operation mode:
    - the sender transmits a frame and waits for the acknowledgement
    - after receiving the frame, the receiver returns an ACK or NAK
    - after receiving an ACK the sender transmits the next frame, after receiving an NAK the sender repeats the current frame
  - example:



  - disadvantages:
    - long (= unused) time between the transmission of two frames (especially in case of faulty frames)

ARQ Schemes

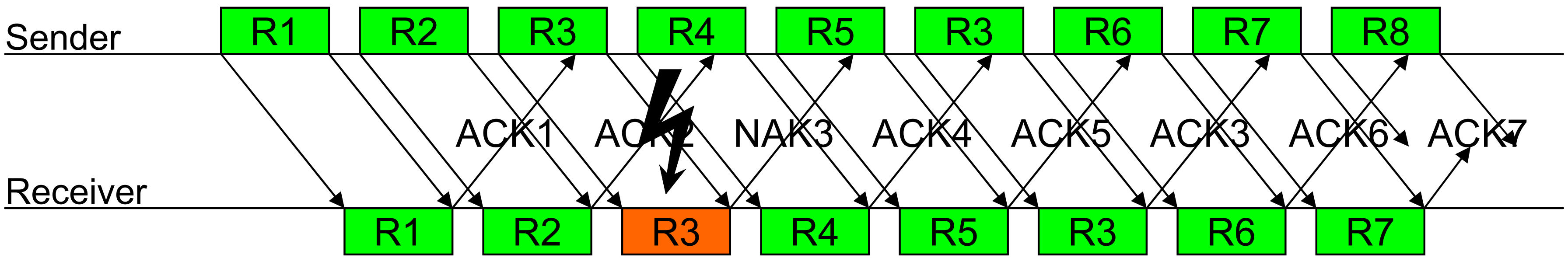
- Go-back-N ARQ
  - operation mode:
    - the sender transmits frames without waiting for an acknowledgement; but: only a limited number of unacknowledged frames are allowed (sliding window)
    - if a NAK is received for frame F_k then all frames up to frame F_k are transmitted again
  - example:



  - disadvantages:
    - the scheme is more complex than Stop-and-Wait ARQ: it requires numbered frames and a transmission buffer
    - correctly received frames might be discarded

ARQ Schemes

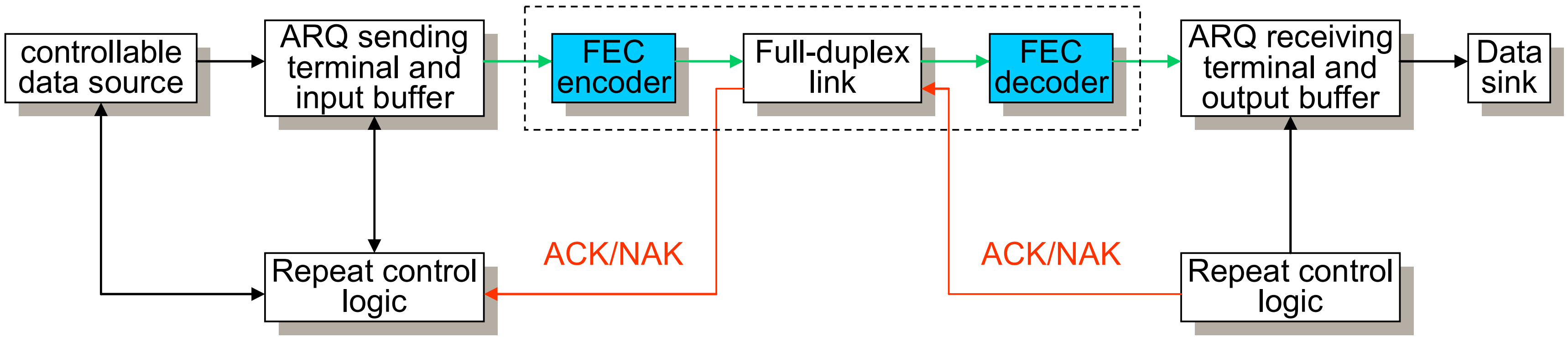
- Selective-repeat (request) ARQ
  - operation mode:
    - the sender transmits frames without waiting for an acknowledgement (sliding window principle)
    - if a NAK is received for frame F_k then *only* frame F_k is transmitted again; meanwhile all correctly received frames are buffered at the receiver to allow to restore the right order of frames
  - example:



  - disadvantages:
    - the receiver also requires a buffer
    - this scheme is even more complex to implement

Hybrid ARQ Schemes

- Motivation:
  - mobile communication might be affected by strong channel impairments; therefore the probability of receiving faulty data blocks (frames) is quite high and a high number of retransmissions might be required
  - if data blocks are repeated more than once on average, Forward Error Correction (FEC) with a code rate of 1/2 is more efficient than ARQ
  - therefore it is reasonable to align the FEC and the ARQ mechanism → this reduces the number of transmissions of faulty data blocks
  - basic principle:



Hybrid ARQ Schemes

- Type 1 Hybrid ARQ:
  - operation mode:
    - every frame contains the same number of redundancy bits for error detection/correction (the code rate depends on the specific environment; usual code rate values are 1/3, 2/3)
    - in case a faulty frame is received and the error cannot be corrected following actions are performed:
      - the (faulty) frame is discarded
      - an identical copy of the frame is requested

      this is repeated until the frame is received correctly or the faulty frame can be corrected
  - disadvantage:
    - *fixed* number of redundancy bits $\Rightarrow$ no adaptation to the channel quality
      (in case of good channel quality the achievable throughput is lower compared to a pure ARQ scheme)

Hybrid ARQ Schemes

- Type 2 Hybrid ARQ:
  - basic idea:
    - adaption to the current channel quality
  - operation mode:
    - the first version of a frame contains only a few redundancy bits (usual code rate values: 7/8, 8/10)
    - faulty frames are buffered
    - only redundancy bits are transmitted again (with code rate 1/3 or 1/2)

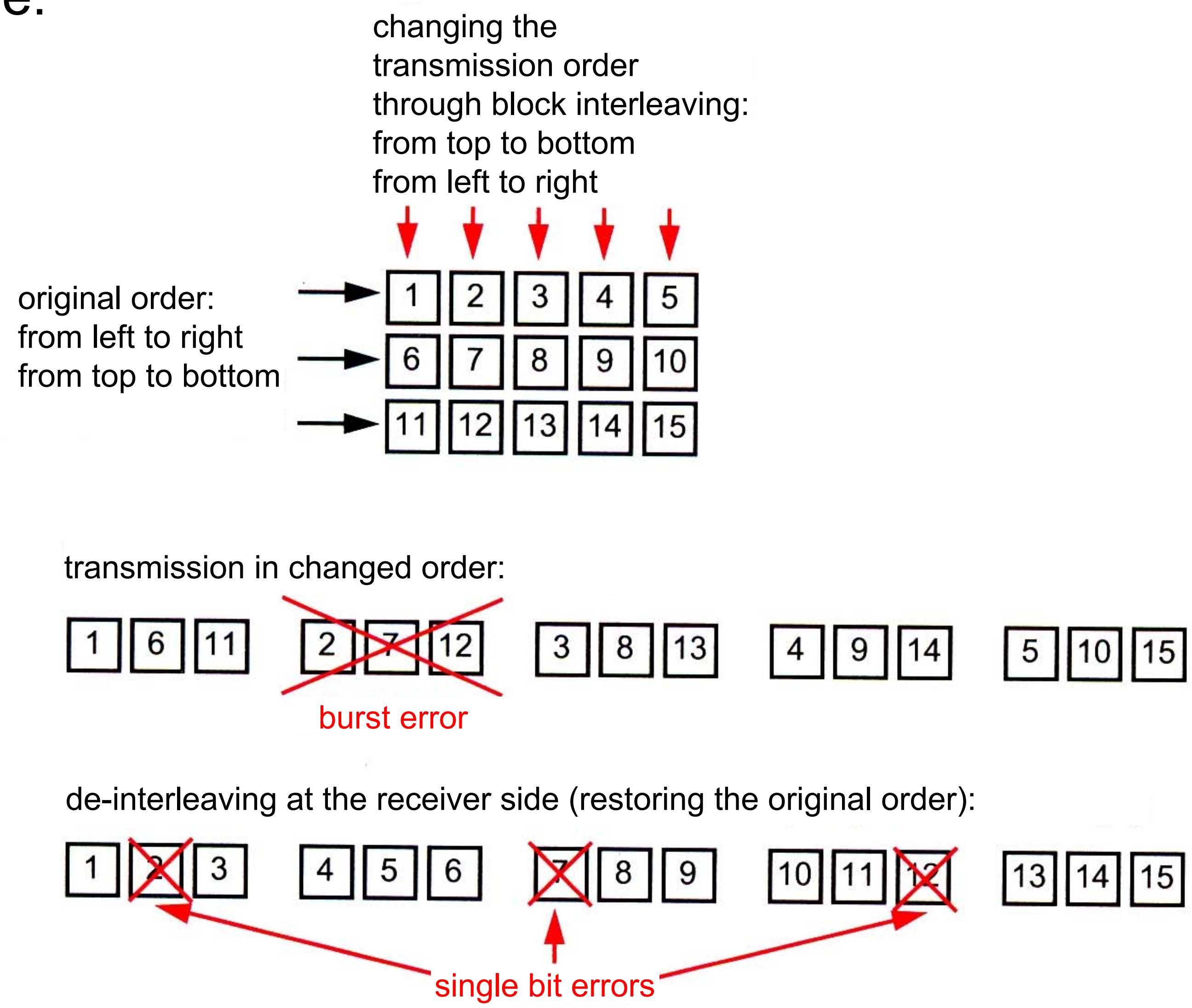
      $\Rightarrow$ stronger encoding (of redundancy bits) for error correction
    - finally, if it is not possible to correct the frame $\Rightarrow$ the first version of the frame is requested again and is going to be transmitted
  - advantage:
    - in case of low error probability the low redundancy (= low overhead) of the first version frame is sufficient
  - disadvantage:
    - it possibly takes a longer time until the faulty frames are transmitted correctly

Interleaving

- Interleaving is used in mobile communication to compensate burst errors (caused e.g. by Slow Fading)
- Operation mode:
  - the transmitter re-sorts a data block (containing user bits and redundancy bits) before starting the transmission
  - the receiver recovers the data block i.e. resumes the original order; the bit errors (that during transmission typically occur in bursts i.e. subsequent bits) are now spread over the whole data block
    → burst errors become single bit errors
- Definition of interleaving depth:
  - the interleaving depth is the length (in bits) of the data block on which the re-sorting is applied; the larger the interleaving depth is, the more bits are spread and the interleaving becomes more efficient
- Disadvantage:
  - additional transmission delay
- Remark: frequency hopping might also help in case of burst errors

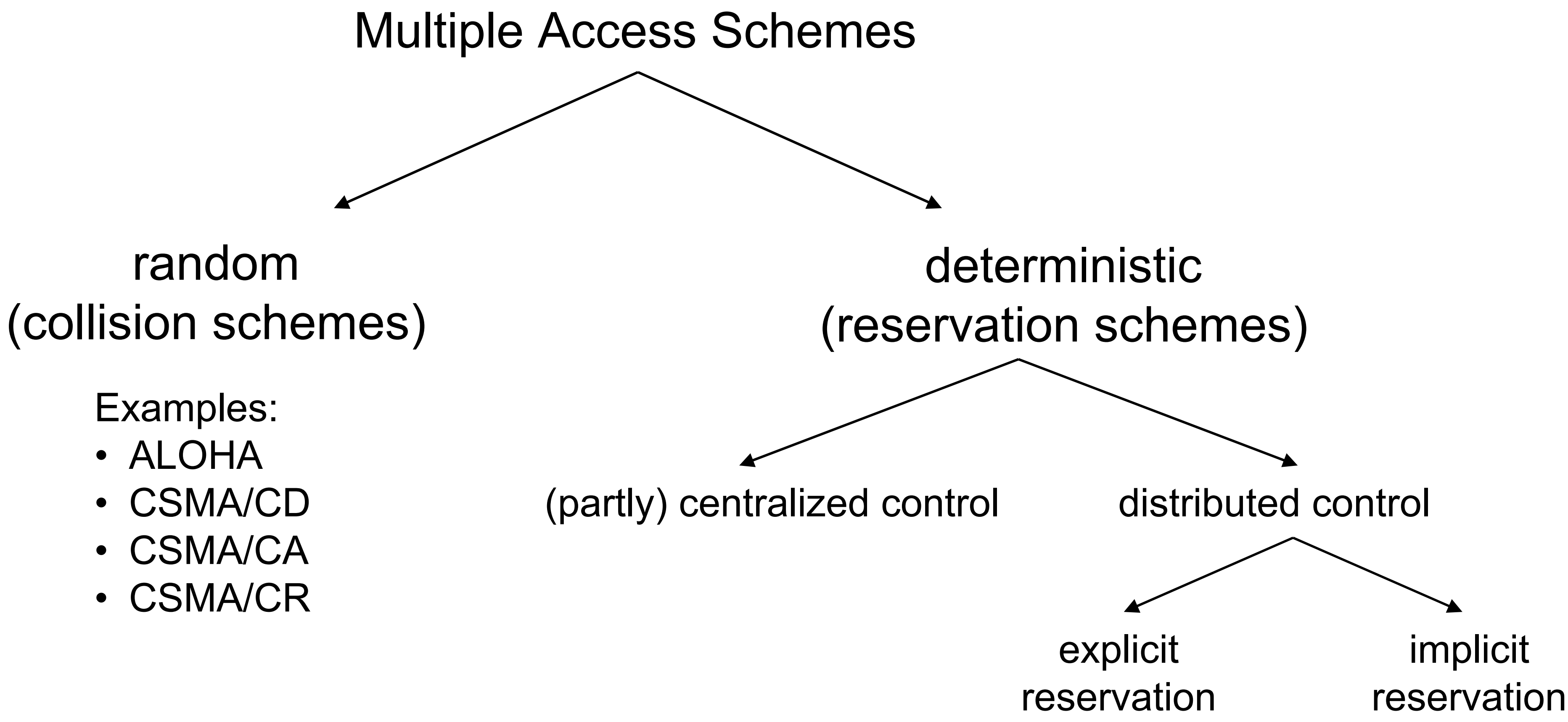
Interleaving

- Example:



Multiple Access Schemes

Multiple Access Schemes - Overview



Deterministic vs. Random Access

- Deterministic (fixed assignment) Access:
  - typically used in voice-oriented networks
  - efficient use of communication resources when each user has a steady flow of information to be transmitted
  - access is coordinated by dedicated signaling channels that exchange short signaling messages to obtain resources in the network at the beginning of the conversation and to release resources at the end
  - these access methods assign a time slot, a frequency channel, or a specific code to the user (preferably for the entire length of the conversation)
- Random Access:
  - typically used in data-oriented networks
  - provides a more flexible and efficient way of managing channel access for transmitting short, bursty messages
  - each packet of the data flow has to carry full information related to the destination and the source
  - provides the user with varying degrees of freedom in gaining access to the network whenever information is to be sent
  - randomness of user access may cause contention among the users for access to a channel, resulting in collisions of contenting transmissions

(Radio) Channel Access Schemes in Wireless Networks

- Frequency Division Multiple Access (FDMA)
  - example: C-Netz
- Time Division Multiple Access (TDMA)
  - example: DECT
- Combination of FDMA and TDMA
  - example: GSM
- Combination of Orthogonal FDMA (OFDMA) and TDMA
  - example: LTE/SAE
- Code Division Multiple Access (CDMA)
  - example: UMTS, UTRAN-FDD Mode
- Combination of CDMA and TDMA
  - example: UMTS, UTRAN-TDD Mode
- Space Division Multiple Access
  - only useful in combination with other schemes
  - example: multiple usage of radio resources through beamforming