Radio Access and Link Control in Cellular Networks

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August 2015

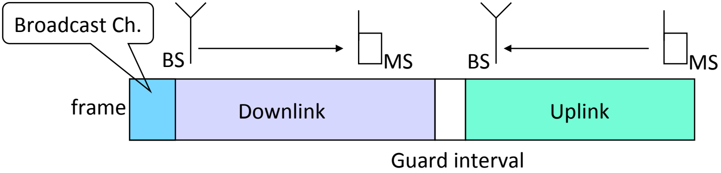
Outline I

Duplexing schemes

- Uplink : communications from the Mobile Station (MS) to the Base Station (BS)
- Downlink : communications from BS to MS
- Duplexing : the way uplink and downlink are allocated to orthogonal radio resources
- Two main approaches : TDD and FDD.

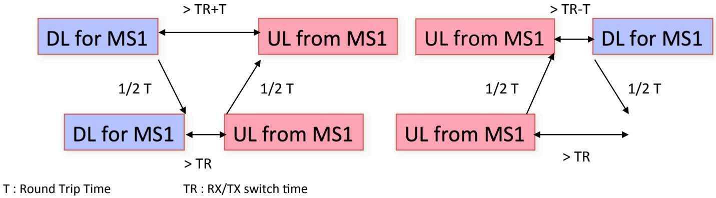
TDD I

- Time is divided in frames and every frame in two sub-frames
- UL and DL use the same carrier frequency during different sub-frames
- Guard interval between UL and DL sub-frames to maintain orthogonality bw links despite propagation delays
- Synchronization is needed bw UL and DL



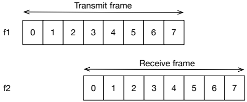
TDD II

- No need for stations to support full-duplex
- TDD systems can be deployed on non-peered spectrum bands
- Channels characteristics are similar on both links : open loop channel estimation is possible (and useful for MIMO, beamforming and power control)
- TDD is more adapted to small cells because of the small propagation time

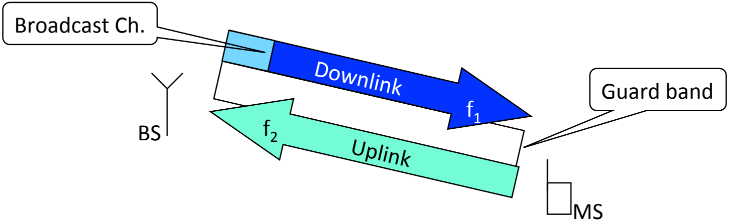


FDD

- UL and DL use different carrier frequencies
- In duplex mode, RX and TX can be simultaneous
- In half-duplex, RX and TX cannot be simultaneous (e.g. in GSM)

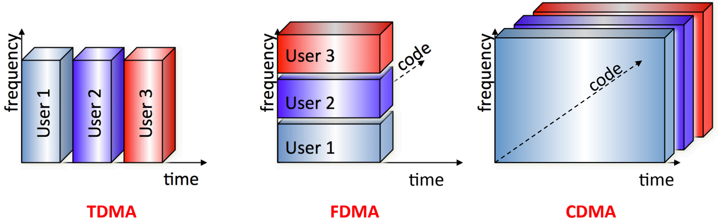


- Two peered frequency bands are usually needed
- Time synchronization is not needed in duplex mode
- Channel conditions on UL and DL are independent



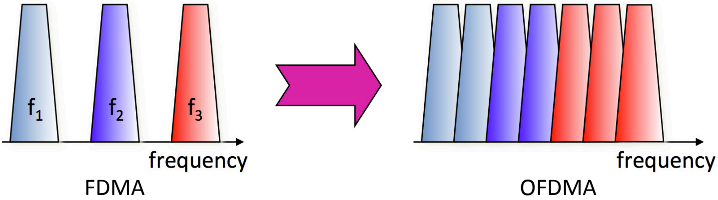
Multiplexing schemes

- Multiplexing schemes allocated different radio resources to different users sharing a common link (UL or DL)
- Signal space can be divided in time, frequency or codes to obtain TDMA, FDMA or CDMA (Time/Frequency/Code Multiple Access)



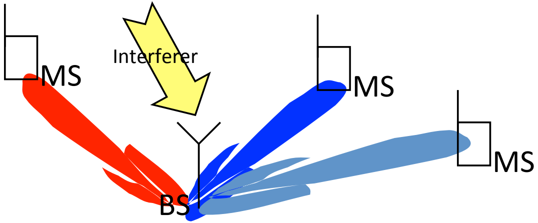
OFDMA

- OFDMA (Orthogonal Frequency Division Multiple Access) can be seen as a special case of FDMA
- Every elementary frequency band is called a *sub-carrier*
- Main advantages : ease of implementation, better spectral efficiency, easy egalization, flexible resource allocation
- Main drawbacks : high peak to average power ratio, inter-carrier interference



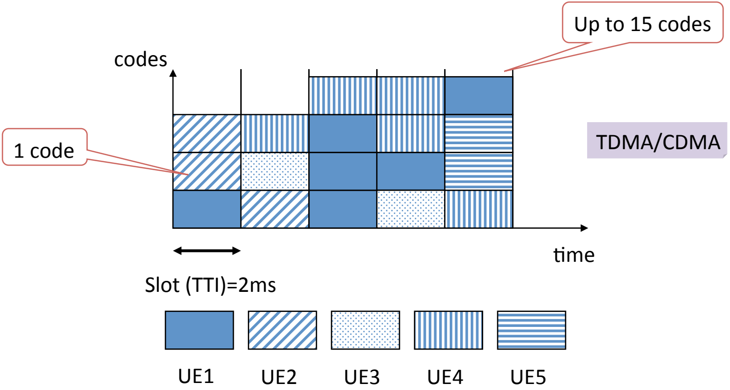
SDMA

- The use of beamforming or Multi-user MIMO adds a new dimension : Space Division Multiple Access (SDMA)



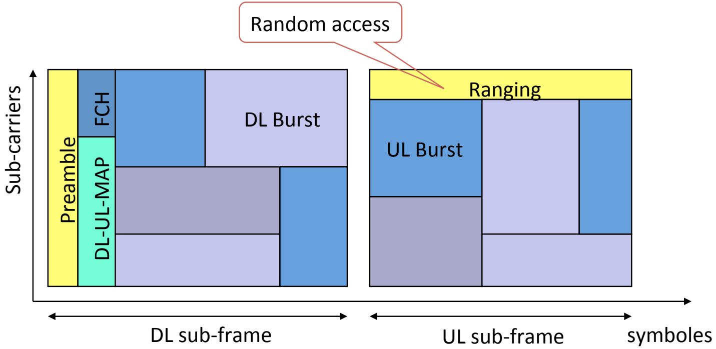
Examples I

- In HSPA, a data channel is slotted and shared by all DL User Equipments (UE). Several UEs can receive data on the same slot (or Transmission Time Interval – TTI) but encoded with different orthogonal codes



Examples II

- In WiMAX there are two degrees of freedom for multiplexing : sub-carriers (of an OFDM symbol) and time (OFDM symbols)
- Duplexing scheme is TDD

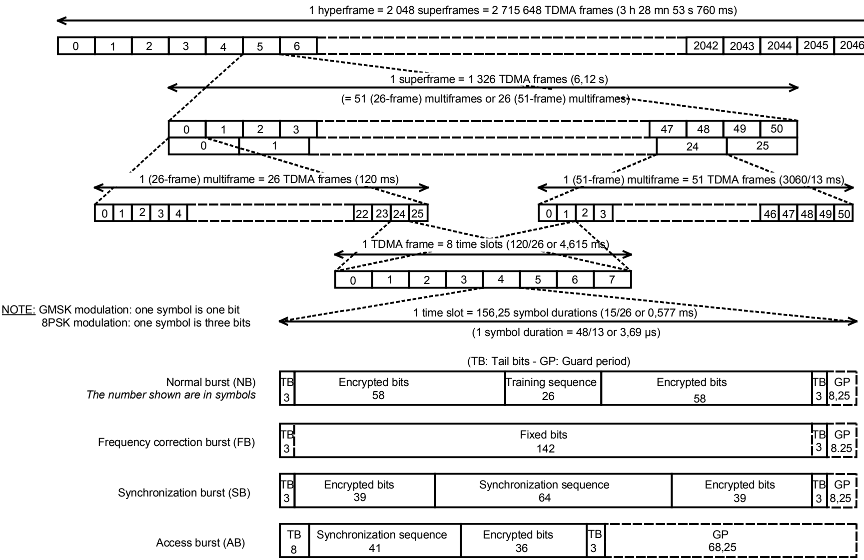


TDMA/OFDMA

Examples III

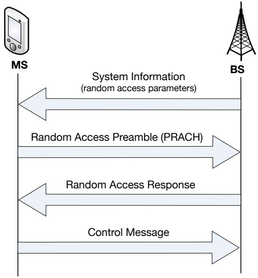
- In GSM, operator spectrum is divided in 200 KHz channels, each frequency channel is divided in 8 slots. Duplexing scheme is FDD.
- Frames are grouped together to form multiframes (26 or 51 frames). Multiframes are grouped in superframes and super frames in hyperframes.

Examples IV


1 hyperframe = 2 048 superframes = 2 715 648 TDMA frames (3 h 28 mn 53 s 760 ms)
0 1 2 3 4 5 6 2042 2043 2044 2045 2046 2047
1 superframe = 1 326 TDMA frames (6,12 s)
(= 51 (26-frame) multiframes or 26 (51-frame) multiframes)
0 1 2 3 47 48 49 50
0 1 24 25
1 (26-frame) multiframe = 26 TDMA frames (120 ms)
1 (51-frame) multiframe = 51 TDMA frames (3060/13 ms)
0 1 2 3 4 22 23 24 25
0 1 2 3 46 47 48 49 50
1 TDMA frame = 8 time slots (120/26 or 4,615 ms)
0 1 2 3 4 5 6 7
NOTE: GMSK modulation: one symbol is one bit
8PSK modulation: one symbol is three bits
1 time slot = 156,25 symbol durations (15/26 or 0,577 ms)
(1 symbol duration = 48/13 or 3,69 µs)
(TB: Tail bits - GP: Guard period)
Normal burst (NB)
The number shown are in symbols
TB 3
Encrypted bits 58
Training sequence 26
Encrypted bits 58
TB 3
GP 8,25
Frequency correction burst (FB)
TB 3
Fixed bits 142
TB 3
GP 8.25
Synchronization burst (SB)
TB 3
Encrypted bits 39
Synchronization sequence 64
Encrypted bits 39
TB 3
GP 8,25
Access burst (AB)
TB 8
Synchronization sequence 41
Encrypted bits 36
TB 3
GP 68,25

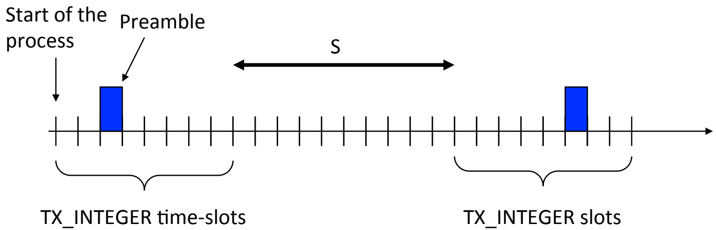
Random access procedure

1. the MS acquires random access parameters from the broadcast channel (position of the PRACH, protocol parameters, transmit power, etc)
2. the MS transmits a short *burst* or *preamble* using slotted Aloha. Transmissions are subject to collisions and capture.
3. in case of collision, the MS wait for a random duration (back off) and retransmit the preamble
4. the BS sends a random access response that includes at least the timing advance and an uplink allocation
5. the MS sends a control message on the allocated resource with at least the reason for performing random access (mobile originated call, response to a paging, location update, etc)



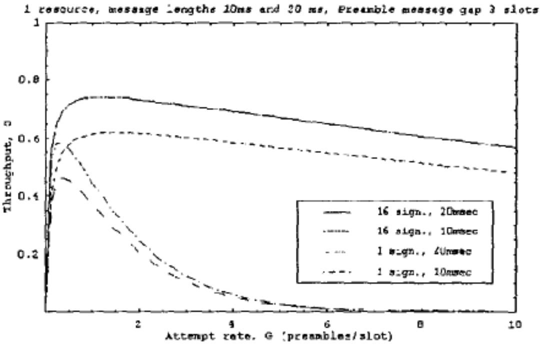
Random access protocol I

- All random access protocols are based on slotted ALOHA
- Example of GSM : TX_INTEGER is the back-off window (e.g. bw 8 and 50 slots), S is the minimum delay before retransmissions (e.g. 250 ms), there is a maximum of MAX_TRANS retransmissions
- All parameters are broadcast by the Broadcast Control Channel (in a System Information Block)
- In GSM : the preamble includes a service class and a random number to cope with the capture effect



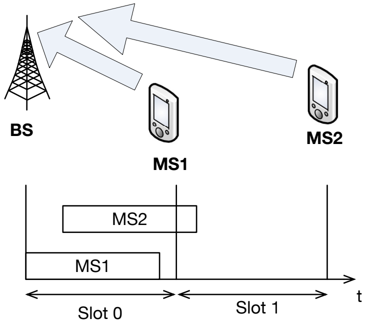
Random access protocol II

- In UMTS and in LTE, the User Equipement (UE) chooses also a random signature (a scrambling code in UMTS, a Zadoff-Chu sequence in LTE)
- UMTS example (source : I. N. Vukovic and T. Brown, "Performance Analysis of the Random Access Channel in WCDMA", IEEE VTC'01) :



Timing Advance I

- Goal : synchronization of UL signals in order to maintain orthogonality between MSs of a cell
- After random access, the BS transmits a timing offset for all subsequent uplink transmissions of the MS. Timing advance is regularly updated.



Timing Advance II

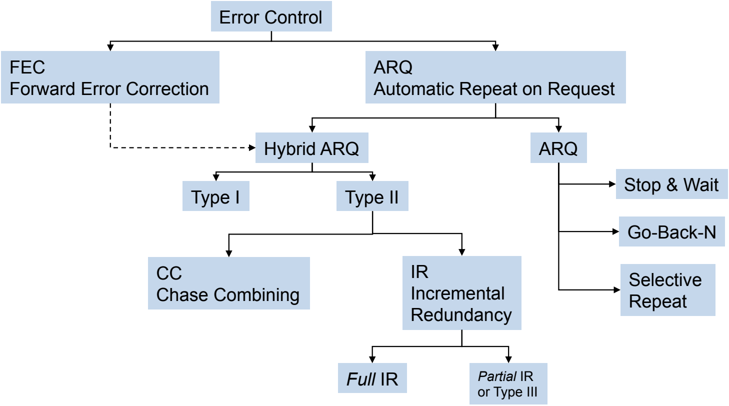
Example in GSM :

- Slot duration = 0.577 ms = 156.25 symbol durations
- An access burst = 88 symbols, so that 68 symbol durations are left free to cater with the transmission of a MS that does not know its timing advance
- Timing advance can take values between 0 and 63 (symbol durations)
- Every TA step represents 550 m, max cell range is 63×550 m = 35 Km
- After successfully receiving an access burst on the RACH, the network sends a response on the Access Grant Channel (AGCH), which contains the TA parameter.

Example in LTE :

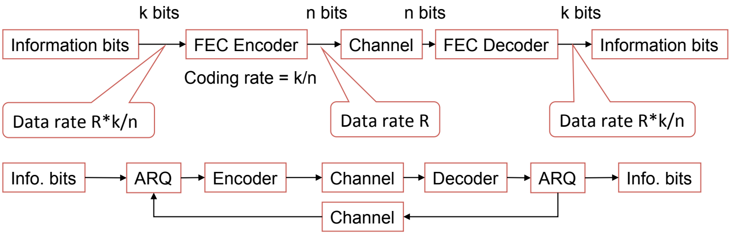
- TA command in Random Access Response is coded on 11 bits, TA indices span between 0 and 1282
- Time adjustment is a multiple of $16T_S$, where $T_S = 1/(2048 \times 15000) = 1/30720000$ s is the sample duration
- Maximum timing adjustment is $1282 \times 16T_S$, which corresponds to a distance of 100 Km

Classification



FEC vs ARQ

- FEC : the transmitter adds redundant information to the useful information and the receiver exploits this redundancy to decode the information bits. E.g. : convolutional codes, LDPC, turbo codes, etc. No need for a feedback channel. There is a data rate vs error correction power trade-off.
- ARQ : based on an error detection code, the receiver sends back an ACK/NACK upon reception of a packet (NB : ARQ can be also implemented at layer 4, cf TCP)

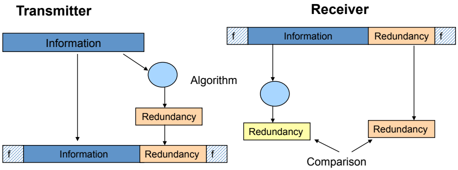


Error detection

- For every ARQ protocol, there is a need for an error detection scheme (source of the figure : [1])
- Cyclic Redundancy Codes (CRC) are usually used for error detection with very good performances (see [18])
- For a rate k/n, a bound on the probability to not detect an error is :

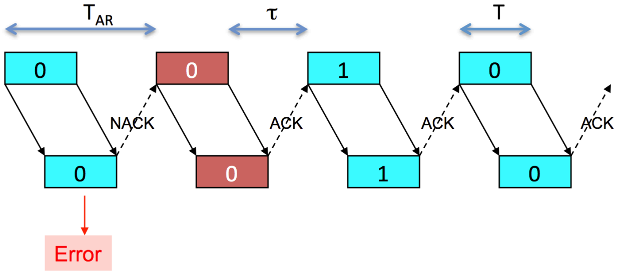
$$P_e < 2^{k-n}$$

- Proof : If the bit error probability is $\epsilon = 1/2$, the 2^n n-tuples that are likely to be received with the same probability 2^{-n}. Among these tuples, $2^k - 1$ are recognized as code words and still different from the transmitted code words. We thus have : $P_e = (2^k - 1)2^{-n} = 2^{k-n} - 2^{-n} < 2^{k-n}$.



Stop and Wait I

- Frames are numbered modulo 2
- Let n be the number of bits, R the data rate and τ the idle time
- Let $T = \frac{n}{R}$ be the transmission time
- Let $T_{AR} = T + \tau$ be the round trip time.



Stop and Wait II

- Consider the Binary Symmetric Channel (BSC) with error probability ϵ
- Let P_c be the probability that a n bit frame has no error.
- Let P_d be the probability that a n bit frame has errors that **can** be detected by the receiver.
- Let P_e be the probability that a n bit frame has errors that **cannot** be detected by the receiver.

$$P_c + P_d + P_e = 1$$

- Let R be the data rate.

Stop and Wait III

- The probability to deliver to higher layers a frame with undetected errors is very small :

$$P(E) = \sum_{i=0}^{\infty} P_e P_d^i = \frac{P_e}{1 - P_d}.$$

It is supposed to be negligible.

- Assume error on bits are independent, we have :

$$P_c = (1 - \epsilon)^n$$

- We have a bound for P_e :

$$P_e < 2^{k-n}$$

- Consider a BCH code with $k = 2014$ and $n = 34$, then $P(E) = 8.10^{-10}$.

Stop and Wait IV

- Average delay to deliver a frame with Stop and Wait :
  (assuming P_e is negligible)

$$\delta_{SW} = \sum_{i=1}^{\infty} iT_{AR}P_c(1-P_c)^{i-1} = \frac{T_{AR}}{P_c} = \frac{\frac{n}{R}+\tau}{1-P_d}$$

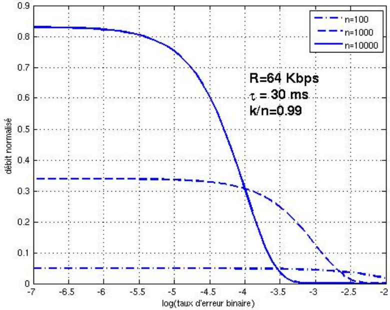
- Goodput of Stop and Wait :

$$\eta_{SW} = \frac{k}{\delta_{SW}} = \frac{1}{1+\frac{\tau R}{n}}\frac{k}{n}R(1-P_d)$$

- Example : $R = 1$ Mbps, $\tau = 700$ ms (satellite link), $\tau R = 7 \cdot 10^5$. In order that $\tau R/n$ be close to 1, we need $n \approx 10^6$ bits. With $n = 10000$ bits, we have already $\tau R/n = 70$.

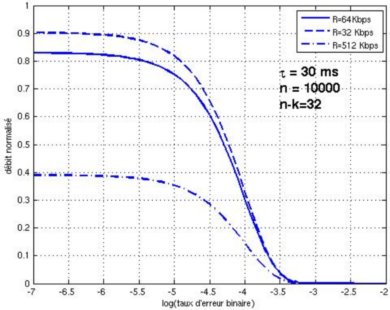
Stop and Wait V

- Goodput as a function of the (log of the) bit error rate
- When n increases, τ becomes small in front of $T = n/R$.



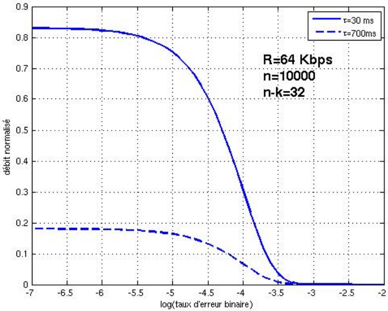
Stop and Wait VI

- When R increases, T becomes small in front of τ.



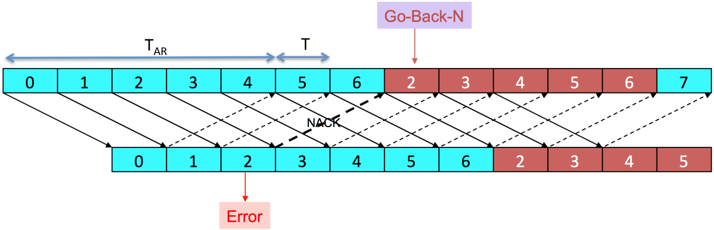
Stop and Wait VII

- When τ increases, idle time increases.



Go Back N I

- Anticipation window : the transmitter is allowed to transmit up to W frames without receiving corresponding ACKs.
- For a window of W frames, frames have to be numbered modulo $W+1$.
- We assume below that the window size is ideal, i.e., $W = T_{AR}/T$.



Go Back N II

- Average delay to deliver a frame with Go Back N :

$$
\begin{aligned}
\delta_{GBN} &= \sum_{i=1}^{\infty} T((i-1)W+1)P_c(1-P_c)^{i-1} \\
&= TP_c(W\sum_{i=1}^{\infty} i(1-P_c)^i + \sum_{i=1}^{\infty}(1-P_c)^{i-1}) \\
&= TP_c\left(\frac{W(1-P_c)}{P_c^2} + \frac{1}{P_c}\right) \\
&= T\left(1 + W\frac{1-P_c}{P_c}\right)
\end{aligned}
$$

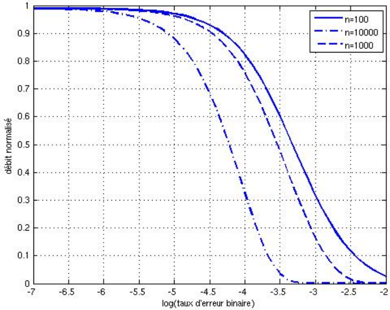
- Goodput of Go back N :

$$
\eta_{GBN} = \frac{1}{1+P_d(W-1)}\frac{k}{n}R(1-P_d)
$$

- If R and τ are small, W can be set to a small value and the good put of Go Back N is good. If R is high and the propagation time is long, the denominator is high and the goodput is low.

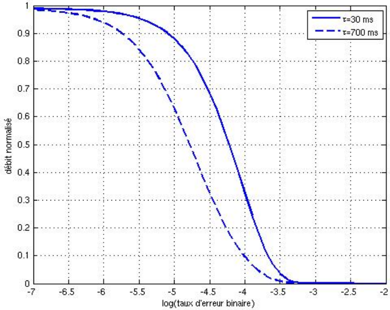
Go Back N III

- When n increases, P_d and W increase.



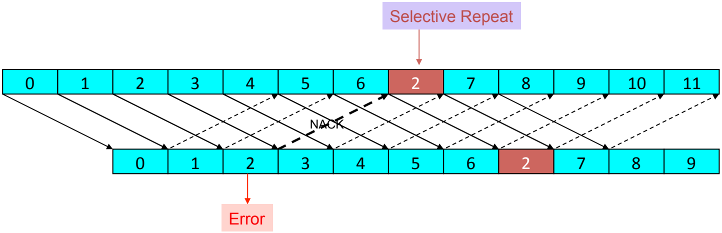
Go Back N IV

- When τ increases, W increases.



Selective Repeat I

- The transmitter retransmits only the frames for which it received a NACK
- For a window of W frames, frames have to be numbered modulo $2W$.



Selective Repeat II

- Average delay to deliver a frame with Selective Repeat :

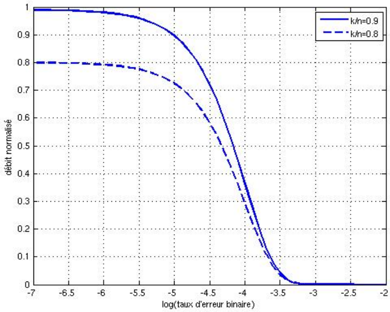
$$\begin{aligned} \delta_{SR} &= \sum_{i=1}^{\infty} iTP_c(1-P_c)^{i-1} \\ &= \frac{T}{P_c} \end{aligned} \tag{1}$$

- Goodput of Selective Repeat :

$$\eta_{SR} = \frac{k}{n} R(1 - P_d)$$

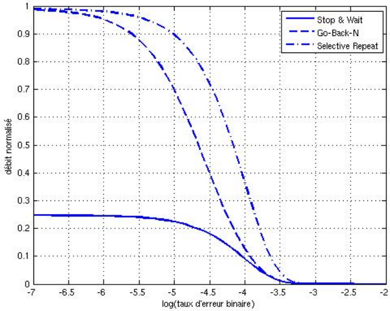
Selective Repeat III

- Impact of k/n.



Selective Repeat IV

- Comparison in terms of goodput.



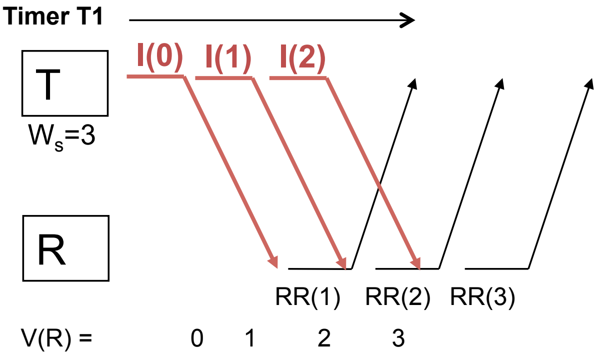
Example : LAP-B I

- Used in X25 but very close to LAP-Dm (used in GSM for control channels) and UMTS/LTE RLC Acknowledge Mode
- Every frame has a sequence number indicated in its header, numbering is done modulo $M \in \{2, 128, 32768\}$
- A connection-oriented protocol : transmitter and receiver sides maintain counters that are initialized when connection is open. After transmissions, connection is closed

- Go-Back-N : the transmitter can send up to W_s frames without receiving corresponding ACKs
- Ready to Receive (RR) : may acknowledge several frames
- After a Reject (REJ) : all frames from the requested SN are retransmitted
- The receiver maintains a counter of the next SN to be received V(R)

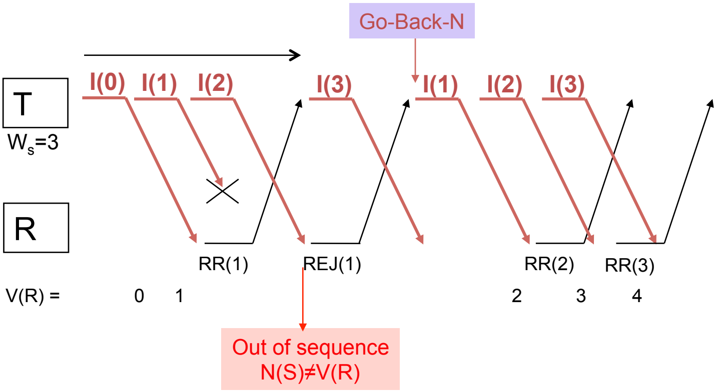
Example : LAP-B II

- Information frames (I) includes N(S) = its sequence number
- RR includes N(R) = sequence number of the next expected frame
- V(R) = receiver counter indicating the next expected frame
- Every transmission is associated to a timer T1



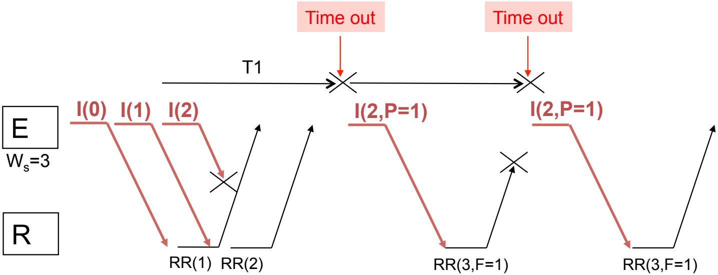
Example : LAP-B III

- Retransmission upon negative acknowledgement (REJ)



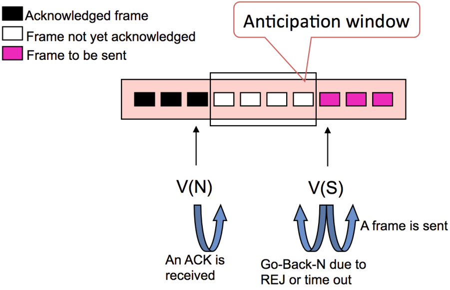
Example : LAP-B IV

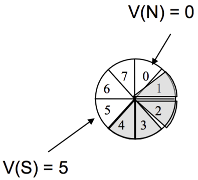
- Retransmission upon T1 time out
- P/F bit : an immediate response is required / immediate response
- There is a maximum number of retransmissions



Example : LAP-B V

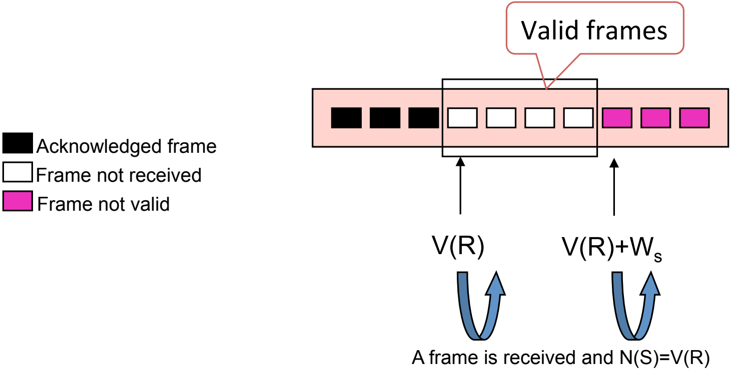
- Transmitter side :
  - V(N) = last acknowledged frame
  - V(S) = next frame to send
  - NB : $V(S) - V(N) - 1 \leq W_s$





Example : LAP-B VI

- Receiver side :
  - V(R) = next expected frame

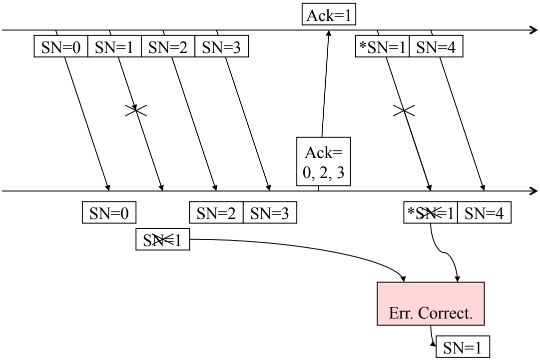


HARQ Types

- Type I : FEC and ARQ are well separated
- Type II : the receiver keeps a copy of packets in error and use the information to decode subsequent packets
- Type II with Chase Combining (CC) : all retransmissions are identical
- Type II with Incremental Redundancy (IR) : the coding scheme is modified at each retransmission of the same packet
- Type II Partial IR : every transmission includes information bits + parity bits
- Type II Full IR : the first transmission includes information bits + parity bits, retransmissions includes only parity bits

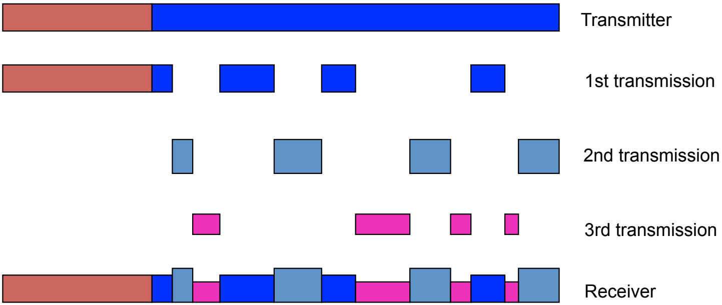
HARQ Type II Illustrated

- On this example, we can obtain a correct copy of packet with Sequence Number SN=1 from two erroneous receptions
- With CC, the two transmissions of (SN=1) are identical. With IR, the second transmission is re-encoded



HARQ Full IR Illustrated

- In IR, information is first coded with a low rate code at the transmitter. Information bits and some parity bits are sent at the first transmission
- If a NACK is received, the transmitter sends additional parity bits. The receiver aggregate received bits to those already received
- At every reception, the effective coding rate is decreased
- In Full IR, information is sent only at the first transmission. In Partial IR, information is sent at every transmission



Chase Combining

To go in more details...

- Main idea : if the initial FEC coding rate is R, then after L transmissions, the effective rate is R/L
- Let $Y_i = (Y_{i,1}, Y_{i,2}, ..., Y_{i,N})$ be the i-th transmitted packet (made of N bits
- After the second transmission, transmitted bits can be written :

$$(Y_{1,1}, Y_{2,1}, Y_{1,2}, Y_{2,2}, ..., Y_{1,N}, Y_{2,N})$$

  which can be seen as the superposition of the initial FEC and a repetition code. There are $N \times R$ information bits and the packet length is $2N$, so that the new code rate is $NR/(2N) = R/2$
- Decoding can be done using maximum likelihood and Viterbi algorithm
- Maximum likelihood : we look for the sequence s that maximizes $P[s|r]$, where r is the received sequence

Incremental Redundancy I

To go in more details...

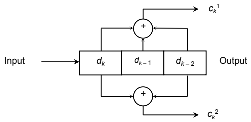
- Main idea : IR is based on a sequence of codes obtained by increasingly puncturing a mother code
- Puncturing : remove periodically some bits from the encoder output (increases code rate and decreases decoding complexity)
- Example : Take a code with rate $R_1 = 1/2$:

$$C_1 = (c_0(1), c_0(2), c_1(1), c_1(2), \dots)$$

We obtain a code of rate $R_2 = 2/3$ by puncturing one bit out of 4 :

$$C_2 = (c_0(1), c_0(2), c_1(1), -, c_2(1), c_2(2), c_3(1), -, \dots)$$

We obtain a code of rate $R_3 = 4/5$ by puncturing 3 bits out of 8, etc



Incremental Redundancy II

- Rate Compatible Punctured Convolutional Codes (RCPC) :
  - A family of codes $\{C_n\}$ is obtained by puncturing a low rate mother code with an increasing number of bits, so that $R_n \leq R_{n+1}$
  - Punctured bits to obtain C_n are included in the set of punctured bits to obtain C_{n+1}
- Example :
  0 bit punctured (mother code) :

$$C_1 = (c_0(1), c_0(2), c_1(1), c_1(2), \dots), \quad R_1 = 1/2$$

  1 bit punctured out of 8 :

$$C_2 = (c_0(1), c_0(2), c_1(1), c_1(2), c_2(1), -, c_3(1), c_3(2), \dots), \quad R_2 = 4/7$$

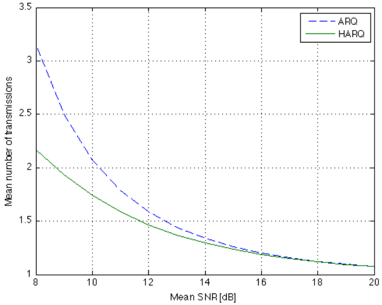
  2 bits punctured out of 8 :

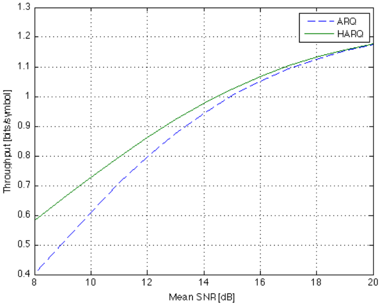
$$C_3 = (c_0(1), c_0(2), c_1(1), c_1(2), c_2(1), -, -, c_3(2), \dots), \quad R_3 = 4/6$$

- C_n offers a higher protection against errors than C_{n+1}, however with a lower useful data rate

HARQ vs ARQ

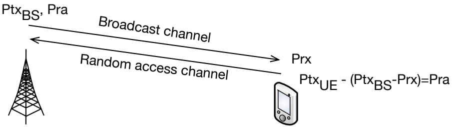
- Performance comparison based on [?] (MCS4, 16-QAM, rate=9/16, 2.25 bits/symbol, $g = 0.664$, $\gamma_M = 7.7$ dB)



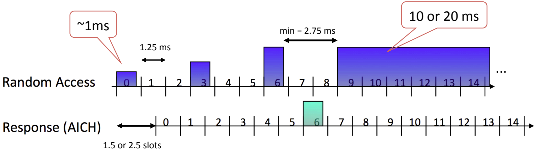


Open Loop Power Control

- Open loop power control : based on BS transmit power (Ptx_{BS}), target received power (Pra) and received power (Prx), the UE sets its transmit power for random access (Ptx_{UE})

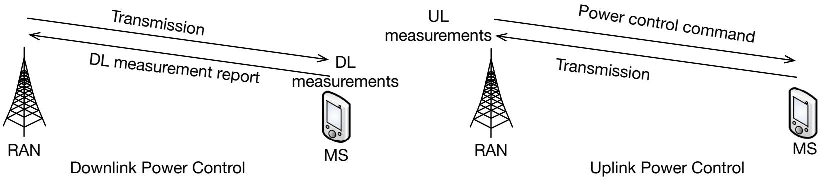


- Power ramping : at each retransmission, transmit power is increased (typically +1 dB) up to some maximum value. Example of UMTS :



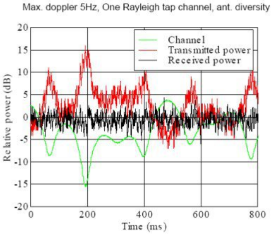
Closed Loop Power control

- DL : based on DL transmissions, the MS makes DL measurements (signal level and quality) and feedbacks them in a measurement report. The RAN adjusts its transmit power
- UL : based on UL transmissions, the RAN performs UL measurements and sends power control command to the MS, which adjusts its power



Slow and Fast Power Control

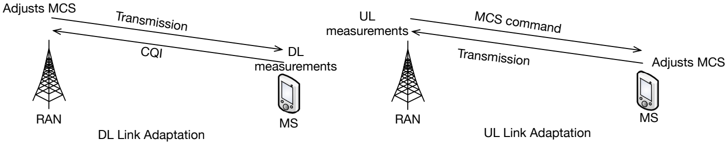
- Slow power control follows the slow variations of the radio channel (due to shadowing). Example in GSM :
    - Measurements reports and PC commands are sent every 480 ms on the SACCH
    - Upon reception of a command, a MS change its transmit power by steps of 2 dB every 60 ms until reaching the target transmit power
- Fast power control follows the fast variations of the channel (due to fast fading). Example in UMTS : Power control rate is every 666 μs (1500 Hz)



Source : mobilewireless.wordpress.com

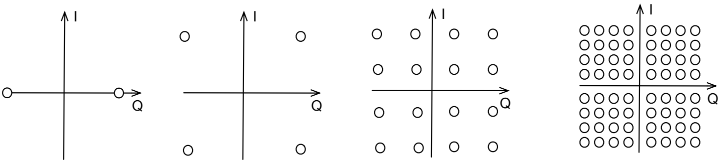
Link Adaptation

- A technique to dynamically adjust modulation and coding rate according to the radio channel variations
- Adjustment is based on measurements. DL : the MS makes channel measurements and feeds back a Channel Quality Indicator (CQI), the network adjusts the Modulation and Coding Scheme (MCS). UL : the network makes measurements and sends a MCS command to the MS, which adjusts its transmission



Modulations and Coding Rate

- Traditional modulations used in cellular networks : BPSK, QPSK, GMSK (GSM), 16-QAM, 64-QAM
- Denser is the modulation, the more sensitive it is to noise and the higher the number of bits per symbol

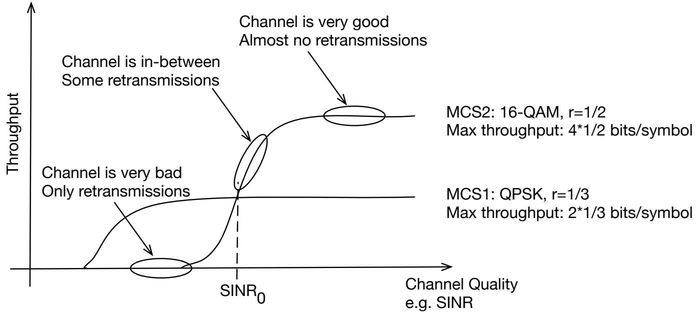


BPSK (1 bit/symbol) QPSK (2 bits/symbol) 16-QAM (4 bits/symbol) 64-QAM (6 bits/symbol)

- A FEC encoder takes k bits as input and outputs $n \geq k$ bits. Coding rate is $r = \frac{k}{n}$. For a data rate R, information rate is rR
- When $r \to 1$: few redundancy bits, less error correction, high information rate. When $r \to 0$: many redundancy bits, more error correction, low information rate.

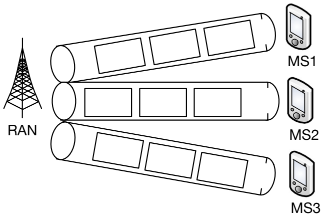
Example of Algorithm

- If SINR $\leq$ SINR$_0$, then choose MCS1
- If SINR $>$ SINR$_0$, then choose MCS2

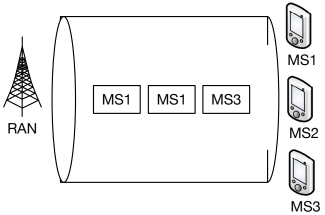


Dedicated vs. Shared Channel

- Dedicated channel (GSM, UMTS R'99) : the traffic of a user is transmitted over dedicated resources (slot, frequency, code) allocated for the whole connection duration. No need to address frames. Channel reconfiguration may be performed by RRC.
- Shared channel (HSPA, LTE) : the whole user traffic is transmitted over a single channel. Allocation is done by the scheduling algorithm at a very short time scale. Every frame includes an address.



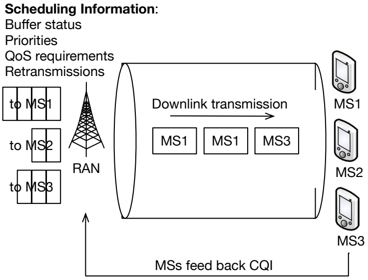
Dedicated channels



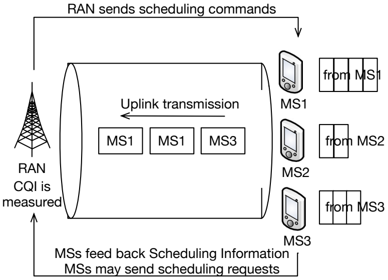
Shared channel

One-to-many vs. Many-to-One

- One-to-many for the DL : scheduling information is available at the scheduler, CQI is fed back by MSs
- Many-to-one for the UL : scheduling is still controlled by the network through scheduling commands. Scheduling information is fed back from the MSs. CQI is measured on UL signals. If a MS is not known from the scheduler, this MS can send some scheduling request.



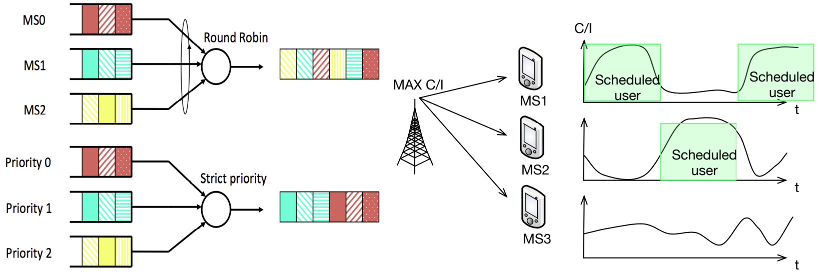
One-to-many scheduling



Many-to-one scheduling

Examples of Scheduling Algorithms

- Round Robin : fairness in resource
- Strict priority : as long as there is traffic for a higher priority queue, it is served
- Max C/I : the MS with the best instantaneous channel is served
- Proportional fair scheduler : the scheduler allocated the resource to the MS with the best $\frac{r_i(t)}{\bar{R}_i}$ ratio, where $r_i(t)$ is the instantaneous data rate and $\bar{R}_i$ is the average throughput



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