



938II - Electronics and communication technologies (2024/25)

Basics of wireless communication systems

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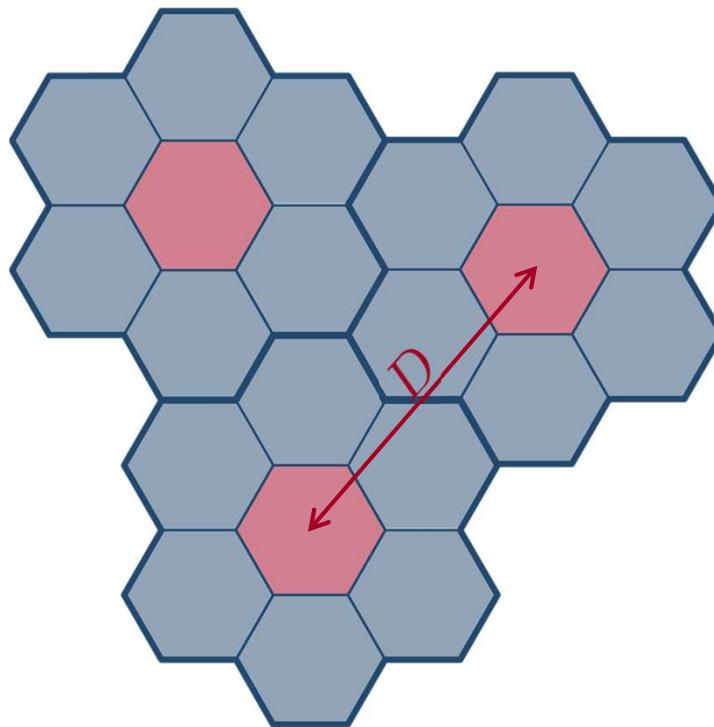
Master's Degree in Cybersecurity [WCY-LM]



Planning of a cellular network

Reuse distance (1/2)

A fundamental parameter which impacts the performance of a cellular network is the **reuse distance** D :



Reuse distance (2/2)

In the attempt to calculate the reuse distance D , we need to better characterize the cluster size K

Experimentally, the only acceptable K 's are those fulfilling

$$K = i^2 + j^2 + i \cdot j, \quad \text{with } i, j \in \mathbb{N}, i + j \neq 0$$

i	0	0	0	1	1	1	1	2	2	2	3
j	1	2	3	1	2	3	4	2	3	4	3
K	1	4	9	3	7	13	21	12	19	28	27
	$\geq 3G$		$2G$				$1G$				



Downlink SIR (1/2)

Since we are considering the worst-case scenario,

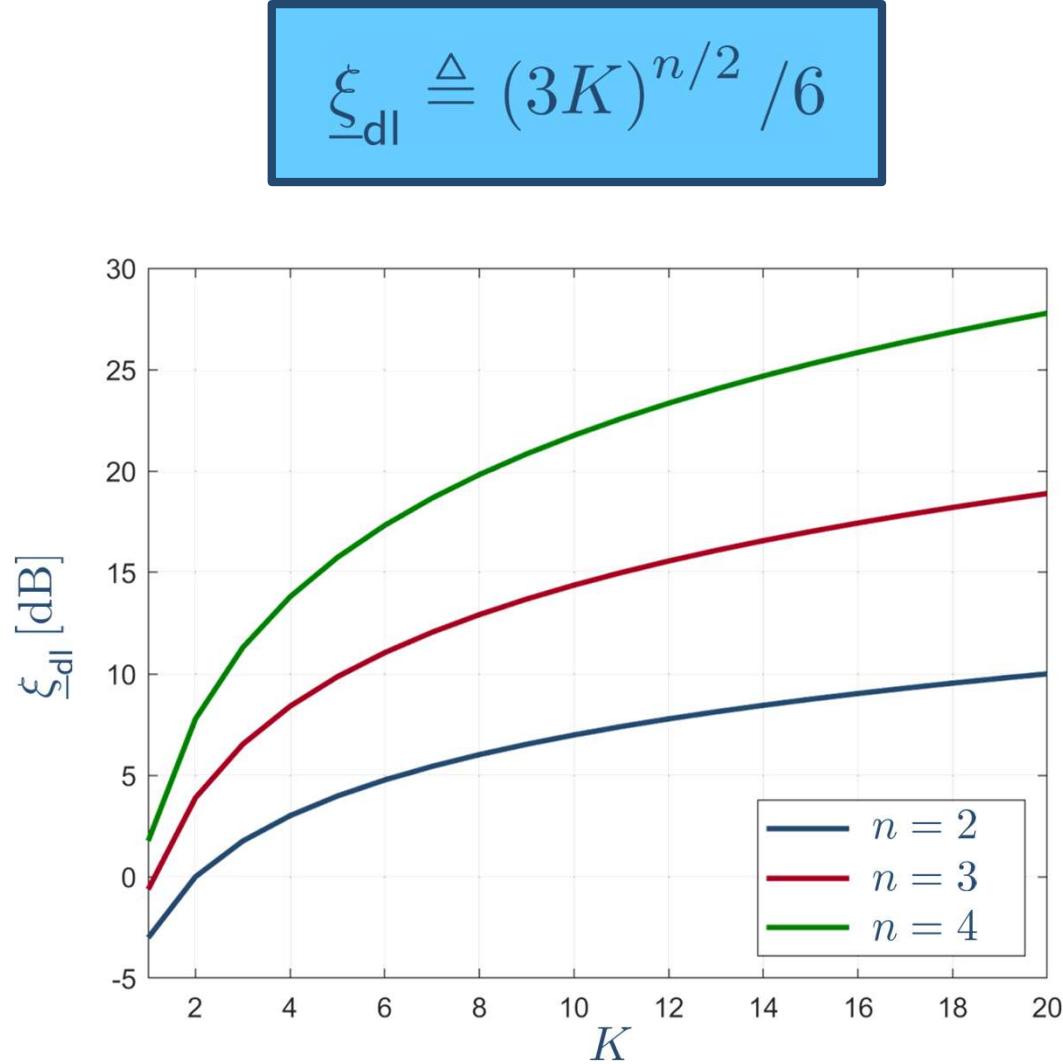
$$\xi \geq \frac{C}{\sum_{k=1}^6 I_k} = \frac{(D/R)^n}{6}$$

Considering that $D = \sqrt{3K}R$

$$\xi \geq \underline{\xi}_{\text{dl}} \triangleq \frac{(3K)^{n/2}}{6}$$

The larger the SIR, the better the performance: hence, increasing K is beneficial in terms of network performance in the downlink

Downlink SIR (2/2)



Summary of the main tradeoffs

To sum up, designing a cellular network is a **cumbersome task**, even when we take the following simplifications:

Degrees of freedom:

- **Reuse factor** K
- **Cell radius** R

System KPIs:

- **Transmit power** P_T
- **Handoff rate** μ_H
- **User density** u
- **Minimum SIR** ξ

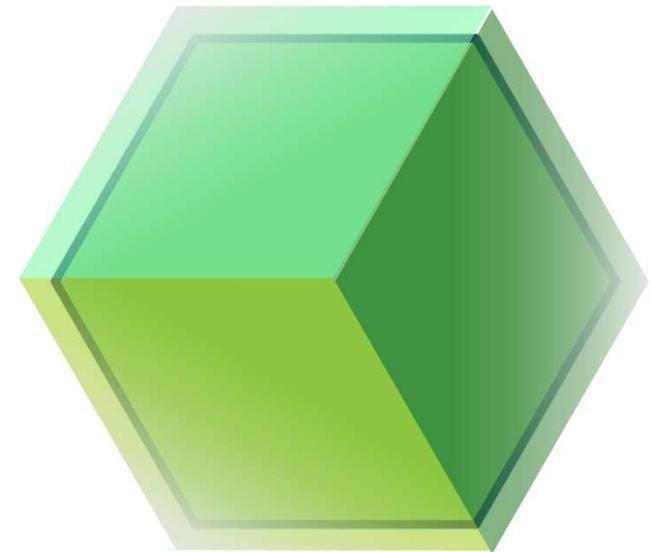
KPI	$R \uparrow$	$K \uparrow$
P_T	↑ (✗)	↔
μ_H	↓ (✓)	↑ (✗)
u	↓ (✗)	↓ (✗)
ξ	↔	↑ (✓)

Improving the downlink SIR* (1/3)

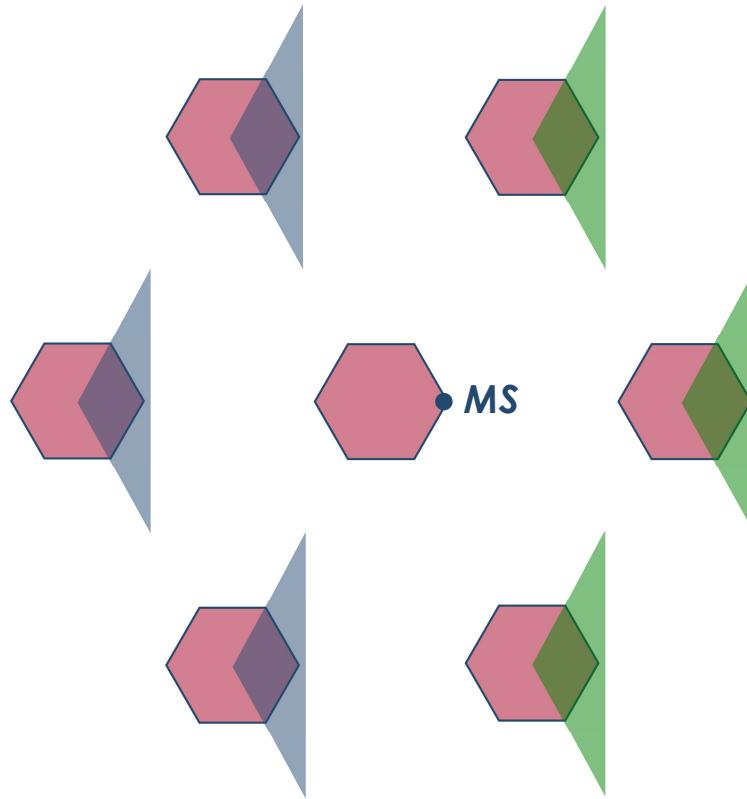
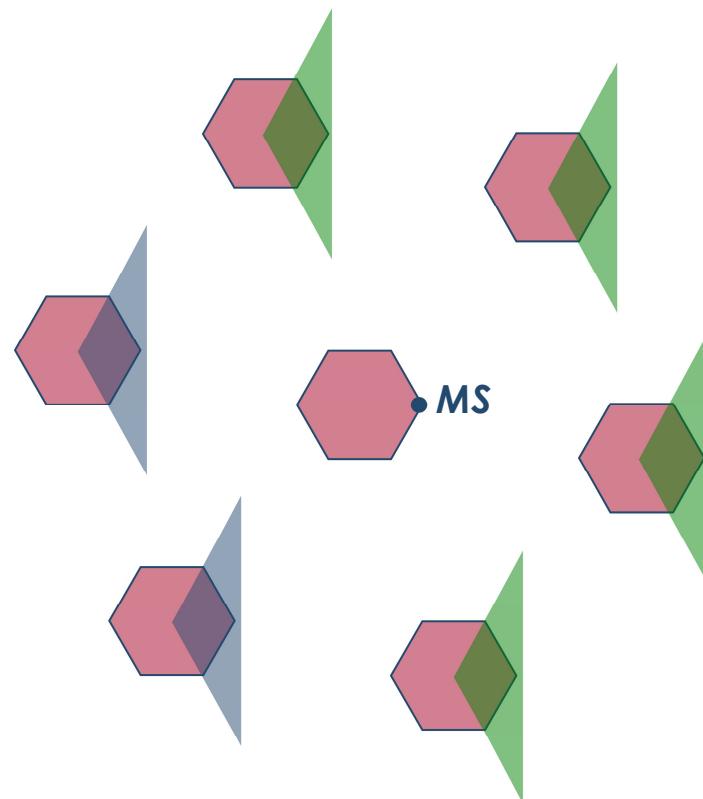
Many techniques exist to further **mitigate** the co-channel interference in the downlink

One of the simplest techniques is using **cell sectoring** at the BTS

Example: 120°
antenna aperture



Improving the downlink SIR* (2/3)

2× SIR increase**3× SIR increase****Drawbacks: increase in the number of antennas and handoffs**

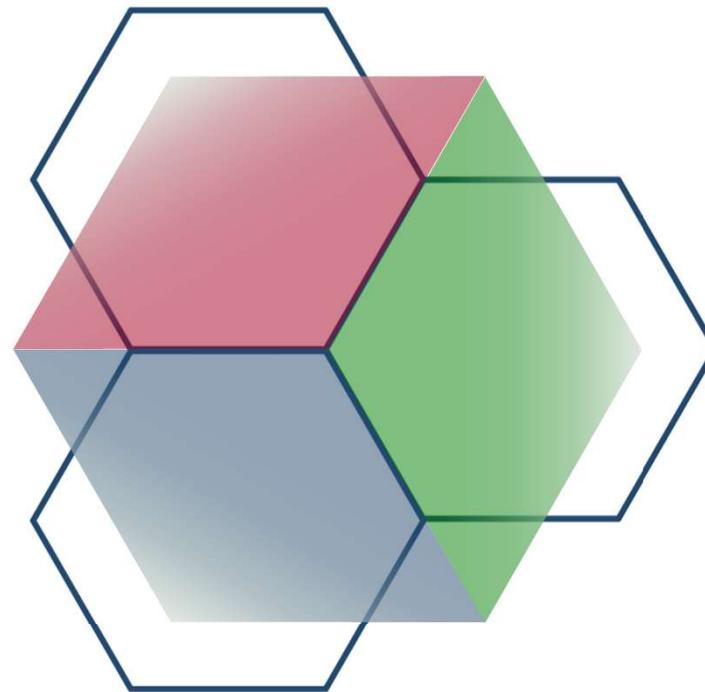
Improving the downlink SIR* (3/3)

The rationale of cell sectoring can be **extended to cover full cells: this leads to tri-cellular sites**

Example:

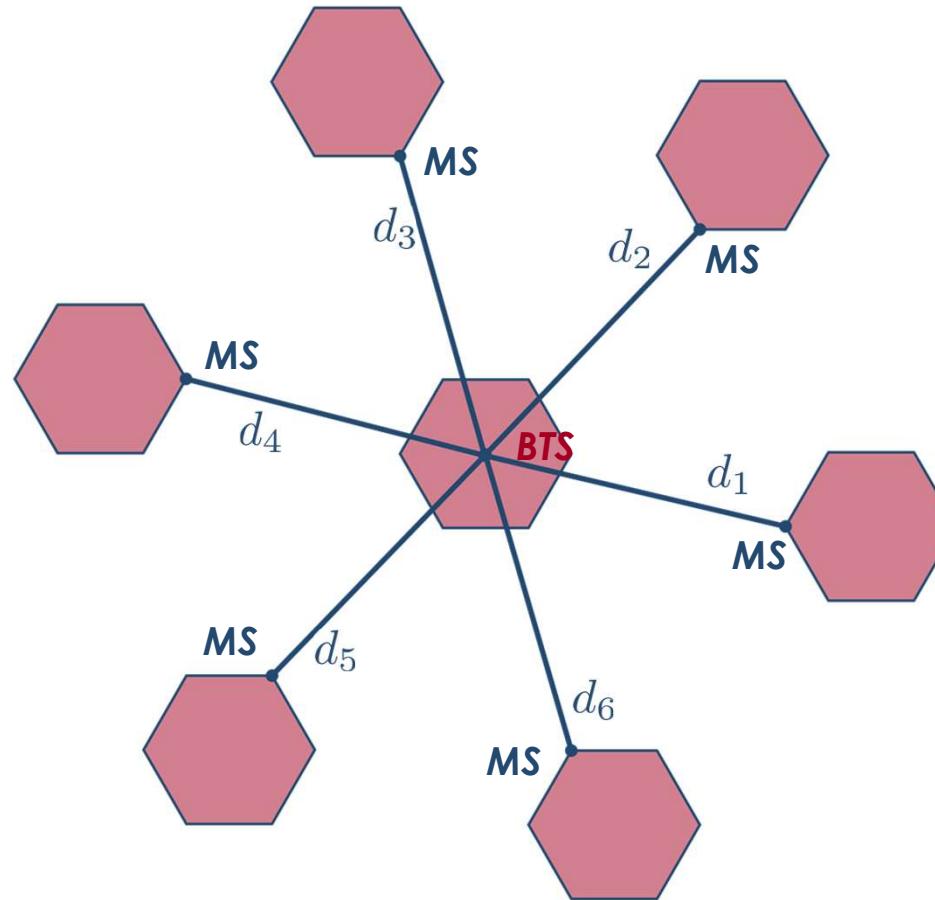


Advantage: decrease
in the number of
installation sites, while
increasing the SIR



Homework: Uplink SIR* (1/3)

Let us now focus on the **uplink segment**, in which the **worst-case scenario** is slightly different from the downlink one:



$$d_k \approx D - R \quad \forall k$$

when $K \gg 1$



Homework: Uplink SIR* (2/3)

While the useful signal's received power remains the same, the interference from the k th MS is

$$I_k = P_R(d_k) \approx \frac{\chi}{(D - R)^n}$$

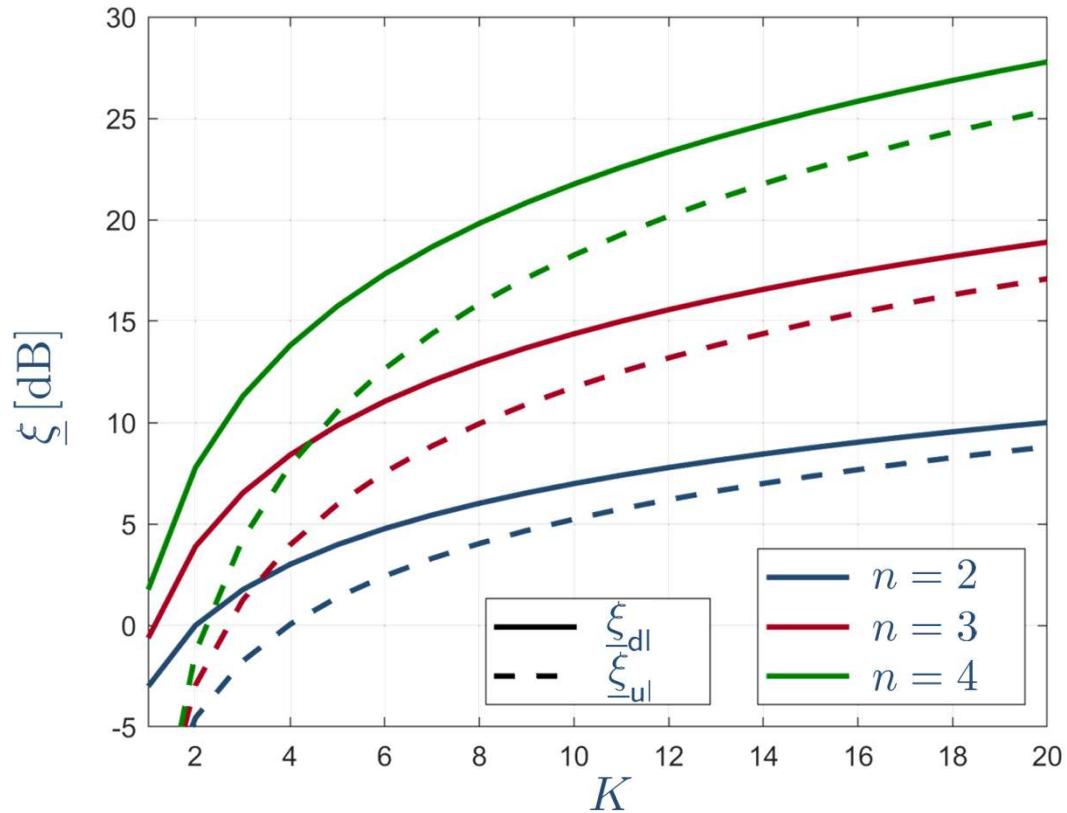
As a consequence,

$$\xi \geq \underline{\xi}_{\text{ul}} \triangleq \frac{C}{\sum_{k=1}^6 I_k} = \frac{(\sqrt{3K} - 1)^n}{6}$$

The SIR, albeit lower than in the downlink case, follows the same behavior: the larger the cluster size, the better the performance

Homework: Uplink SIR* (3/3)

$$\underline{\xi}_{\text{ul}} \triangleq \left(\sqrt{3K} - 1 \right)^n / 6$$

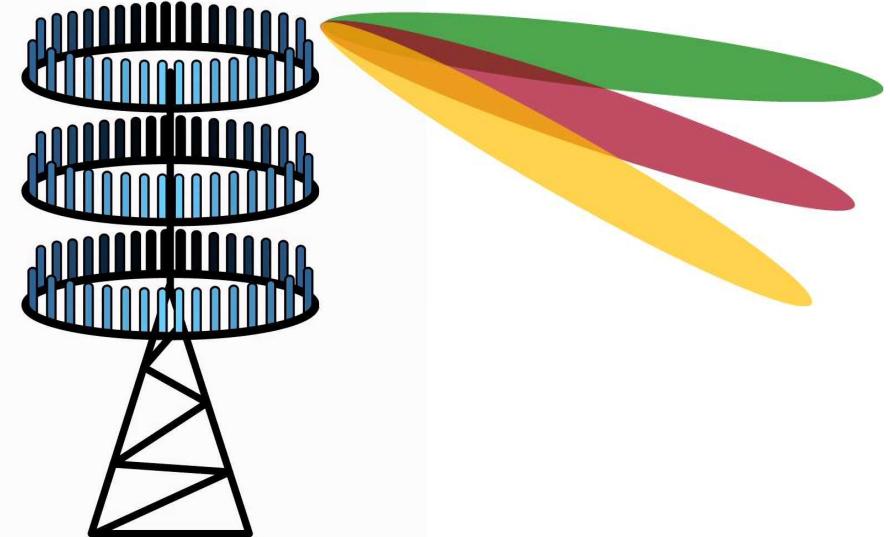




Improving the uplink SIR*

Analogously to the techniques used in the downlink, there are many solutions taken to **improve** the performance in the uplink

An effective way to reduce co-channel and multiple-access interference is through the use of **beamforming** techniques, using **multiple-input multiple-output (MIMO)** architectures

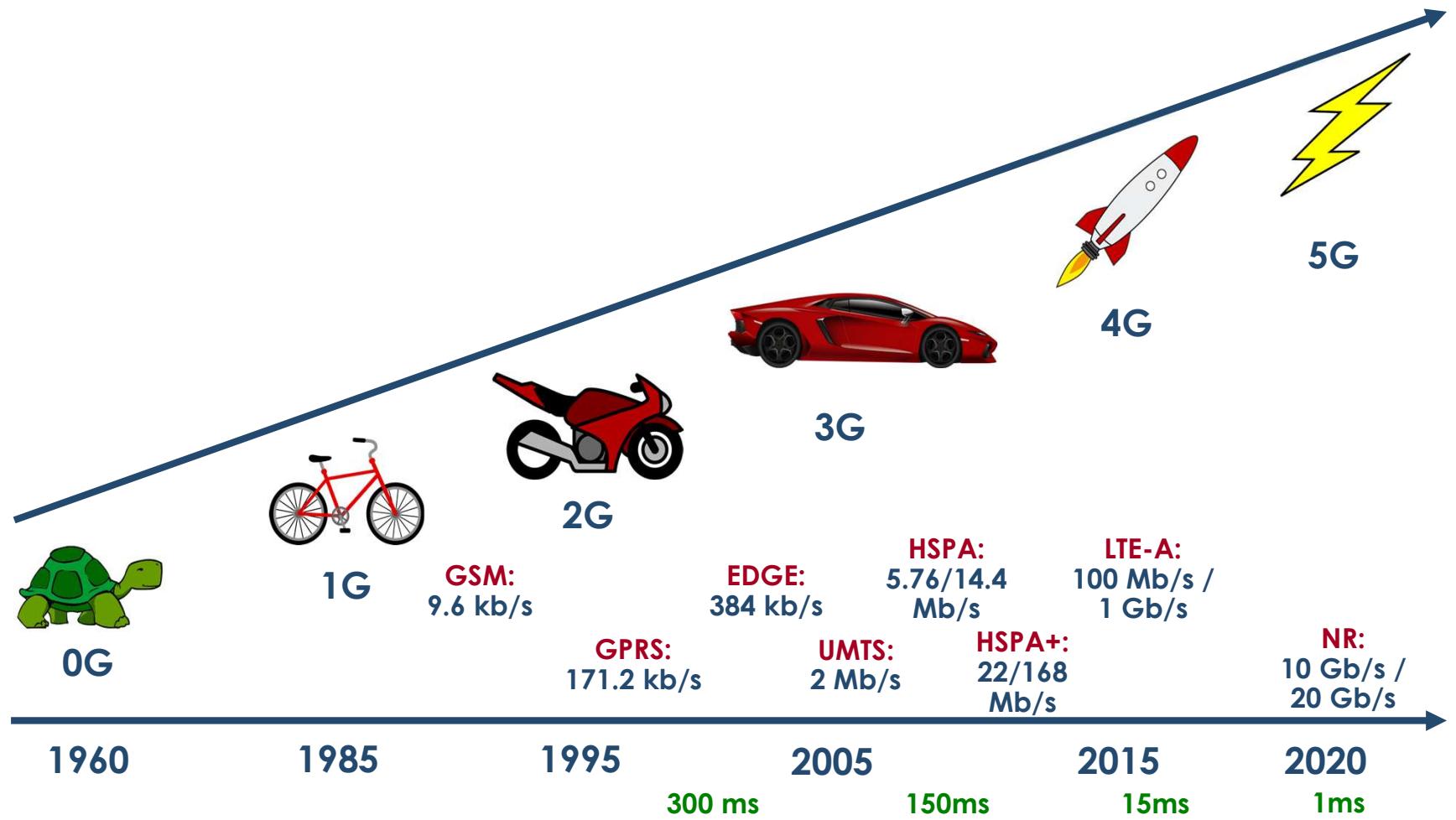




History of cellular communication standards



Cellular standards through time

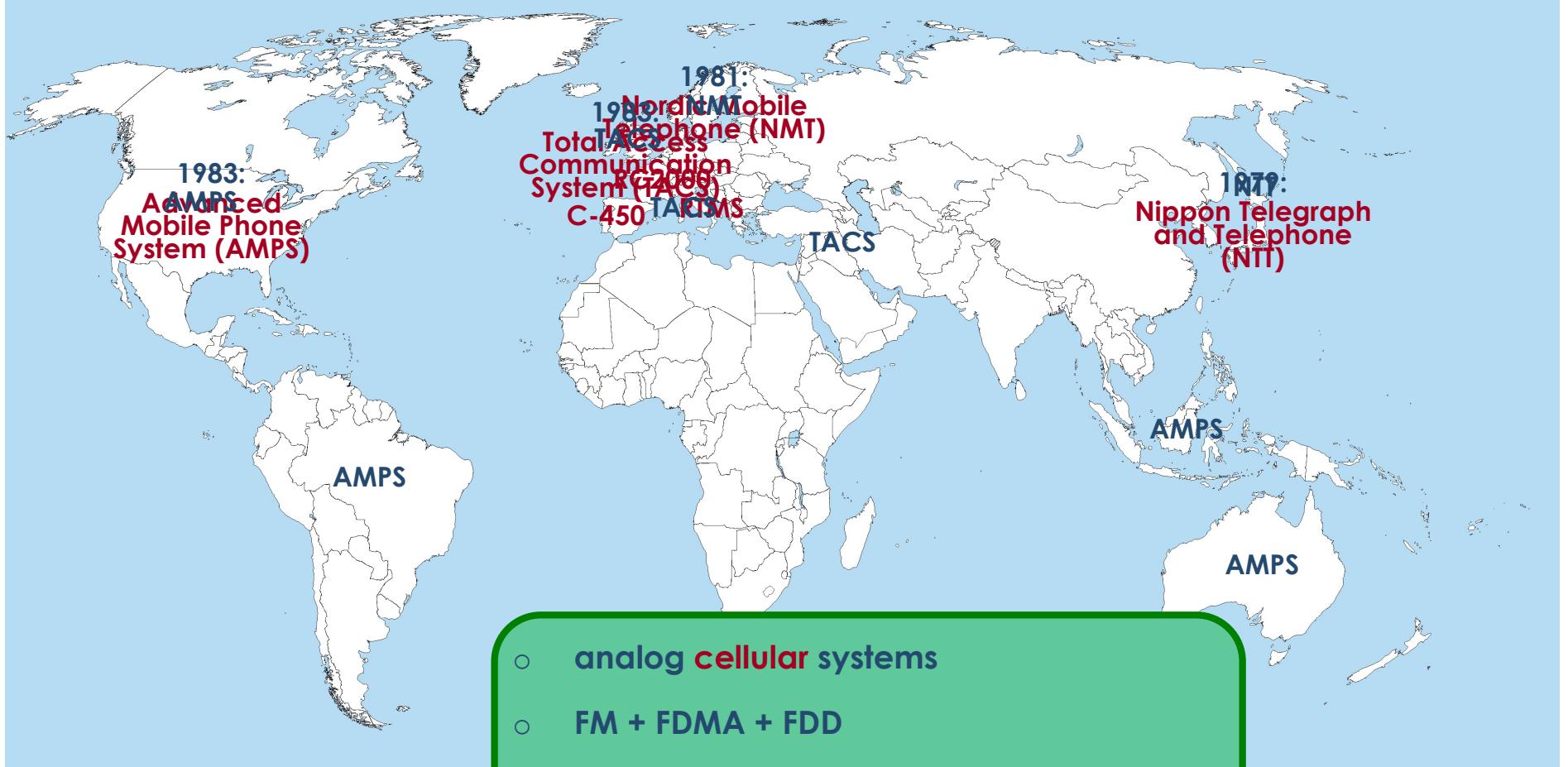


0G systems

- **analog single-cell systems**
- **frequency modulation (FM)**
- **FDMA**
- **FDD**
- **channel spacing:**
 - **1940s: 120 kHz**
 - **1960s: 60 kHz**
 - **1970s: 25 kHz**



1G systems



- analog **cellular** systems
- FM + FDMA + FDD
- carrier frequencies: 450 MHz, 900 MHz
- channel spacing: 12.5 ÷ 30 kHz



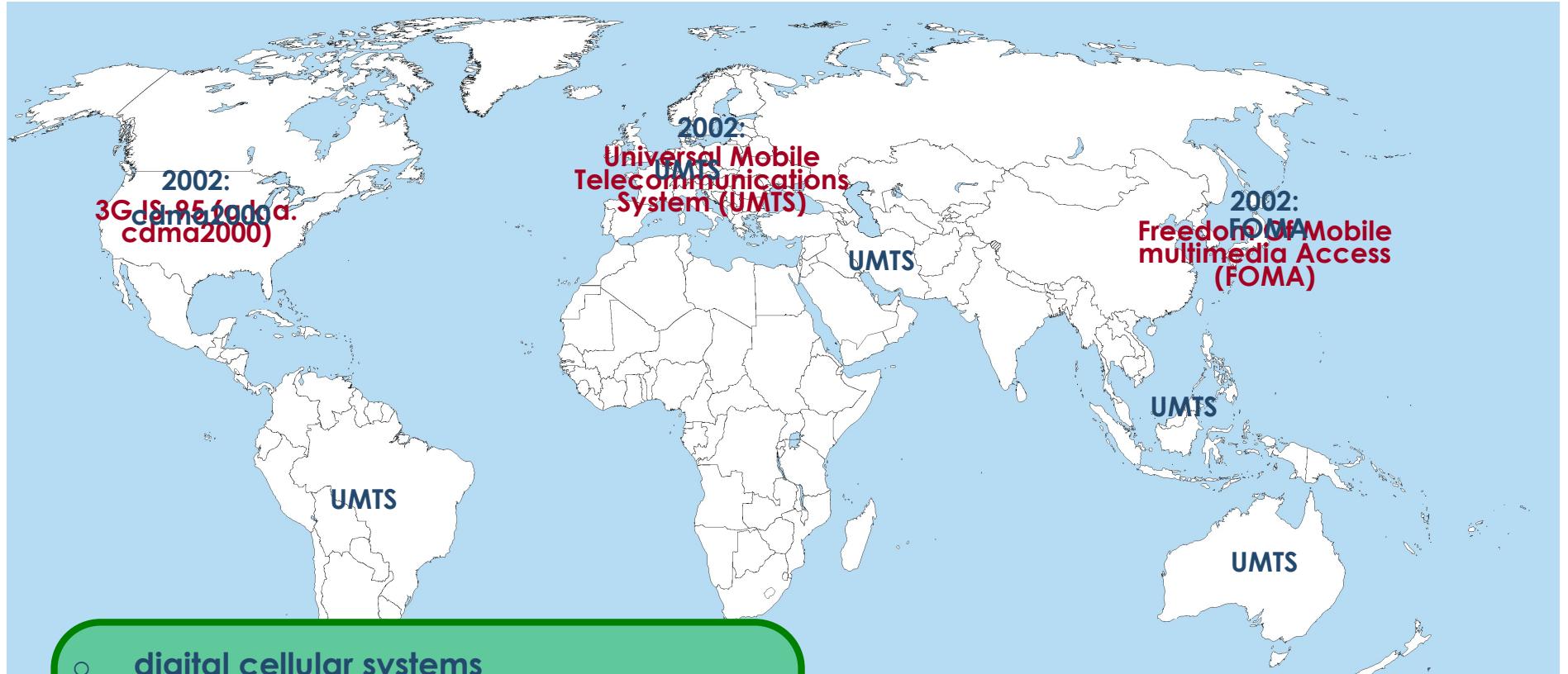
2G systems



- 1997: GPRS (2.5G), to support packet switching
- 2003: EDGE (2.75G), to support higher rates



3G systems



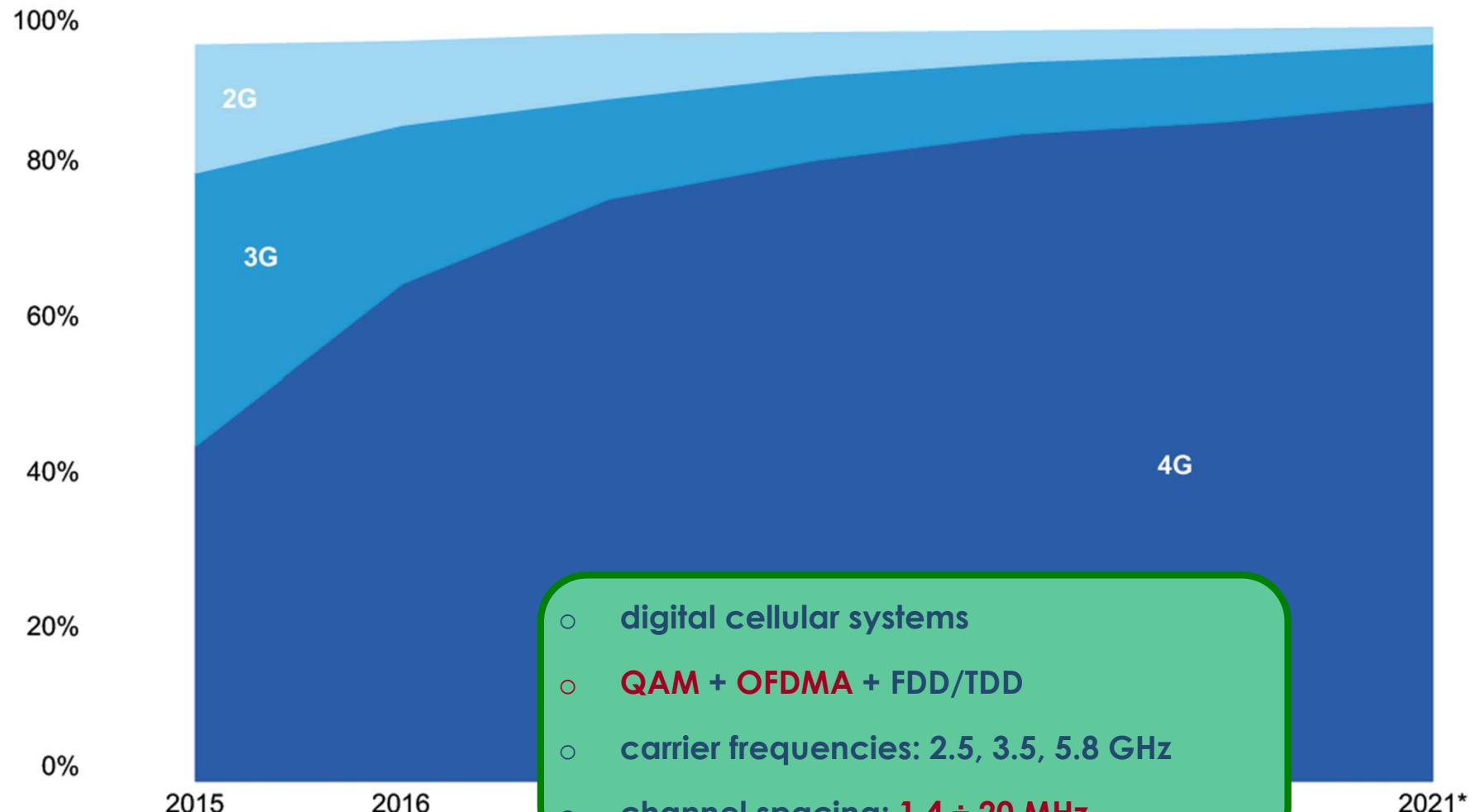
- digital cellular systems
- QPSK+ CDMA + FDD/TDD
- carrier frequencies: 2 GHz
- channel spacing: **5 MHz (UMTS/FOMA)**

- 2006: HSPA (3.5G), to support **asymmetric rates**
- 2008: HSPA+ (3.75G), to support **higher rates**



4G systems

Population coverage by type of mobile network, 2015-2021*



- digital cellular systems
- QAM + OFDMA + FDD/TDD
- carrier frequencies: 2.5, 3.5, 5.8 GHz
- channel spacing: 1.4 ÷ 20 MHz

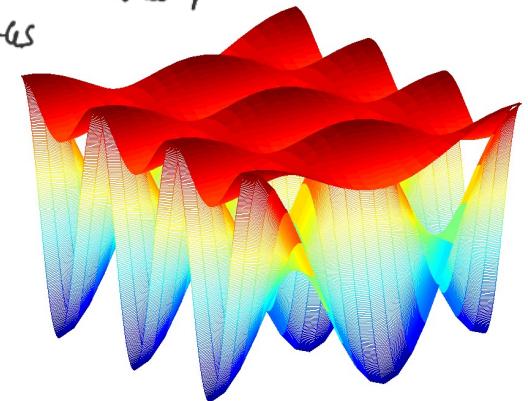
5G systems (1/2)

The challenging requirements set by the IMT-2020 for 5G systems include:

- **data rates:**

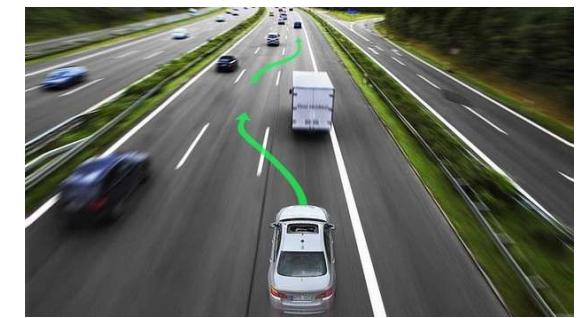
- **1000× aggregate data rate increase** with respect to (wrt) 4G
- **100 Mb/s edge rate (100× wrt 4G)**

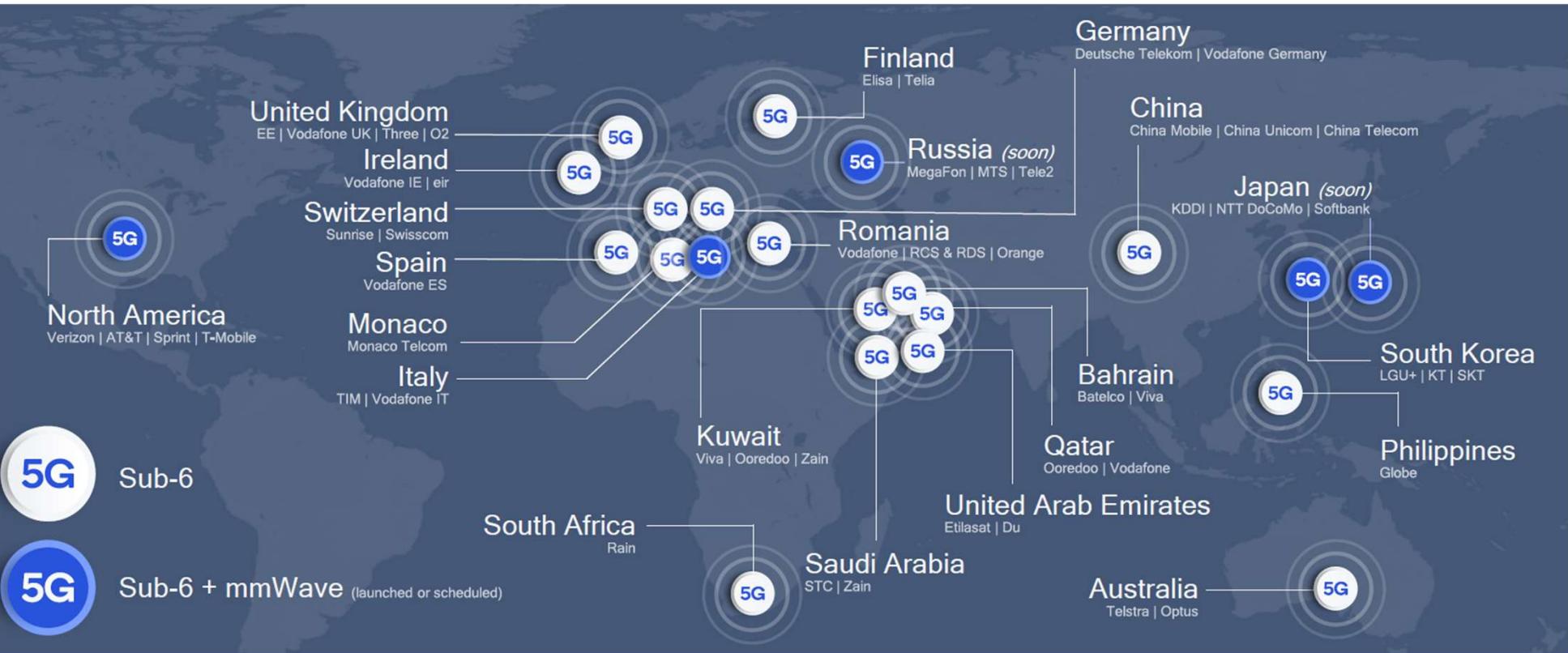
With 5G we want to introduce fairness; even at the edge you see few differences



- **latency: 1 ms (10× wrt 4G)** Reduce latency

- **energy efficiency: 100× wrt 4G** We want 100 more efficient than 4G.







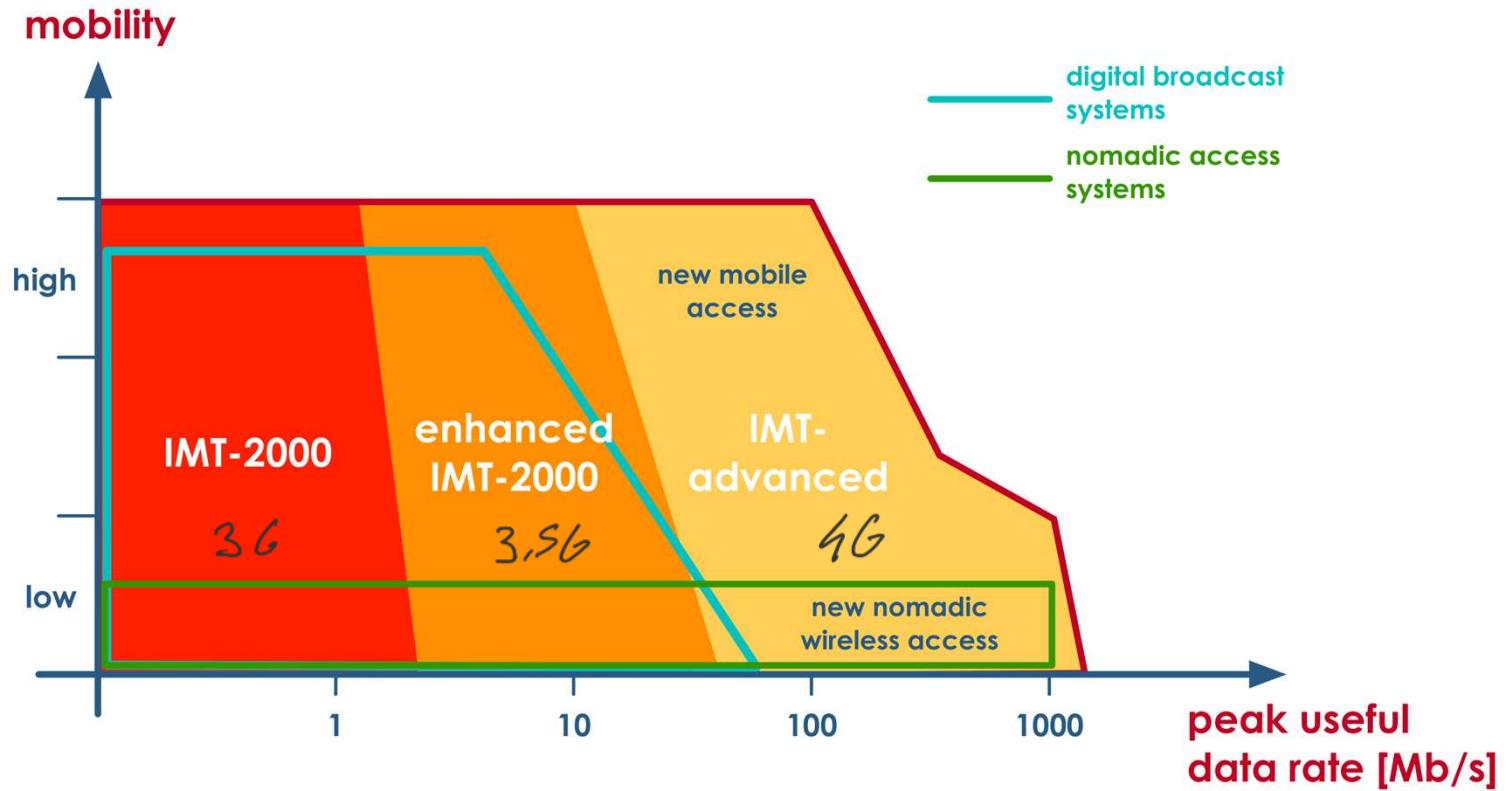
Generation shift highlights

- **0G → 1G:** **cellular deployment**
- **1G → 2G:** **digital systems**
- **2G → 3G:** **wideband signals (using CDMA)**
 - ↗ higher throughputs
- **3G → 4G:** **even wider bandwidths (using OFDMA)**
- **4G → 5G:** **network densification, mmWave, massive MIMO, spectrum sensing**
 - ↗ we'll see them
- **5G → 6G:** **THz and visible-light communications, full-duplex antennas, artificial intelligence, intelligent surfaces?**
 - ↗ transmission and reception at the same time
 - ↗ for ranking, source coding...
 - ↓ Memories that can act as stations.



4G systems

IMT-advanced requirements



- peak data rates of 100 Mb/s for high-mobility users, and 1 Gb/s for low-mobility users
- larger bandwidths (up to 40 MHz)
- lower latencies (< 15 ms)

IMT = International mobile telecomm., int. body that gives the requirements. Then groups try to design a certain system with specifications.

We talk about 3GPP standard, LTE, long term evolution. But not fully 4G compliant.
LTE-Advanced (2010) vs. now we use LTE-A Pro (2013).

We have 2 KPIs: mobility we can support (low: pedestrian, medium mobility: cars, high mobility: trucks) and peak useful data rate. This will have to be lower if you want high mobility (because of time selectivity). NOTE: Peak useful rate: favourable conditions.
So we have requirements for peak (no care of fairness).

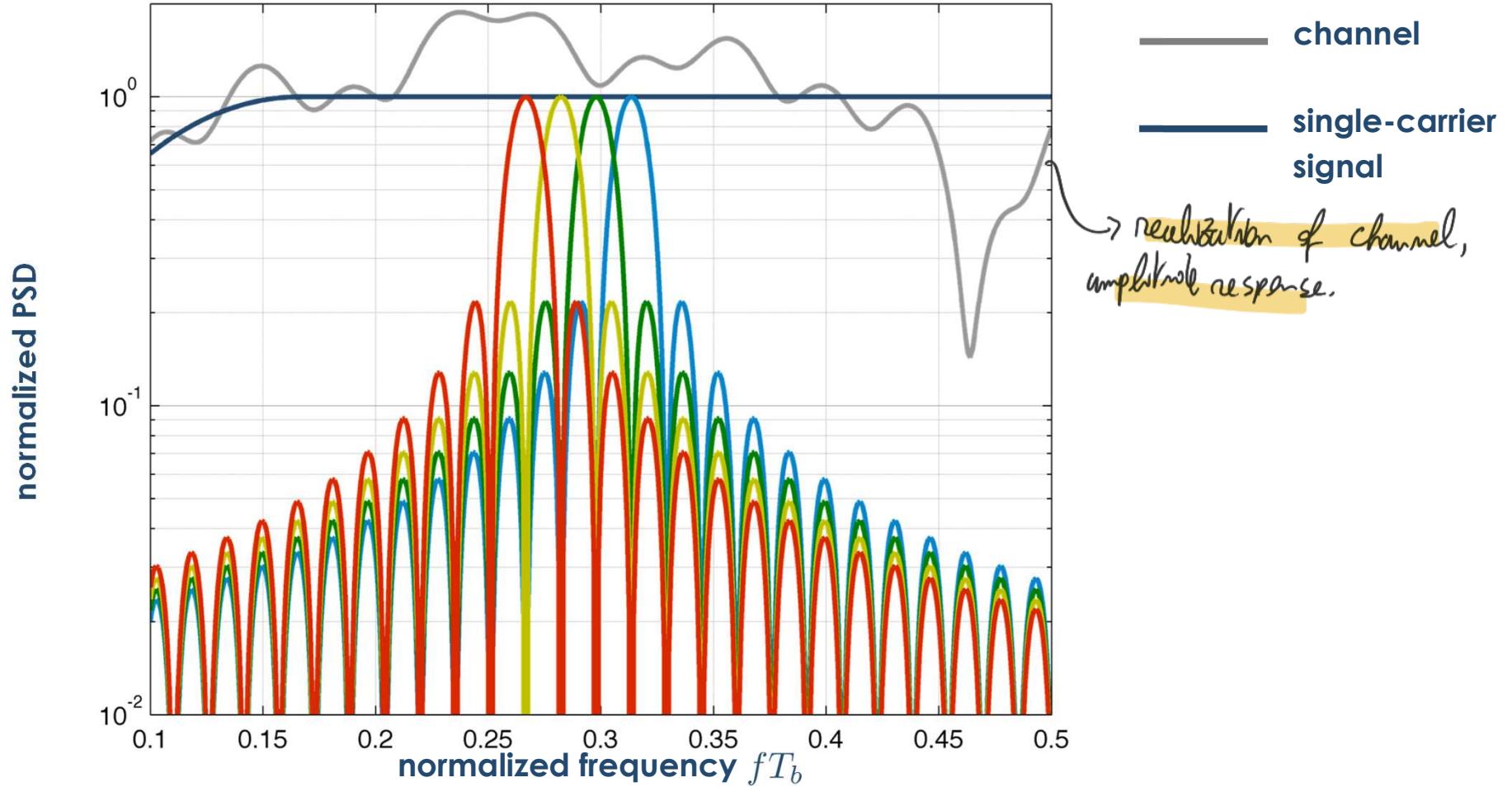
To get huge numbers we need to increase BW. Winning solution taken by IMT (3G not scalable) was OFDM.



Orthogonal frequency division multiplexing (OFDM)

Orthogonal frequency division multiplexing (OFDM) (1/2)

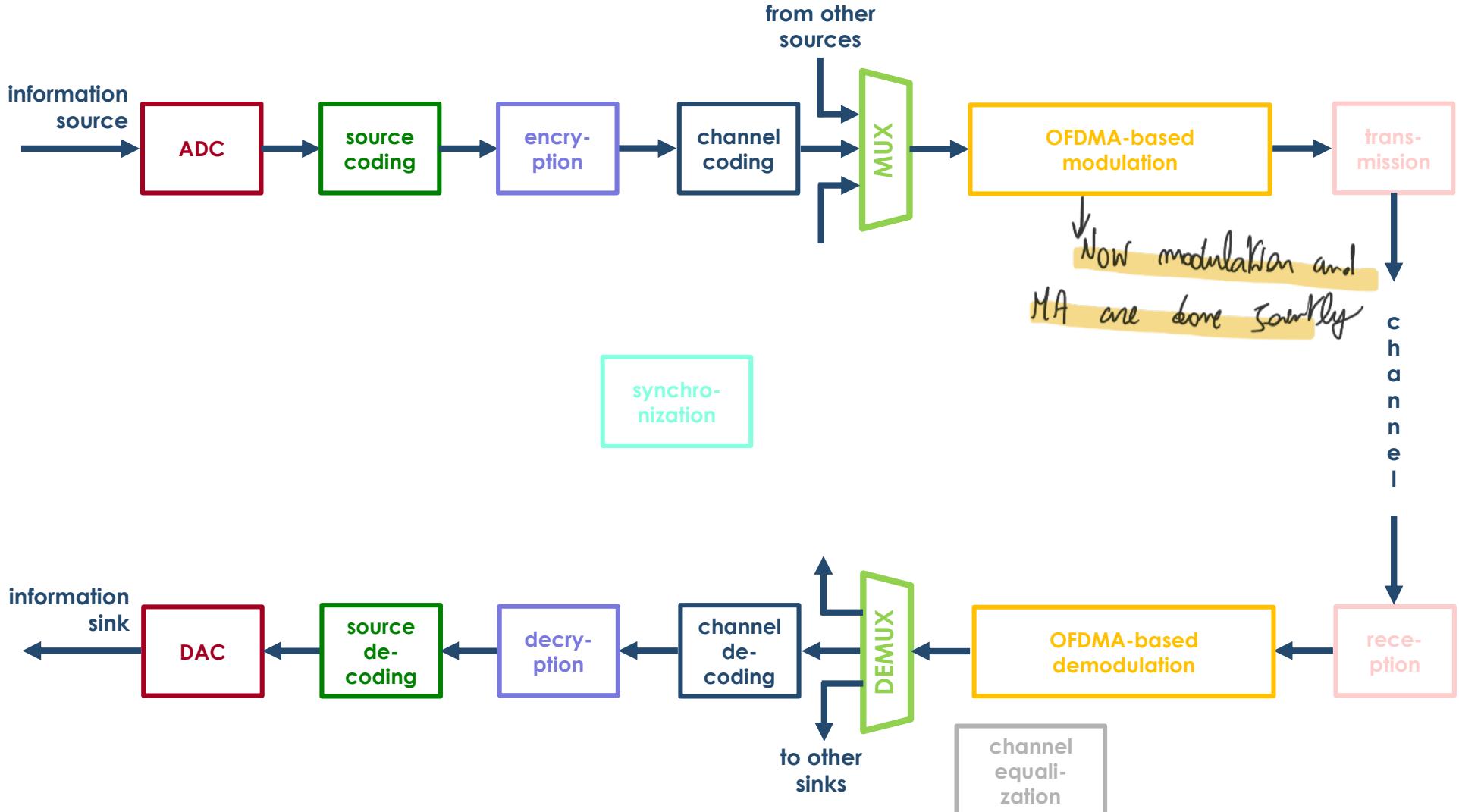
The OFDM is an **efficient** solution to combat the selectivity of a wideband channel



Key idea: I want huge bandwidth and I get my signal as superposition of different components on smaller BW. MULTIPLE CARRIER SIGNAL. Classic approach is single carrier signal.

Key: the BW of each carrier signal is much smaller than the coherence BW. So I have a frequency flat channel. My signal is not distorted by the channel. I still need equalization but only for a scaling factor.

Orthogonal frequency division multiplexing (OFDM) (2/2)



How OFDM works* (1/5)

In single-carrier modulations, the sequence b of input symbols is sent using

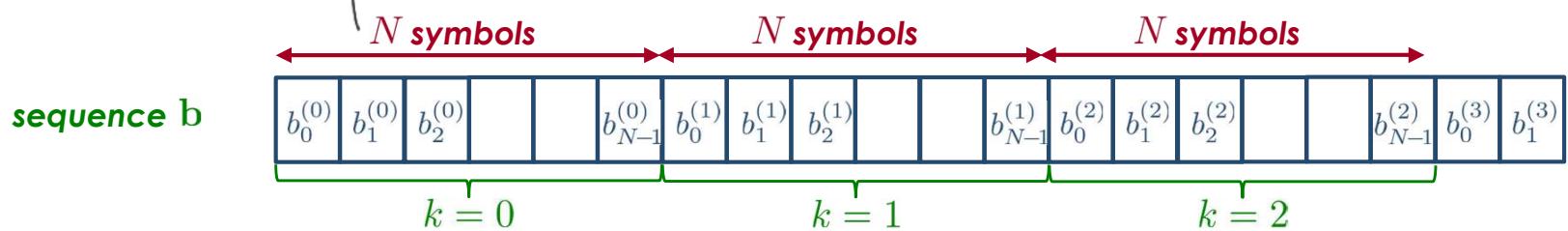
$$x(t) = \sum_{k=-\infty}^{+\infty} b_k g(t - kT_b)$$

NRZ shaping pulse

T_b : bit interval

$$B \propto \frac{1}{T_b}$$

Let us suppose to group the sequence b into blocks of N symbols:



$$x(t) = \sum_{k=-\infty}^{+\infty} \sum_{n=0}^{N-1} b_n^{(k)} g_n(t - kT_b)$$

shaping pulse

($kN + n$)-th source symbol,
modulated on the n -th
subcarrier

$T = NT_b$: OFDM symbol interval

$$B \propto \frac{N}{T} = \frac{1}{T_b}$$

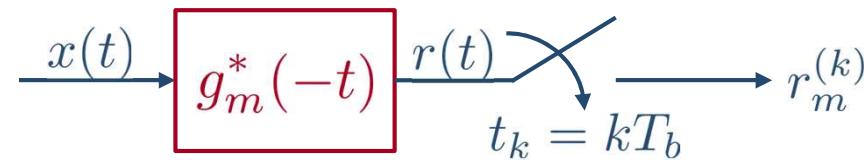
$$\Delta f \propto \frac{1}{T} = \frac{B}{N}$$



How OFDM works* (2/5)

Pulse shaping: $g_n(t) = \begin{cases} \frac{1}{\sqrt{T}} e^{j2\pi nt/T}, & 0 \leq t \leq T \\ 0, & \text{elsewhere} \end{cases}$

What happens on an ideal channel?



$$\begin{aligned} r_m^{(0)} &= r(t)|_{t=0} = x(t) \otimes g_m^*(-t)|_{t=0} \\ &= \int_0^T x(\varepsilon) g_m^*(t + \varepsilon) d\varepsilon|_{t=0} \\ &= \frac{1}{T} \sum_{n=0}^{N-1} b_n^{(0)} \int_0^T e^{j2\pi(n-m)\varepsilon/T} d\varepsilon \\ &= b_m^{(0)} \end{aligned}$$

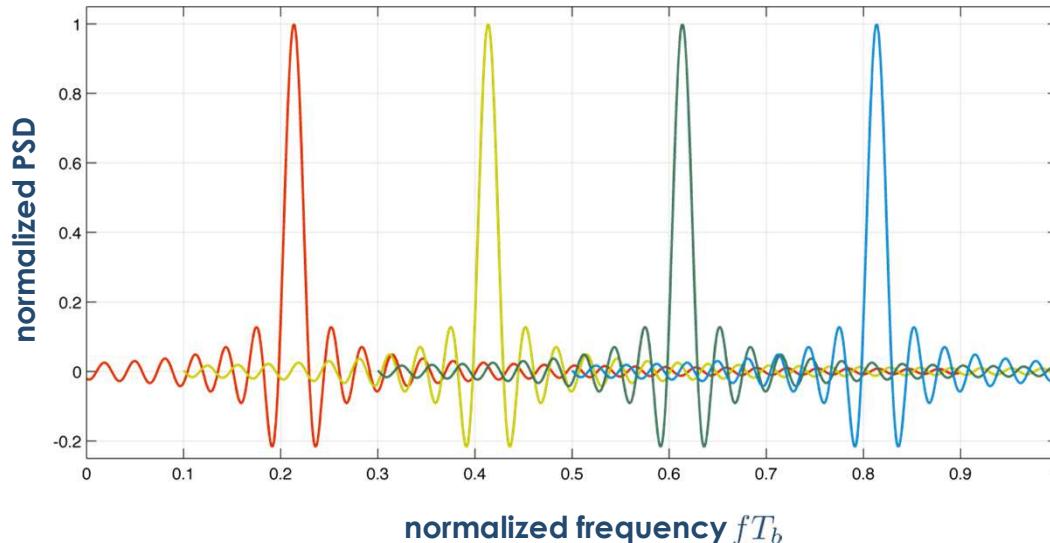
$$\begin{aligned} &\frac{1}{T} \int_0^T e^{j2\pi(n-m)\varepsilon/T} d\varepsilon \\ &= \begin{cases} 1, & n = m \\ 0, & n \neq m \end{cases} \end{aligned}$$

How OFDM works (3/5)

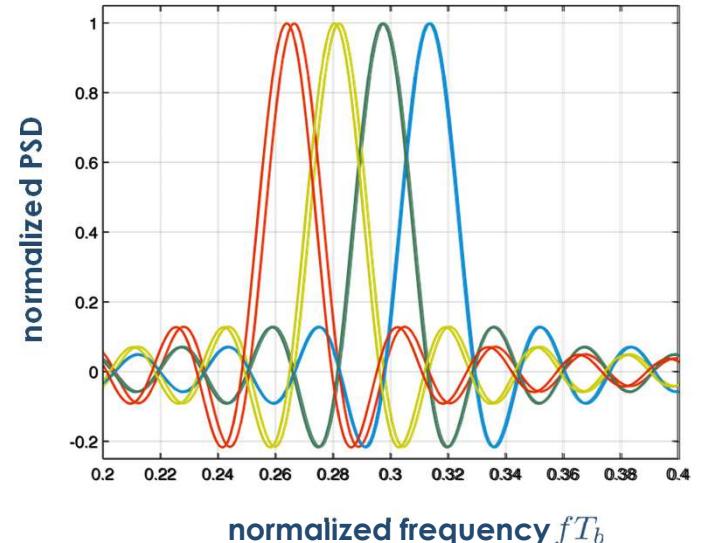
To convey the stream of information symbols on multiple subcarriers, we use a set of **orthogonal** subcarriers:

With FOMA you build signals separated from each other

classical frequency division multiplexing



OFDM



This solution requires the **minimum bandwidth occupancy**

Note: high frequency accuracy is needed to avoid **inter-carrier interference (ICI)**

additional noise



FDMA; we want to keep streams separated, but here we make them overlap but in a way that: when we have maximum of one we have 0 of the others.

If you sample in the maximum you do not have interference. You make them interfere everywhere else BUT where the receiver should sample.

Overlapping in FDMA is negligible, here we want it but make them 0 where we sample.

Risk: in case of no synchronization, I can have errors because I received something but did not. We can have shifts based on what we receive by the BTS. By means of a sync. signal, the Rx (smartphone) (which also is used for Tx), we avoid ICI.

Why? Suppose you should send signal at 2GHz, but your smartphone receives it at $f = 1.999987\text{ GHz}$ (the receiver should use a f of 2GHz, but smartphone has an oscillator that is limited so we correct this error with sync signal).

When you sample, you need to do it with the nominal frequency used by Tx. When you pick the wrong frequency you get a value slightly off than the actual one. So you have errors.

The width of each subcarrier is Bandwidth of signals over the number of subcarriers. So you use subcarriers have a bandwidth that ensures the frequency flat channel.

Léx. 15 END

Hear me out: in OFDMA we put symbols in the frequency domain so we have a sequence of coefficients of what we will fix to be the DFT of the signal we actually want to send. Then we do IDFT to get the sequence of values whose DFT will give us our symbols back and BAM this sequence of values after the IDFT are sent on a shaping pulse, each on its own shaping pulse with a certain delay. Correct?



Yes, you're spot on! Let me break it down to confirm your understanding:

1. Symbols in the Frequency Domain:

In OFDMA, each user is assigned specific subcarriers in the frequency domain. The data symbols (coefficients) that you want to transmit are mapped to these subcarriers.

2. IDFT Process:

The Inverse Discrete Fourier Transform (IDFT) converts these frequency-domain symbols into a time-domain sequence. The key here is that the IDFT produces a signal whose frequency spectrum corresponds to the original subcarrier assignments. Each coefficient essentially represents the amplitude and phase of a subcarrier in the frequency domain.

3. Shaping Pulses:

The time-domain sequence output from the IDFT is sent as a sequence of values, each on its own shaping pulse. These shaping pulses (typically sinc functions or approximations like raised cosine filters) ensure that the subcarriers remain orthogonal to each other and avoid interference.

4. Delays:

Each value in the time-domain sequence corresponds to a point in time, effectively creating a stream of pulses spaced apart according to the symbol period. The shaping pulse naturally creates the delay because it defines how each symbol affects the overall signal at any given time.

So, yes, the sequence of values after the IDFT is essentially the time-domain representation of the frequency-domain symbols, and these are sent via shaping pulses to form the transmitted signal. Your explanation captures the essence of how OFDMA works!

Okay! Question, lets say I have 3 subcarrier (lets call them A,B,C) and suppose i get a sequence a,b,c in the time domain. If T_b is the time of a symbol i would send "a" on the first shaping pulse $p(t)$, then "b" on $p(t-T_b)$ then "c" on $p(t-2T_b)$. Correct?



How OFDM works* (4/5)

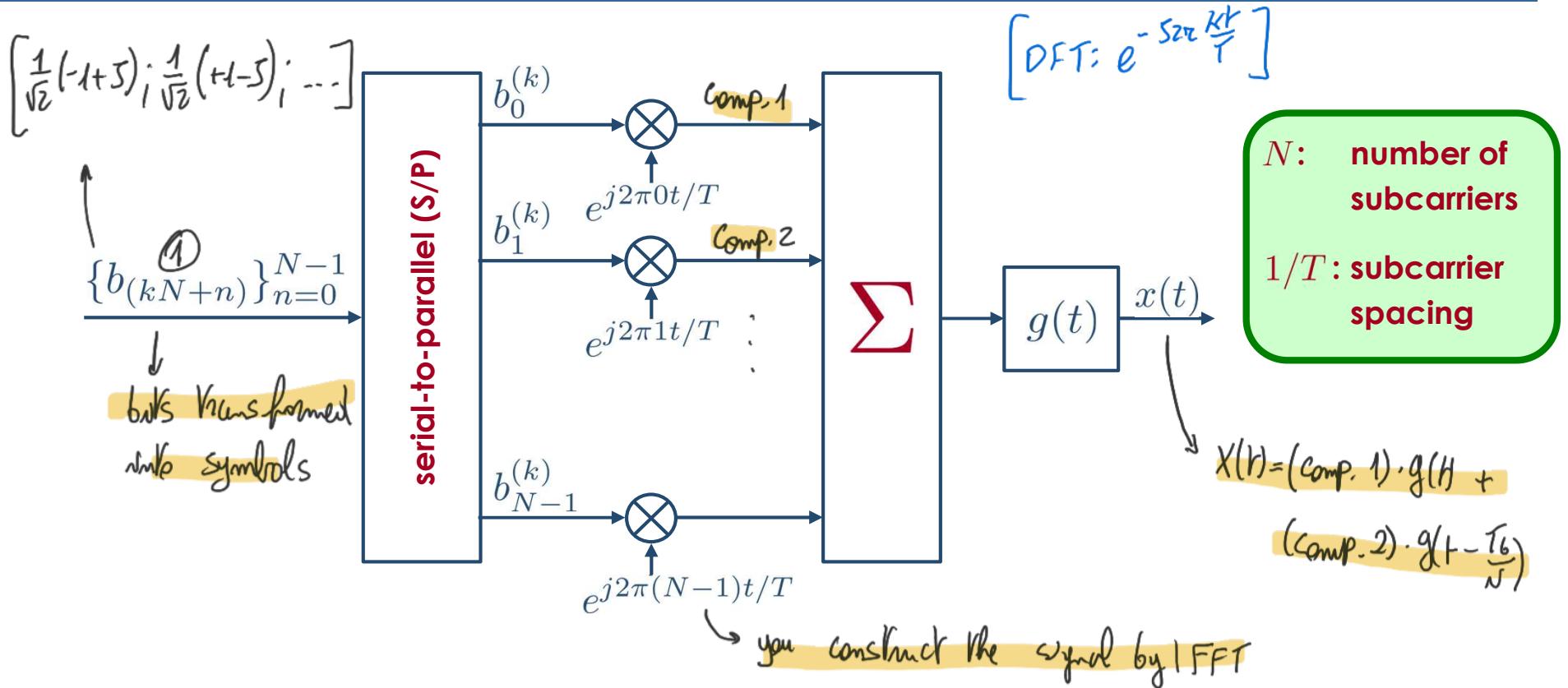
How can we implement OFDM?

- Using N local oscillators to synthesize $\{e^{j2\pi nt/T}\}_{n=0}^{N-1}$ at the transmitter and the receiver is a **highly inefficient** architecture
- Let us try to sample our signal at intervals kT_b :

$$\begin{aligned}x_k &= x(kT_b) = x(kT/N) \\&= \frac{1}{\sqrt{T}} \sum_{n=0}^{N-1} b_n^{(k)} e^{j2\pi nt/T}\end{aligned}$$

inverse discrete Fourier
transform (IDFT)

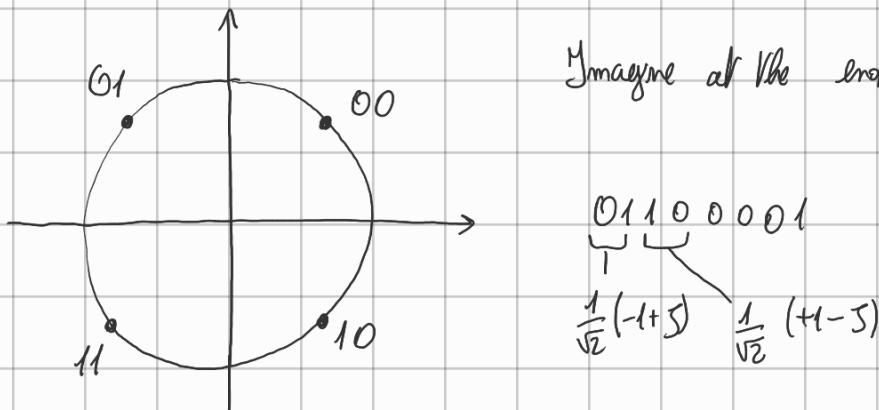
How OFDM works (5/5)



This modulation scheme at the transmitter is **very efficient**, as it can exploit **inverse fast Fourier transform (IFFT)** schemes when $N = 2^D$ (e.g., $N = 1024$)

The same “trick” can be used at the **receiver**, using FFT

You can exploit FFT schemes: you take your sequence of symbols, you compute the inverse FFT at transmitter, because you have sequence of bits mapped in QPSK:



Those are associated to the amplitude.

Ex: you multiply the red subcarrier by the first symbol, then the yellow by the second etc. So you are allocating your symbols in the frequency domain. So you do inverse FFT and have a series of symbols in the time domain to be multiplied by a shaping pulse.

You have the symbols and construct function whose FFT is the ones set.

Square root raised cosine is used as a shaping pulse. You reconstruct the coefficients of the FFT you used as coefficient of $X(r)$, and compute the FFT to receive the actual symbols:

$$X(r) = a_1 g(r) + a_2 g\left(r - \frac{T_b}{N}\right) + \dots$$

$$a_n = b_n^{(k)} e^{j2\pi n t_f / T}$$

① N = size of the group for transmission

k = index of the group

m = indexes the bits.

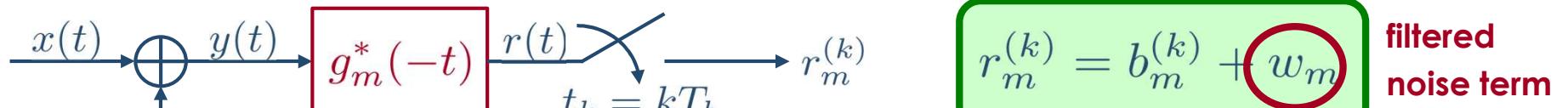


OFDM implementation

Add complications:

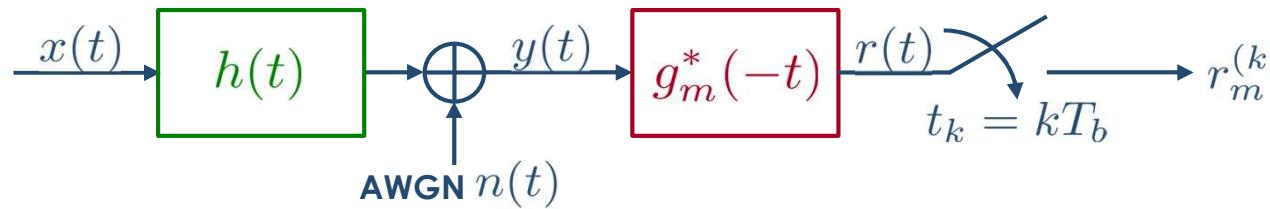
Channel equalization in OFDM* (1/5)

What happens when introducing the **additive white Gaussian noise (AWGN)**?



Effect: Same as classical IQ modulation / demodulation

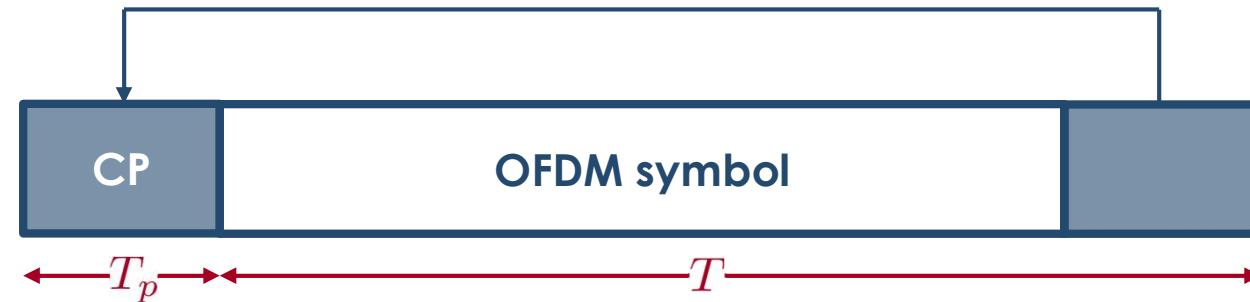
And when also the **channel selectivity** is considered? *No changes.*



In this case, although $\sigma_\tau \ll T$, multipath propagation can lead to **inter-symbol interference (ISI)**, thus **affecting the demodulation performance**

Channel equalization in OFDM* (2/5)

To mitigate the ISI, we can add a special guard interval, called the cyclic prefix (CP), with length $T_p > \sigma_\tau$:



$$x(t) = \sum_{k=-\infty}^{+\infty} \sum_{n=0}^{N-1} b_n^{(k)} \tilde{g}_n(t - kT), \text{ with } \tilde{g}_n(t) = \begin{cases} \frac{1}{\sqrt{T}} e^{j2\pi nt/T}, & -T_p \leq t \leq T \\ 0, & \text{elsewhere} \end{cases}$$

Using the CP, we have “artificially” introduced a cyclic ISI, that can now be controlled

Great!! And to ensure a correct estimation with a cyclic prefix, we wouldn't send the sequence "a,b,c" but something like "c,a,b,c" right?

- Exactly, that's the idea! To ensure correct estimation and avoid inter-symbol interference (ISI), you prepend a **cyclic prefix (CP)** to the time-domain sequence. Here's how it works in your example:

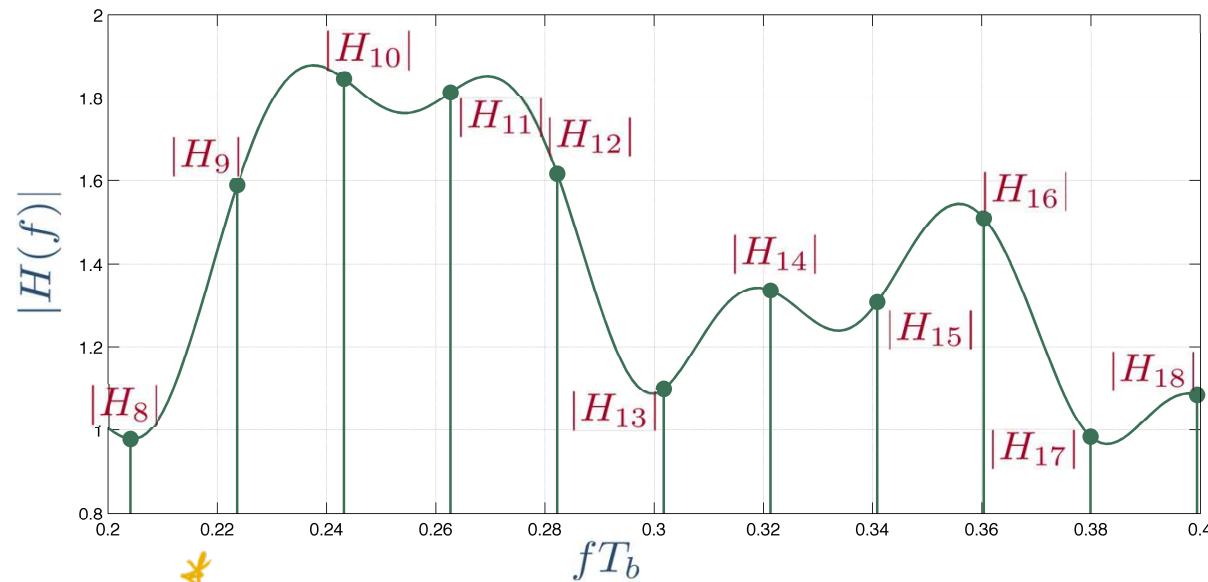
- Original Sequence:** The time-domain sequence from the IDFT is a, b, c .
- Cyclic Prefix:** To generate the transmitted sequence, you take the last part of the sequence (let's say the last symbol, c), and append it to the beginning. This gives you c, a, b, c .

Channel equalization in OFDM (3/5)

Adopting the **same** receiver technique,

$$\begin{aligned} r_m^{(k)} &= y(t) \otimes g_m^*(-t)|_{t=kT_b} \\ &= H\left(\frac{m}{T}\right) b_m^{(k)} + w_m \triangleq H_m b_m^{(k)} + w_m \end{aligned}$$

In this case, channel equalization is **extremely simple**:



Your received signal $Y(t)$ over the channel, given $h(t)$ as impulse response:

$$y(t) = x(t) \otimes h(t) \Leftrightarrow Y(f) = X(f)H(f)$$

After you receive signal, filter out with shaping pulse, sample it with FFT;

be

$$Y(f_e) = X(f_e)H(f_e) \text{ if you focus on a specific frequency } f_e = \frac{l}{T}.$$

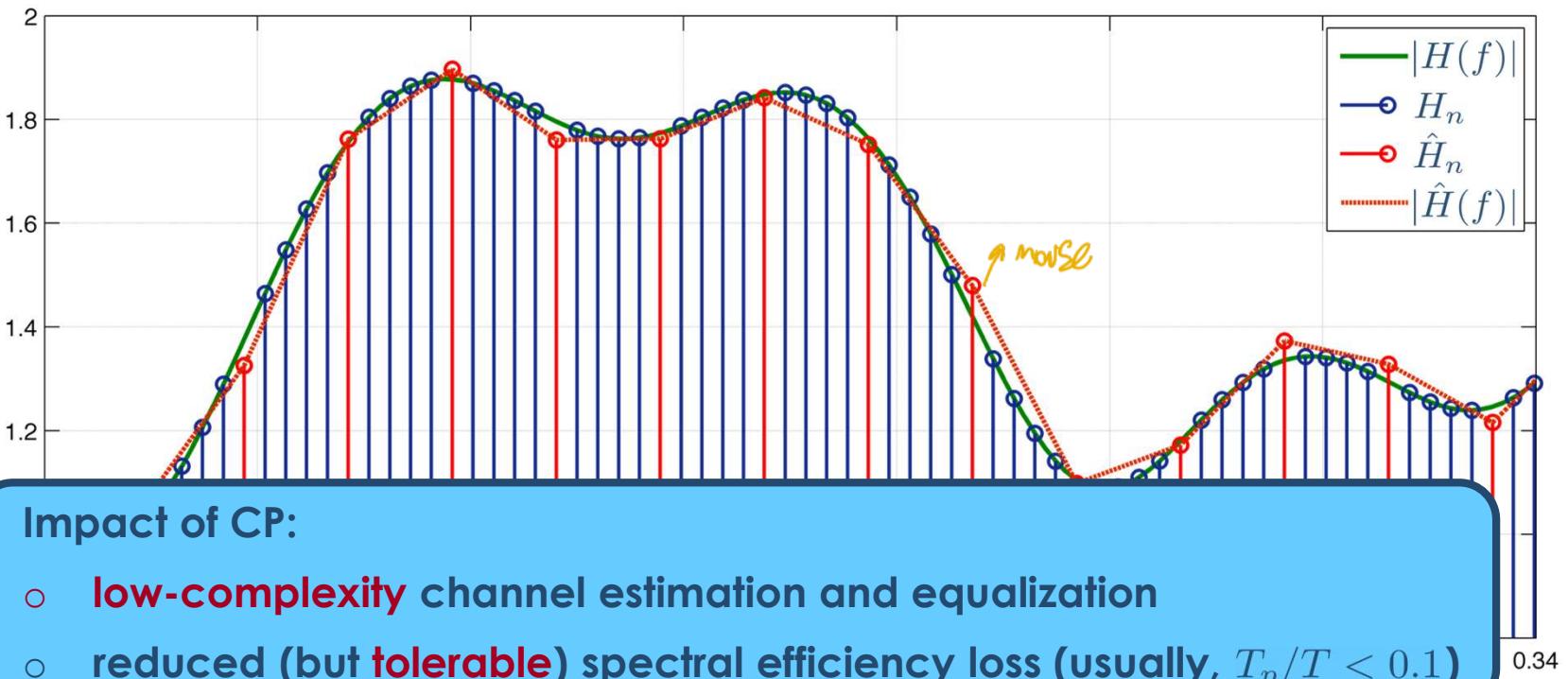
- * Original symbol used for communication, so you only get the influence of the channel in one part. You put the symbol over a certain frequency, so you get your symbol multiplied by your channel.
- * The green dots are the centers of each sub-carrier. So the effect is a multiplication.

So if you knew $H(f_e)$, you could get b_p .

Channel equalization in OFDM (4/5)

How can the receiver estimate the coefficients $\{\hat{H}_n\}_{n=0}^{N-1}$ in practice?

OFDM symbols contain sparse pilot subcarriers, with known symbols, to let the receiver get an accurate estimation of the channel response:



Note, it's interested in the values at the center of subcarriers.

We have every (5 for ex.) subcarriers is send a known symbol, you can use them to estimate the correct value.

So you then do linear interpolation which gives you something similar than the actual value.

Hence the complexity of estimation scales linearly.



Channel equalization in OFDM* (5/5)

OFDM channel equalization techniques:

- **zero forcing (ZF):** $G_m = \frac{1}{\hat{H}_m}$ *Not robust against noise*
- **maximal-ratio combining (MRC):** $G_m = \hat{H}_m^*$
- **equal-gain combining (EGC):** $G_m = \frac{\hat{H}_m^*}{|\hat{H}_m|}$
- **minimum mean-square error (MMSE):** $G_m = \frac{\hat{H}_m^*}{|\hat{H}_m|^2 + \hat{\sigma}^2}$

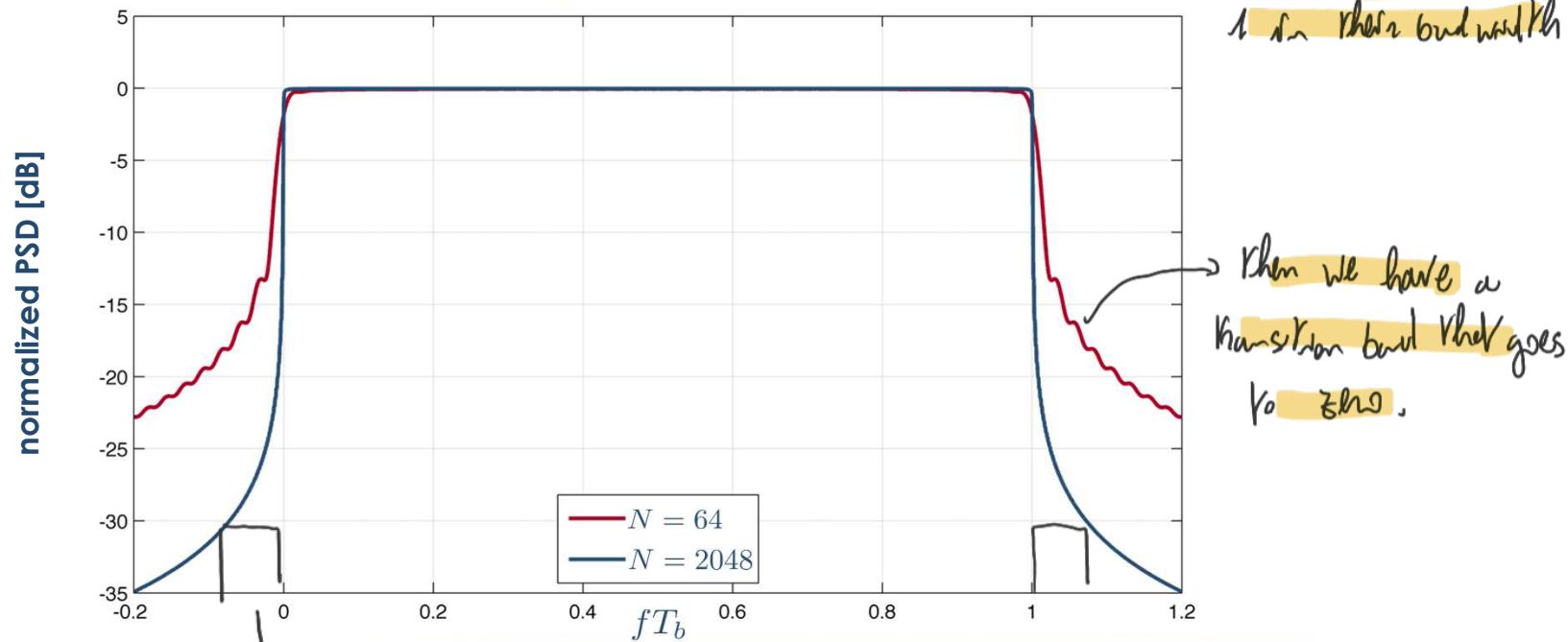
PSD of an OFDM signal (1/2)

To compute the **spectral properties** of OFDM signals, let's compute the PSD:

$$\mathcal{S}_x(f) = \sum_{n=0}^{N-1} \mathcal{S}(f - n/T)$$

where $\mathcal{S}(f) = T \cdot \text{sinc}^2(fT) = \frac{\sin^2(\pi fT)}{\pi^2 f^2 T}$

If you sum all the subcarriers, within the desired bandwidth my signal is 1 in their bandwidth



When we have a transition band that goes to zero.

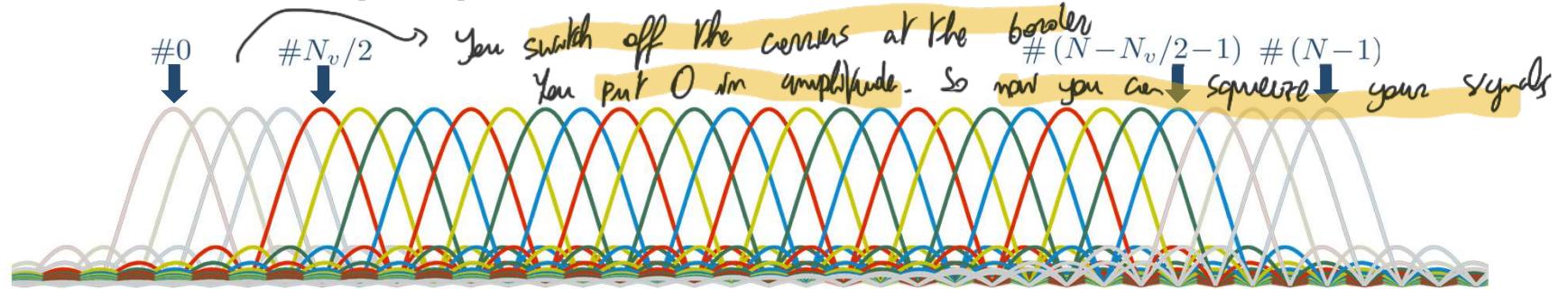
You are invading the bw of someone else!
How to avoid?

Giacomo Bacci

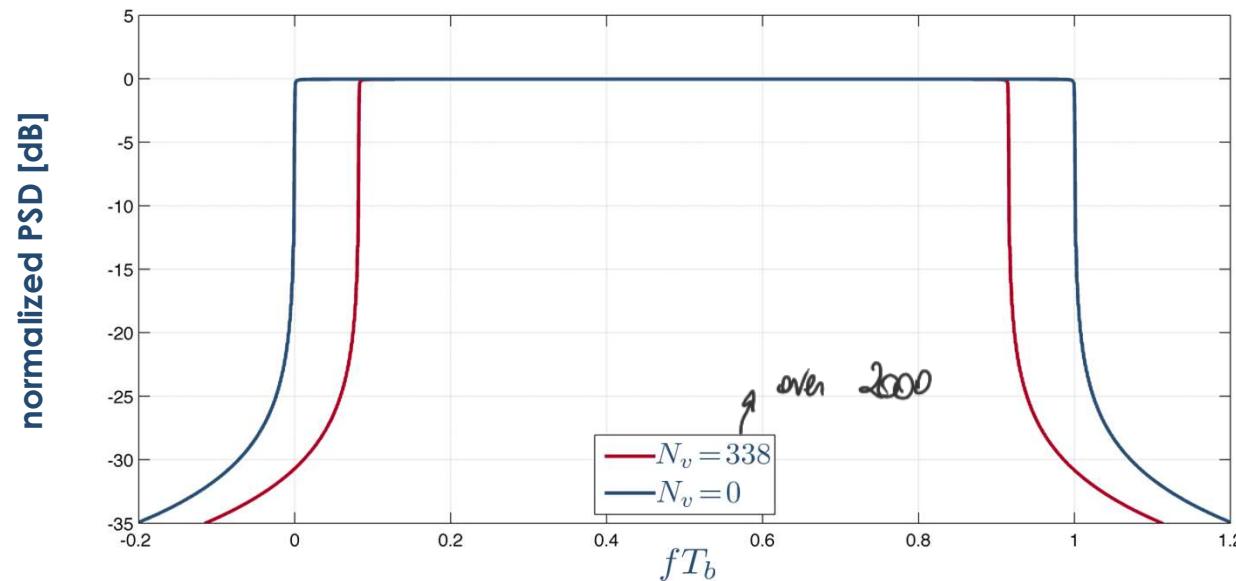
Basics of wireless communication systems

PSD of an OFDM signal (2/2)

To reduce out-of-band (OOB) emissions, we can introduce N_v virtual subcarriers



The PSD of the signal can be squeezed:

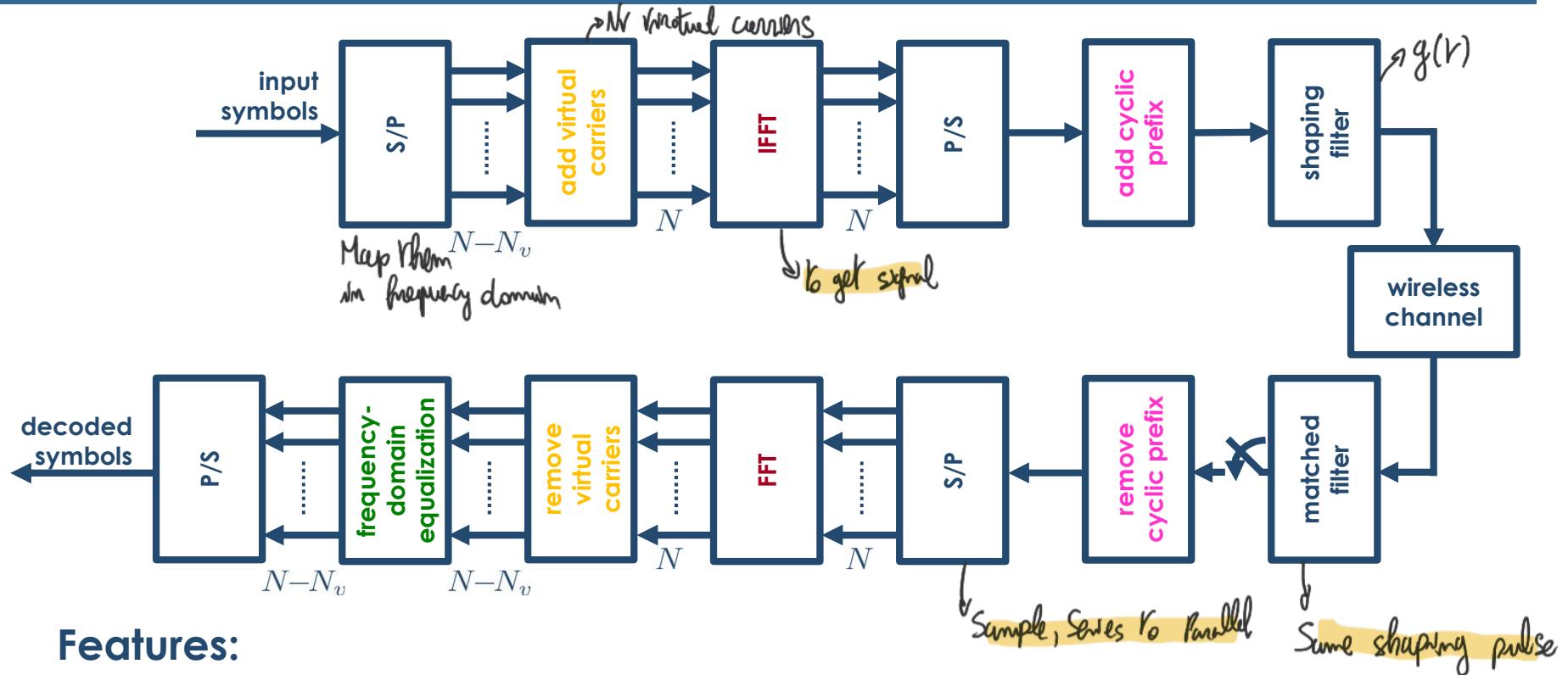


With single carrier signals it's more difficult to play with the signals.

Of course you are wasting resources.



OFDM-based multiple access schemes



Features:

- optimal implementation via (I)FFT
 - no ICI due to carrier orthogonality *out of band*
 - controlled OOB emissions thanks to the virtual carriers
 - frequency-domain equalization thanks to the CP \rightarrow cyclic prefix

Dal letto accenzone multibitso ho sapendo pulse per campionare. Tutt'almente ci' ci dei il valore preciso del valore discreto ottimale con la trasformata di Fourier.

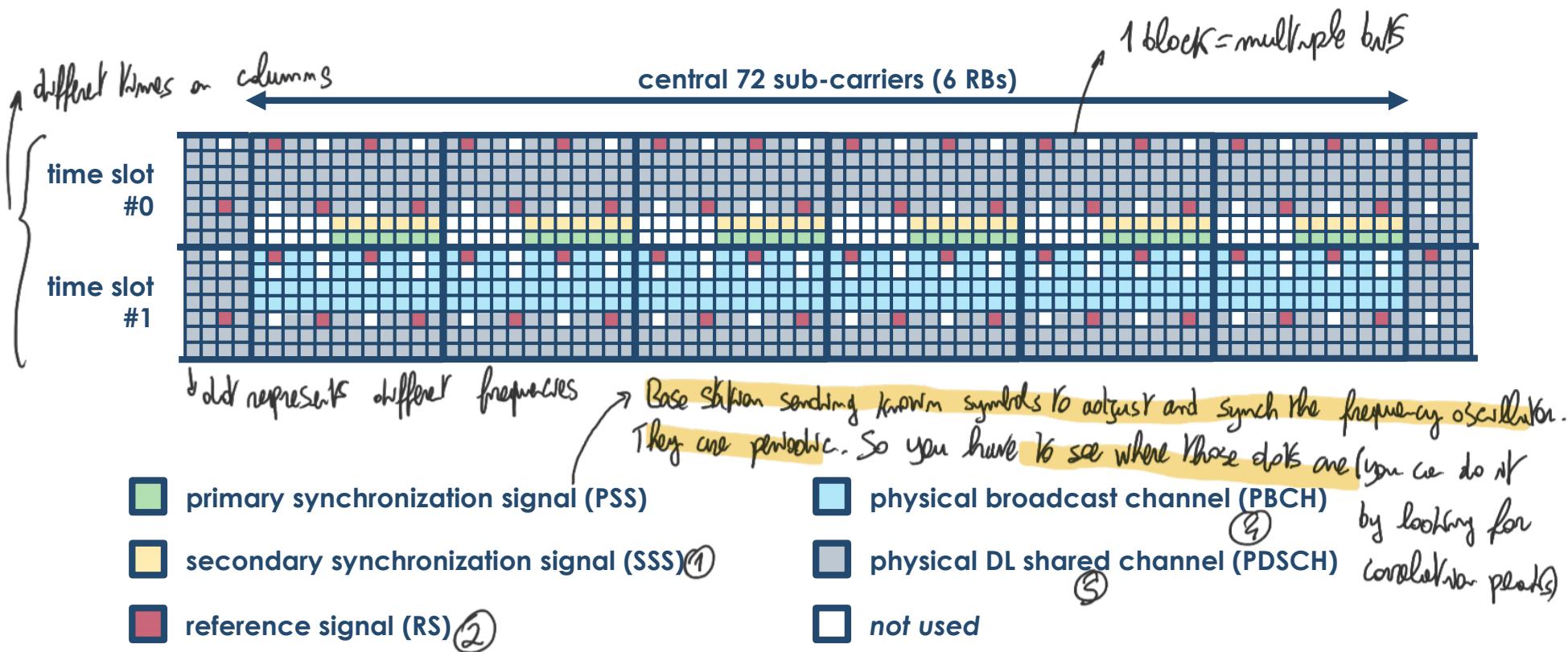
Questa domanda: Il segnale ricevuto non è periodico. Che possiamo fare? Farlo quasi periodico. Io tolgo la parte finale del segnale nel dominio del tempo e la metto al principio. Utilizzando risorse (SMP 37). Questo è abbastanza per farlo quasi periodico per fare l'approssimazione

$Y(f_e) = X(f_e)H(f_e)$ possibile. Abbastanza per implementare l'equalizzazione. Quanto tempo dovrebbe essere? Più lungo che la distanza temporale tra il percorso più veloce e il percorso più lento.

Quando si dice che $Y(f_e) = X(f_e) \cdot H(f_e)$, questo è vero solo nel caso di segnali periodici nel dominio del tempo. Quindi l'approssimazione è quasi perfetta con una buona accuratezza.

Example of resource element mapping

RE mapping for DL frame using the FDD mode (type I), normal CP, 10- or 20-MHz bandwidth, and single-input single-output configuration:



① For fine timing, you refine your estimation. With this you know the grid in the frequency domain. In the time domain you can have cyclic prefix for sync, using a correlation window. ② By the standards those are pilot symbols received for channel estimation. Then after some iterations you can refine the estimation (2nd round).

④ They contain general info: cell size, constellation size etc. All of those for the control plane. So you are aligned in frequency domain, time domain and estimation.

⑤ Those contain control symbols that tell you where your information symbols are. Then you have information bits.

⑥ Info about service provider, modulation etc. Once you decoded them you know what to look for.

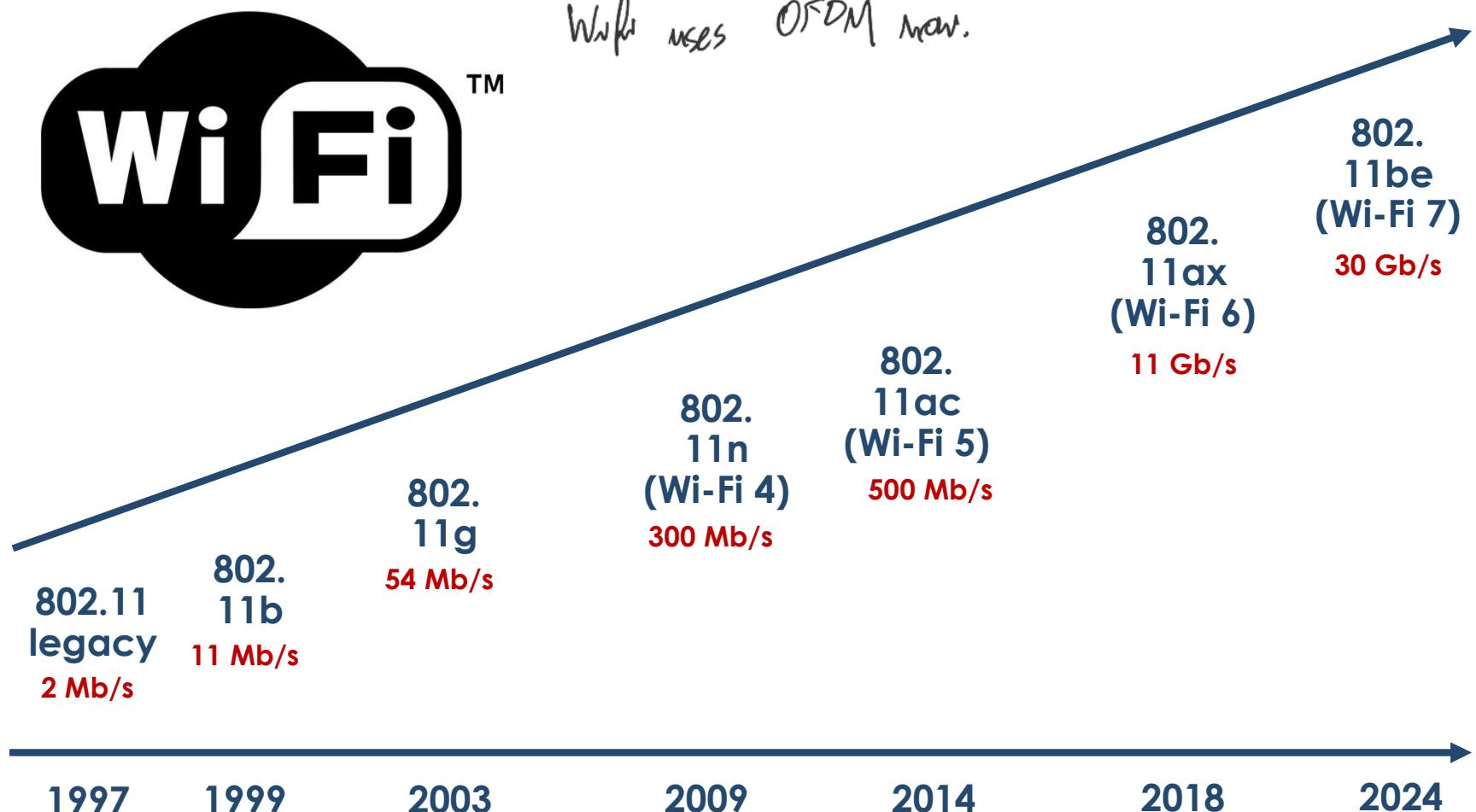
⑦ Are shared, some of the boxes are for control and others are for data, but you only know that when you decode ④.



An example of OFDM-based systems: WLAN standards (1/2)

Standard	Freq. range [GHz]	Throughput [Mb/s]	MA scheme	Max mod. order
802.11 legacy	2.4	2	DSSS/FHSS	BPSK
802.11b	2.4	11	DSSS	DQPSK
802.11g	2.4	54	DSSS	64-QAM
802.11n	2.4 / 5	300	OFDM	64-QAM
802.11ac	2.4 / 5	500	OFDM+ MIMO	256-QAM
802.11ax	2.4 / 5	11,000	OFDM+ MU-MIMO	1024-QAM
802.11be	2.4 / 5 / 6	30,000	OFDM+ CMU-MIMO	multiple antennas. n/a

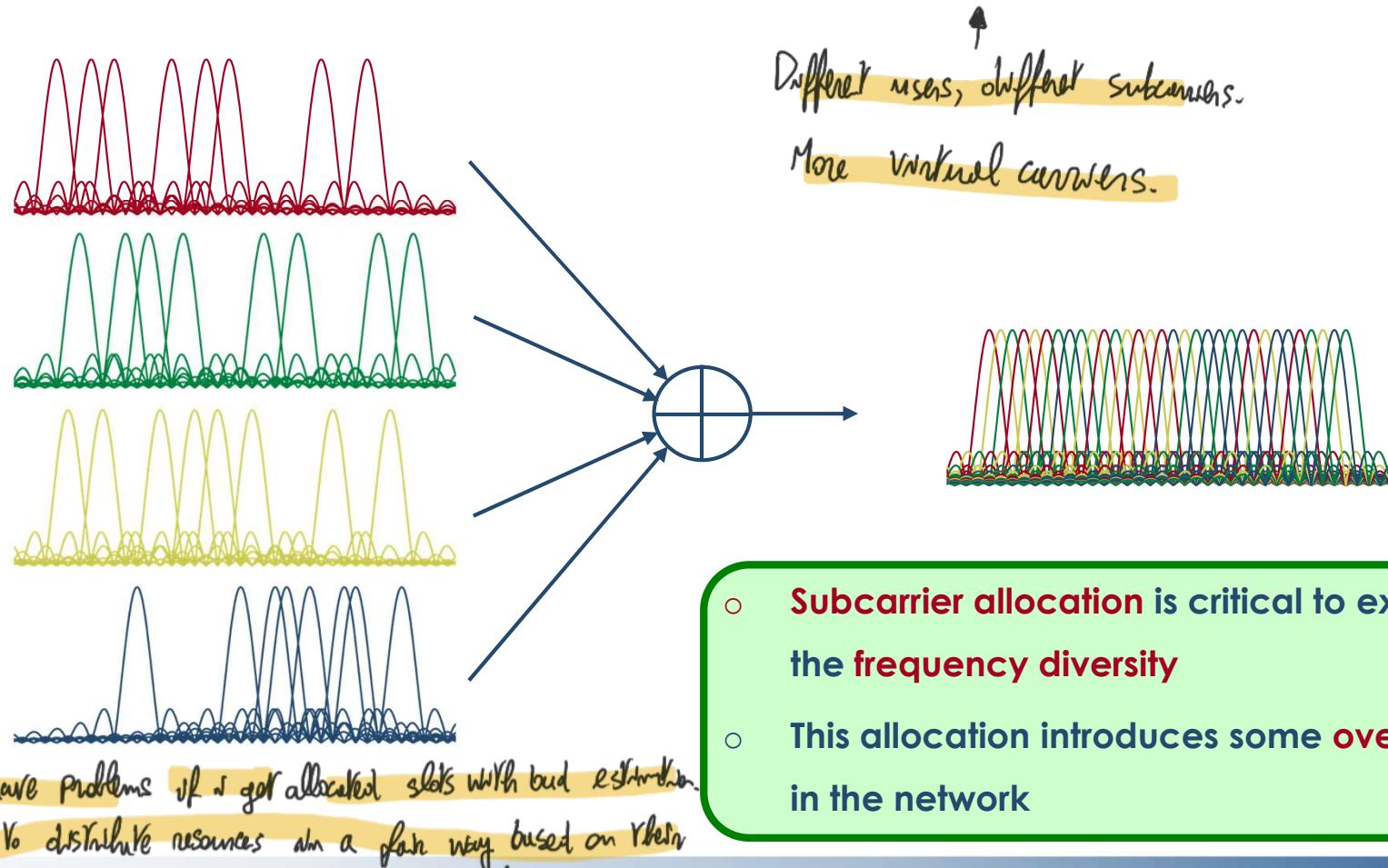
An example of OFDM-based systems: WLAN standards (2/2)



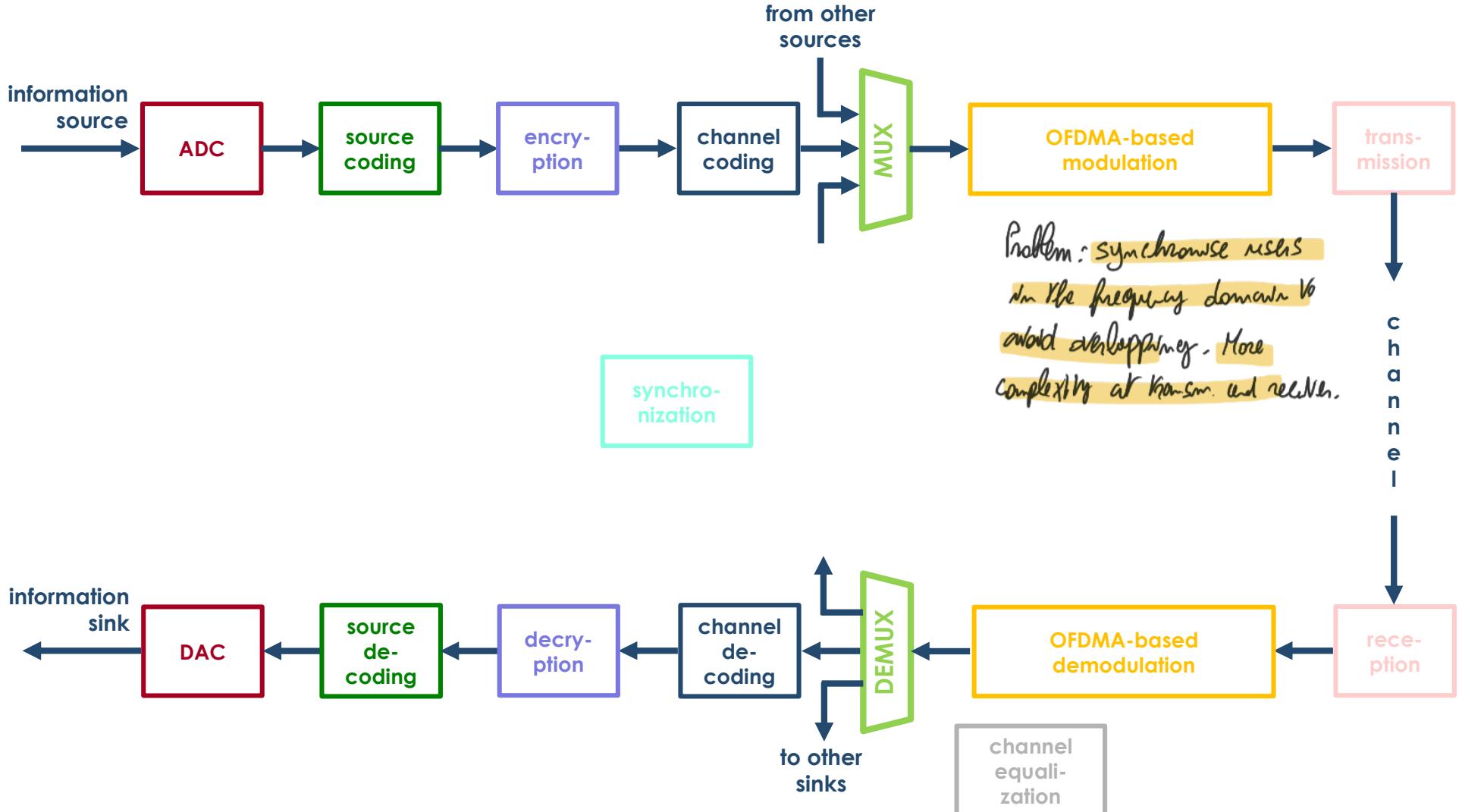
Orthogonal frequency division multiple access (OFDMA) (1/2)

How can we adapt the OFDM technology to the multiuser case?

Each user can be assigned a subset of subcarriers, by zeroing the inactive subcarriers



Orthogonal frequency division multiple access (OFDMA) (2/2)





Limits of OFDM(A) (1/2)

- **sensitivity to synchronization errors:**
 - **single-carrier systems:** the residual frequency offset ν must be $\nu \ll \frac{1}{T_b}$
 - **multicarrier systems:** $\nu \ll \frac{1}{T} = \frac{1}{NT_b}$
- **high peak-to-average power ratio (PAPR):** *Another issue:*
 - the **superposition** of N sinusoidal signals yields a large PAPR, thus calling for linear radio-frequency (RF) amplifiers
 - to improve the **efficiency** of the RF stage at the MSs, the uplink can adopt a modified version of OFDMA, called **single-carrier FDMA (SC-FDMA)**

When you move back to the time domain after having symbols in the frequency domain, you get something like:

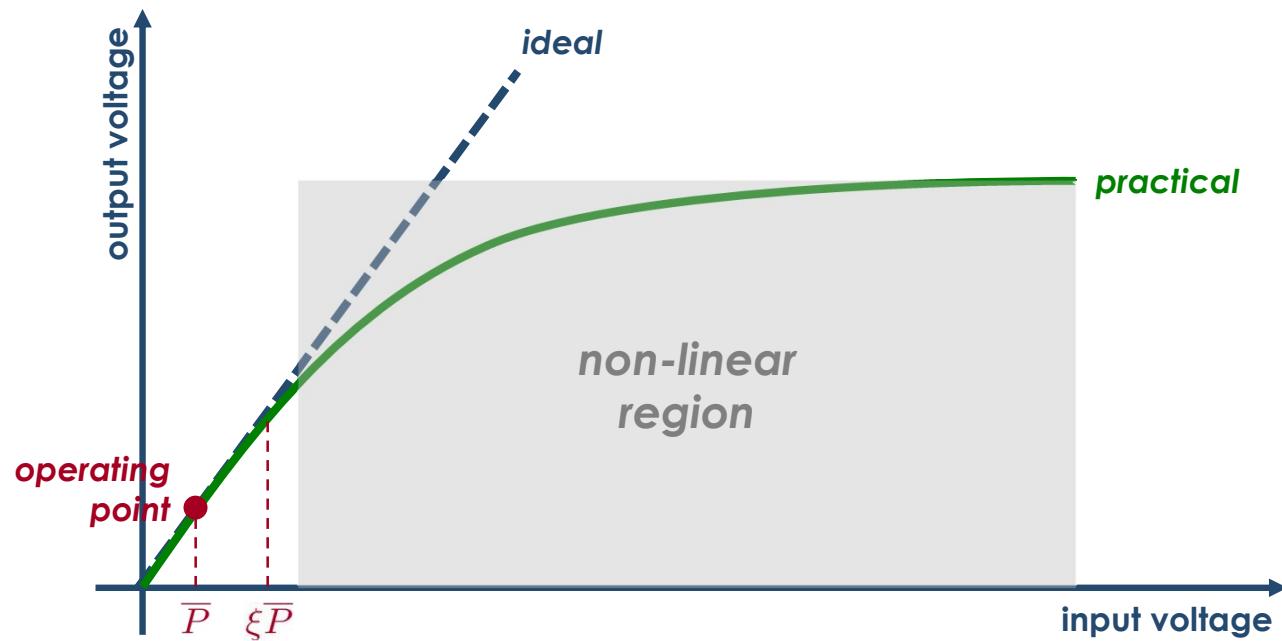


IFFT provides value that can vary a lot from the average. So you have to maintain this amplitude. So you need linear amplifiers

We have huge PAPR because of IFFT so we need to equip transmitters with linear amplifiers but we don't want that. So we do allocation in the time domain (you take your coefficients and transform

You add an additional scheme of FFT, add window cavers, do IFFT and get a reduced PAPR. And you keep all the advantages. You gain in PAPR.
M symbols in time domain, finish the job in frequency domain.

High PAPR in OFDMA:

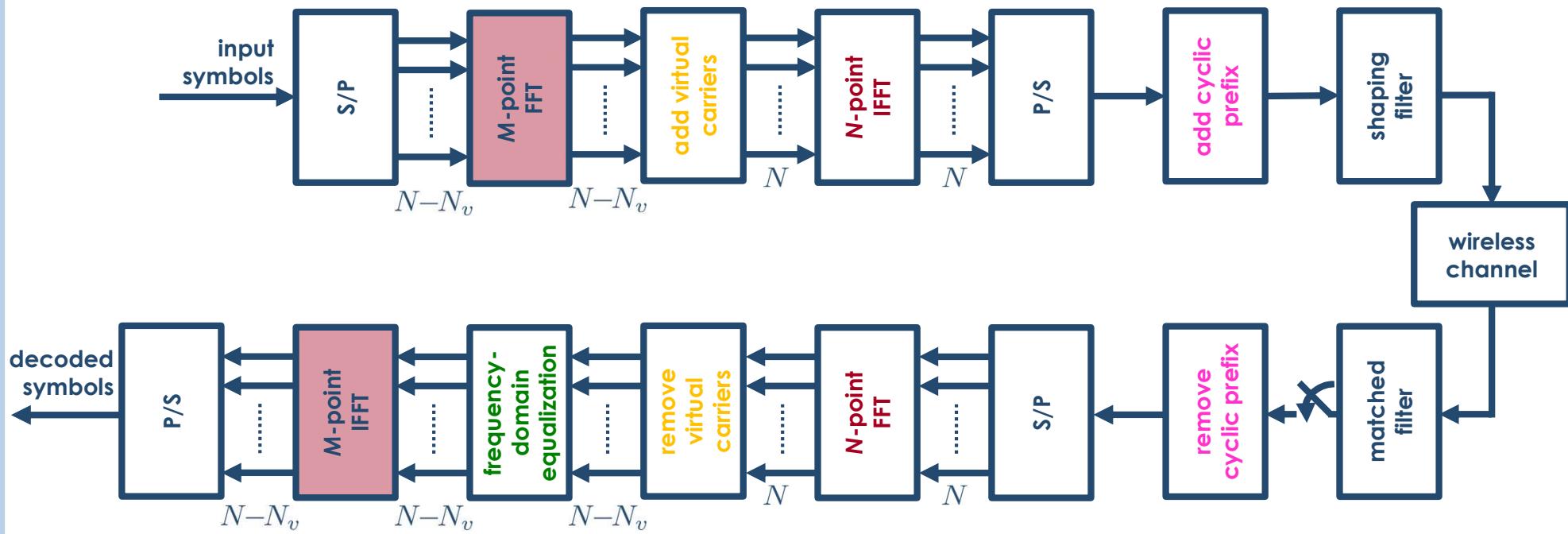


We then need either to transmit at a lower power (hence, reduced coverage) or to use linear yet inefficient high-power amplifiers (hence, reduced battery life)

Single-carrier FDMA (SC-FDMA) (1/2)

To significantly **reduce** the PAPR, we can modify the OFDMA scheme as follows, by implementing the so called **single-carrier FDMA (SC-FDMA)**:

In the downlink you don't have problems, you can use linear amplifiers.



LEZ. 15 END

Ok great! Now explain the difference with SC-FDMA trying to use my same language

- Alright, let's break it down using the same kind of reasoning and language we used for OFDMA. The difference between OFDMA and SC-FDMA comes down to **how we handle the frequency-domain symbols** and **what sequence we send in the time domain**.
-

OFDMA Recap

In OFDMA:

1. We take the frequency-domain symbols (e.g., A, B, C), which are directly assigned to subcarriers.
 2. We apply the IDFT to get a time-domain sequence a, b, c .
 3. This sequence a, b, c is sent as individual values, each on its own shaping pulse.
 4. The result is that the transmission is inherently **multi-carrier**, meaning multiple subcarriers (frequencies) are active at the same time.
-

SC-FDMA:

SC-FDMA tweaks this process to make the transmitted signal **look like a single-carrier signal**, even though it's still using multiple subcarriers. Here's how it works:

1. Frequency Domain Input:

We still start with symbols A, B, C , but before assigning them to subcarriers, we first perform a **DFT** on them. This gives us a new sequence of coefficients X_1, X_2, X_3 .

2. Subcarrier Mapping:

The X_1, X_2, X_3 values are then mapped to subcarriers (like A, B, C were in OFDMA).

3. IDFT:

Just like in OFDMA, we take the IDFT of these mapped values to get a time-domain sequence. Let's call it y_1, y_2, y_3, \dots



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3. IDFT:

Just like in OFDMA, we take the IDFT of these mapped values to get a time-domain sequence. Let's call it y_1, y_2, y_3, \dots .

4. Transmission:

This new sequence y_1, y_2, y_3 is sent over the air. Just like in OFDMA, each value y_i is transmitted using its own shaping pulse, spaced by T_b .

The Key Difference:

- **OFDMA** sends the IDFT of the original frequency-domain symbols A, B, C , so the time-domain signal is inherently multi-carrier.
- **SC-FDMA** first applies a DFT to the symbols A, B, C , which "spreads" their information across all subcarriers before the IDFT. This makes the time-domain sequence behave more like a single-carrier signal (hence the name).

Why This Matters:

SC-FDMA has a key advantage: **lower Peak-to-Average Power Ratio (PAPR)**.

- In OFDMA, the transmitted signal is a sum of multiple subcarriers, which can lead to high peaks in the waveform.
 - In SC-FDMA, the DFT step ensures that the signal behaves more like a single-carrier waveform, reducing these peaks and making it more power-efficient for transmission (especially for devices like mobile phones).
-

So in simpler terms:

- **OFDMA:** Each subcarrier is tied directly to a frequency-domain symbol.
- **SC-FDMA:** The symbols A, B, C are first "mixed together" via a DFT, so the time-domain sequence is smoother and looks like a single-carrier signal.



Single-carrier FDMA (SC-FDMA) (2/2)

Together with a valuable PAPR reduction, the SC-FDMA also shows the following features:

- the IFFT at the transmitter operates on the Fourier coefficients rather than on information symbols (as in OFDMA) (hence, computationally efficient implementation)
- the distinctive feature with respect to a traditional FDMA is the presence of the CP, that allows for frequency-domain equalization

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Okay and one last question: when we say that at the receiver we sample with the shaping pulse the signal, this means that we find where the autocorrelation is maximum so we know when we actually have a symbol and get the value?

- ⌚ You're very close, but let me refine that understanding a bit. When we say that at the receiver we "sample with the shaping pulse," we're essentially aligning the received signal with the shaping pulse to correctly extract the transmitted values. Here's how it works:
-

1. Symbol Timing Synchronization (*Finding When to Sample*):

- At the receiver, the first step is to determine the correct **timing** of the symbols.
 - This involves finding the point where the signal aligns best with the expected symbol duration.
 - One way to do this is by maximizing the **autocorrelation** of the received signal with the known shaping pulse $p(t)$. This tells us where the symbols are positioned in time.
-

2. Extracting the Symbol Value (*Sampling*):

- Once the timing is known, the receiver samples the signal at intervals of the symbol duration T_b , where the shaping pulse $p(t)$ has its peak contribution.
 - The value of the received symbol is extracted at these sampling points because that's when the contribution of the shaping pulse is strongest (and ideally, there's minimal interference from other symbols due to orthogonality).
-

Key Insight:

- The **autocorrelation** process helps with **synchronization** (knowing *when* to sample).
- The **sampling itself** extracts the value of the transmitted symbol at the right time (aligned with the peak of the shaping ).

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- The **autocorrelation** process helps with **synchronization** (knowing *when* to sample).
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-

Why This Works:

Shaping pulses like raised cosine or sinc are designed to be orthogonal at the sampling points. This means:

- At the sampling point for one symbol, all other shaping pulses (from earlier or later symbols) contribute zero interference.
 - This allows the receiver to extract the correct symbol value cleanly.
-

Summary:

Yes, you're correct in thinking that autocorrelation helps you find the timing (when the symbols occur), but the actual sampling step involves directly taking the signal's value at those aligned points, which represents the transmitted symbol.

