1.

a)

The spreading factor is the ratio between the chip rate and the bitrate:

$$SF = \frac{3,85}{0,55} = 7$$

The signal from station A, $\boldsymbol{SA}$, is:

+1	+1	+1	-1	-1	+1	-1	+1	+1	+1	-1	-1	+1	-1

The signal from station B, $\boldsymbol{SB}$, is:

-1	+1	+1	-1	+1	-1	-1	-1	+1	+1	-1	+1	-1	-1

b)

Due to different attenuation, the signal from station A is weaker. The sum of the arriving signals is $R = 0,5 \times SA + SB$, which corresponds to:

-0,5	+1,5	+1,5	-1,5	+0,5	-0,5	-1,5	-0,5	+1,5	+1,5	-1,5	+0,5	-0,5	-1,5

Each bit is decoded separately. Since both stations repeat the transmission of the logical bit, it is enough to decode one half of the received signal:

$$DA(0) = KA \cdot R(0 \dots 6) = +4,5 \geq +2 \rightarrow "1"$$
$$DB(0) = KB \cdot R(0 \dots 6) = -7,5 \leq -2 \rightarrow "0"$$

c)

The keys are not perfectly orthogonal, thus it is most probably a pseudo-noise sequence.

d)

Orthogonal codes assure that two encoded sequences encoded with different keys arriving at the receiver can be perfectly decoded, without interfering between each other, as long as the transmitted sequences arrive tightly synchronized and the received powers are not too different (otherwise, noise and interference can greatly hamper the weaker signal). This usually requires tight synchronization and power control. Besides, orthogonal codes are scarcer than pseudo-noise codes. On the other hand pseudo-noise codes are easier to get in significant numbers and do not require tight synchronization. Power control will be needed, since with pseudo-noise, the codes are quasi-orthogonal rather than perfectly orthogonal, which means that two simultaneously arriving signals will interfere with each other.

e)

Assuming that the employed code has good auto-correlation properties (internal product between of a code sequence with itself has high value if the sequences are synchronized and is approximately zero if they are out-of-synch), the RAKE receiver multiplies the received signal by different version of the code sequence, which start at different time intervals – i.e., with different delays between each other. In this way, the several multipath components of the signal are independently decoded and added together to produce a better quality output.

2.

a)

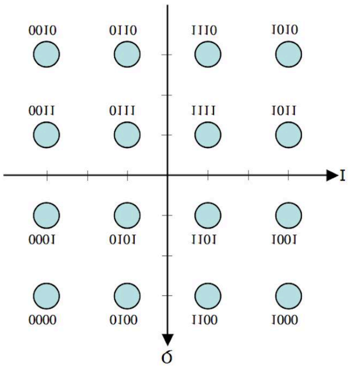
At SNR of 15 dB, the maximum number of modulation symbols that can be used is $M = 4$. This corresponds to the transmission of $L = \log_2(M) = 2$ bits per symbol. The achieved bitrate will be $R_b = L \cdot R_s = 2 \cdot 6000000 = 12$ Mbit/s.

b)

Yes, for example using 8-PSK (coherent PSK), which allows the transmission of 3 bits per symbol.

c)

i)

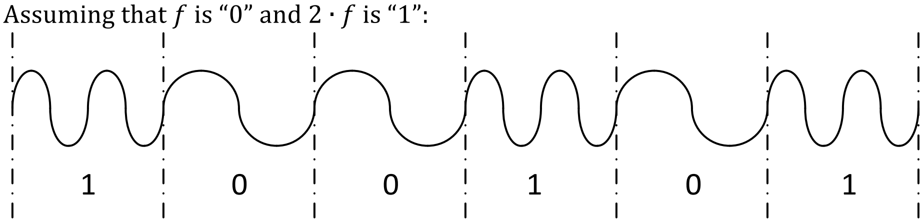


ii)

There are three different amplitude levels, since in each quadrant there are two symbols with the same amplitude.

d)

Assuming that f is "0" and $2 \cdot f$ is "1":



3.

a)

The theoretical maximum capacity is calculated with Shannon's formula, which needs the SNR and the bandwidth. In order to calculate the SNR, we need the received power. The problem description implies a log-distance path loss model, since it establishes the propagation model (free space) within a reference distance ($d0$), and indicates the path loss exponent beyond that distance (2,5). The first step is to calculate the wavelength, since then we can calculate the path loss at $d0$. The wavelength is $\lambda = \frac{300000000}{2400000000} = 0{,}125\ m$. We can now calculate the path loss at $d0$ using the Friis free space model:

$$PL0 = 20 \times log10\left(\frac{4 \cdot \pi \cdot d0}{\lambda}\right) = 20 \times log10\left(\frac{4 \cdot \pi \cdot 100}{\lambda}\right) \approx 80$$

Then, we can apply the log-distance model and calculate the received power:

$$\mathrm{Pr}[dBm] = Pt[dBm] + Gt + Gr - PL0 - 10 \cdot \alpha \cdot log10\left(\frac{d}{d0}\right) \approx 24{,}77 + 2 \times 15 - 10 \cdot 2{,}5 \cdot log10\left(\frac{3000}{100}\right) \approx -62{,}20\ dBm$$

In order to calculate the noise power, we need the bandwidth:

$$B = \frac{5100 - 100}{2} = 2500\ kHz$$

The noise power is then calculated as follows:

$$N = N0 \cdot B = 10^{\frac{-170}{10}} \cdot 2500000 = 2{,}5 \times 10^{-11}\ mW$$

We can now apply Shannon's formula to calculate the capacity:

$$C = B \cdot log2\left(1 + \frac{Pr}{N}\right) \approx 2500000 \cdot log2\left(\frac{10^{\frac{-62{,}20}{10}}}{2{,}5 \times 10^{-11}}\right) \approx 36{,}39\ Mbit/s$$

b)

Since we have calculated the received power in a), which is $Pr \approx -62{,}20\ dBm$, it is now a question of comparing this power with the receiver sensitivities of the different transmission modes. The maximum bitrate that can be used corresponds to 4 Mbit/s.

c)

$$Ae = G\frac{\lambda^2}{4\pi} \approx 0{,}039\ m^2$$

$$Aphy = \frac{Ae}{\mu} = \frac{Ae}{0{,}8} \approx 0{,}049\ m^2$$

d)

R=1 Mbit/s -> $\frac{R}{B} = \frac{1000000}{2500000} = 0{,}4\ bit/Hz$

R=2 Mbit/s -> $\frac{R}{B} = \frac{2000000}{2500000} = 0{,}8\ bit/Hz$

R=3 Mbit/s -> $\frac{R}{B} = \frac{3000000}{2500000} = 1{,}2\ bit/Hz$

R=4 Mbit/s -> $\frac{R}{B} = \frac{4000000}{2500000} = 1{,}6\ bit/Hz$

4.

a)

$$Packet\ size\ (l) = 8\ kbps \times 40\ ms = 320\ bits = 40\ bytes$$

$$T_{packet} = DIFS + Backoff + PHo + \frac{(MACh + RTP/UDP/IP + data)}{R} + SIFS + PHo + \frac{ACK}{R} =$$

$$= 0.034 + 0.067 + \left(0.096 + \frac{34\times8 + 40\times8 + 40\times8}{11000}\right) + 0.016 + 0.096 + \frac{14\times8}{11000} = 0.402ms$$

$$Throughput = \frac{40\times8}{0.402}\ kbps = 795{,}840\ kbit/s$$

b)

$Nstreams = \lfloor 795{,}840/8 \rfloor = 99\ streams$ → Since each call is bi-directional → 49 telephones

c)

$FER_{MAC} = (FER_{PHY})^{1+7} = 0.04^8 = 6.554\times10^{-12}$