Royal Institute of Technology

Project in Wireless Communications
EQ2440

Project Report

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Chapter 1

Results and discussion

1.1 Signal-to-Noise Ratio and Channel Capacity

There is noise and imperfections in the channel that cause errors in the transmission. We have used two different approaches to measure the magnitude of the noise in comparison to the signal. The first approach was to measure the variance of the received signal while nothing is transmitted. With the noisy samples $n[k]$ for $k = 0, ..., N$ with estimated mean $\hat{\mu}_n$ we estimate the variance as

$$\hat{\sigma}_n^2 = \frac{1}{N-1} \sum_{k=1}^{N} (n[k] - \hat{\mu}_n)^2 \tag{1.1}$$

We then find the maximum amplitude A_m of a sinusoid that can be received from the channel. Then we can calculate the signal-to-noise ratio

$$\mathrm{SNR} = \frac{A_m^2}{2\hat{\sigma}_n^2} \tag{1.2}$$

Since we cannot know if the noise have the same characteristics when the signal is present we used a second approach to which measure the signal-to-noise-and-distortion ratio (SINAD). We again receive samples $r[k]$ for $k = 0, ..., N$ of a transmitted sinusoid with known frequency ν_c. The received signal is a sinusoid apart from some unknown error $n[k]$. We can estimate the error by least square fitting a sinusoid $a_c \cos(2\pi\nu_c k) - a_s \sin(2\pi\nu_c k)$ with the free parameters (a_c, a_s) to the samples $r[k]$. An example of the method can be seen in Fig.1.1. We can then use

$$\hat{n}[k] = r[k] - a_c \cos(2\pi\nu_c k) - a_s \sin(2\pi\nu_c k) \tag{1.3}$$

as an estimation of the noise and calculate the variance of this noise according to (1.1). This variance is then used to calculate

$$\mathrm{SINAD} = \frac{a_c^2 + a_s^2}{2\hat{\sigma}_n^2} \tag{1.4}$$

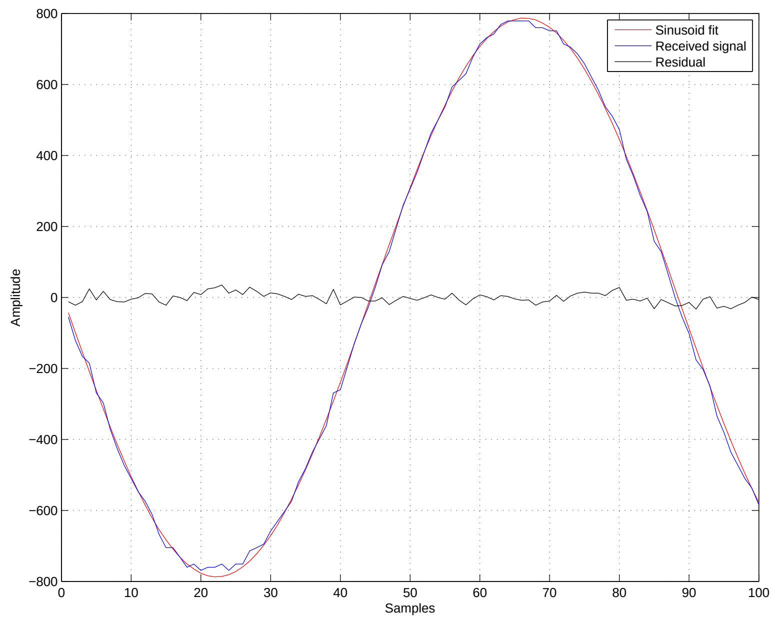

Figure 1.1: Example of a received sinusoid with frequency of 500 Hz together with sinusoid fit.

For any channel there is a maximum achievable rate called the channel capacity. In the case of a real valued AWGN channel with bandwidth W this capacity can be shown to be

Do we need a reference for this?

$$C = \frac{W}{2}\log_2(1 + \mathrm{SNR})\ \text{[bits/s]} \tag{1.5}$$

1.2 Signal-to-Noise Ratio Measurements

Both measurement methods were evaluated on received signals of length $N = 44.1 \times 10^4$ samples corresponding to a signal of 10 seconds. For the SINAD measurement a sinusoid with frequency 11025 Hz ($F_s/4$) was used. The histogram of the residual noise in the SINAD measurement can be seen in Figure 1.2. This figure show that the noise have a distribution of Gaussian character. The resulting values of SNR and SINAD measurements can be seen in 1.1.

Method	**Measured value** [dB]	**AWGN capacity**[kbps]
SNR	51.19	218
SINAD	29.71	375

Table 1.1: Results of measurements of SNR with the two different approaches together with the resulting channel capacity.

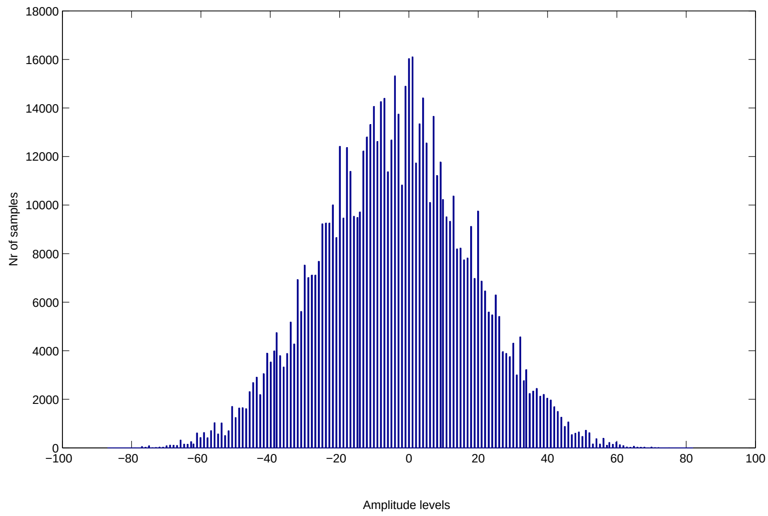

Figure 1.2: Histogram of residual noise on transmitted sinusoid in SINAD measurement.

1.3 4-FSK

In the implementation, 35 samples per symbol and the frequencies $kF_s/24$ for $k = 1, 4, 7, 10$ were used. This resulted in a rate of 2520 bps with no errors. This value for the rate considers all the bits that were transmitted, including guard band and training sequence in total consisting of 400 bits. Hence, the effective rate is lower than this value but still well above the basic requirements of 1 kbps. In Figure 1.3 plots of the transmitted spectrum is shown with rectangular and Gaussian window.

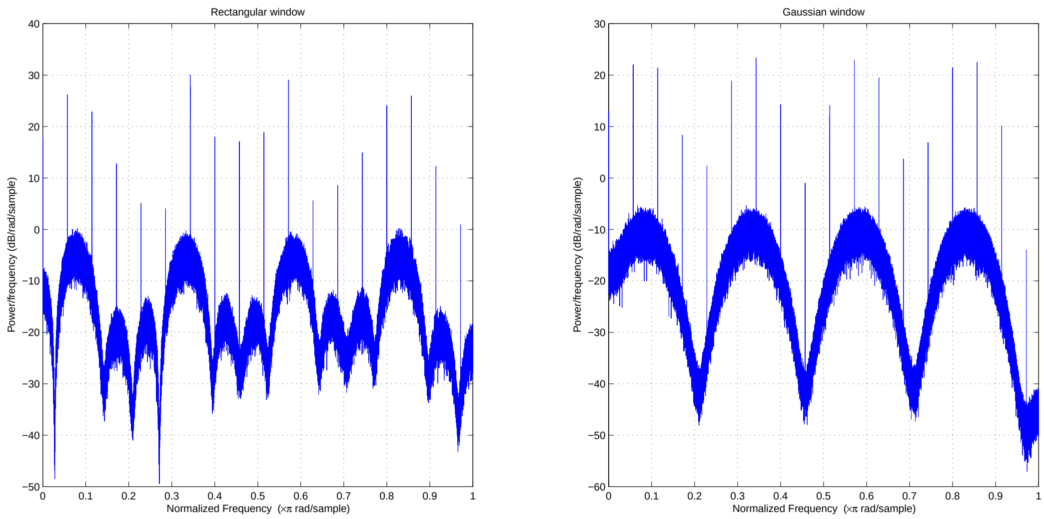

Figure 1.3: Spectrum of the transmitted 4-FSK signals with rectangular and gaussian window.

In order to improve the rate and more efficiently exploit all the available bandwidth, a frequency division multiplexing (FDM) system was used. The idea is to put the frequencies in the 4-FSK scheme closer together and divide the spectrum into N_r parallel channels. $N_r = 7$ was the highest number of channels that the system could demodulate reliably i.e 0 % BER. However, it was necessary to lower the rate by increasing n_{sym} to 88 samples per symbol. The resulting rate was

$$R = \frac{N_r \times log_2(4)}{n_{sym}} \times f_s = 7 \times \frac{2}{88} \times f_s = 7015 \text{ bps}$$

Hence, this approach more than doubles the rate that was achieved with the single 4-FSK system. A Gaussian pulse shape was used to reduce the interference between frequencies close to each other. We tested the system to find the minimum number of samples per symbol needed for 0 % BER with $N_r = 2, 4, 7$. The results can be found in Table 1.2. The transmitted power spectrum for the case $N_r = 4$ is depicted in Figure 1.4.

N_r	Samples per symbol	Rate [bps]
2	37	4767
4	63	5600
7	88	7015

Table 1.2: 4-FSK + FDM results for different values of N_r

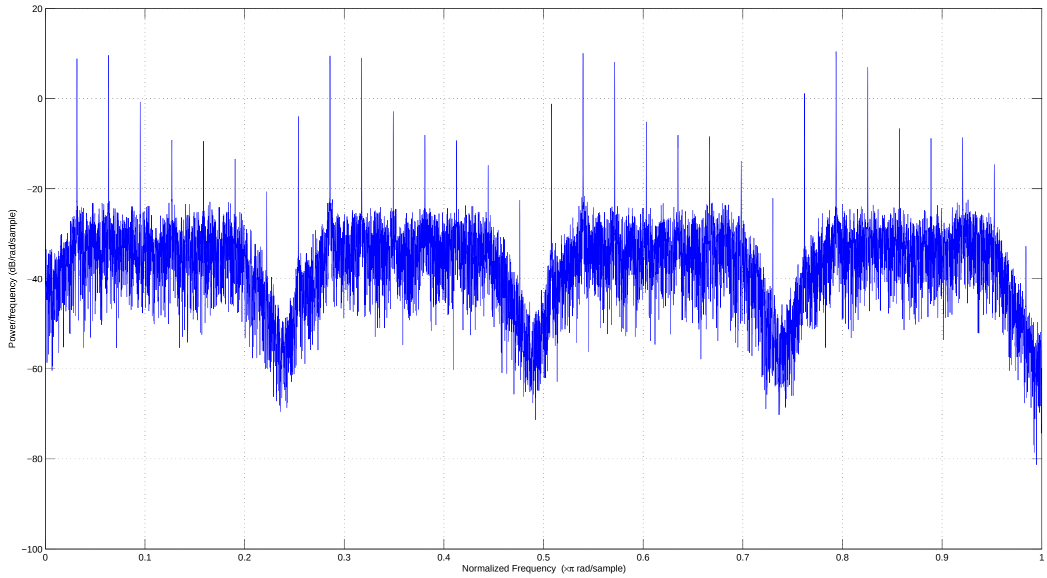

Figure 1.4: Spectrum of the 4-FSK signal extended to 4 FDM.

1.4 M-QAM

In this implementation we found the best performance with 64-QAM and 8 samples per symbol. We used a carrier frequency of 11025 Hz and a Gaussian pulse shape for the modulation which gave us an effective rate of 28.6 kbps when transmitting a file of 10 kB. The power spectrum of the transmitted signal and the received constellation is shown in Figures 1.5 and 1.6. Reducing the number of samples per symbol to 7 widens the main lobe of the signal such that its bandwidth exceeds the bandwidth of the system. This and possibly the effect of ISI made it impossible to demodulate such a signal without getting errors.

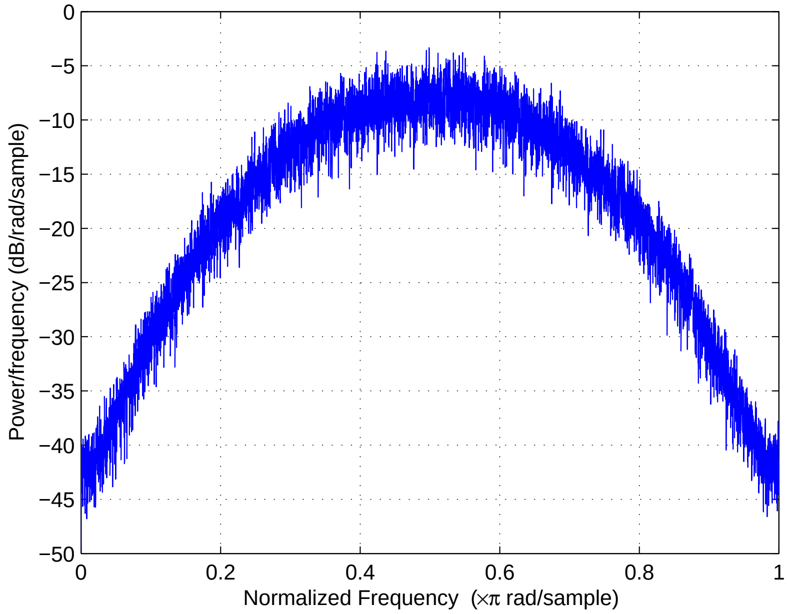

Figure 1.5: Power spectrum of the transmitted 64-QAM signal with gaussian window.

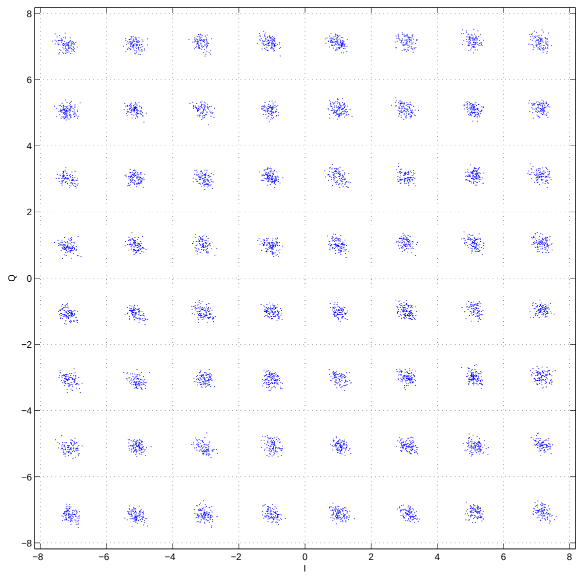

Figure 1.6: Received constellation after offline transmission with 64 QAM.

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