# **Project GFS: The GFSK Receiver**

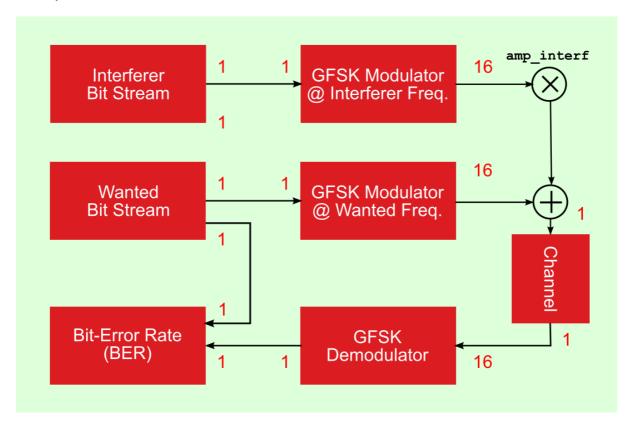
This project is a compulsory part of the examination for the <u>Implementation of Digital Signal Processing</u> course at the University of Twente. The goals of this project are:

- To exercise in fixed-point optimizations.
- To apply various aspects of the course on a larger design considering area, time and power tradeoffs.

### **Preparation**

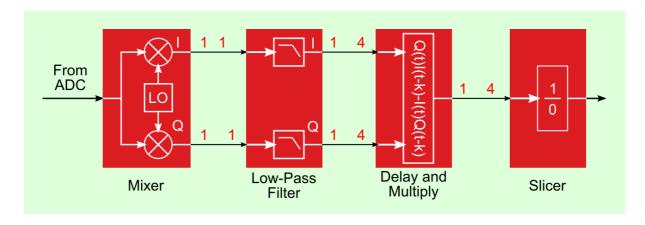
Read the document <u>Some Background on GFSK Modulation</u> carefully. It contains the theory and some practical remarks that you will need in the rest of this project. Consult also the lecture slides on GFSK modulation as presented as part of the CORDIC theory.

The GFSK system considered looks as follows:



As compared to the setup presented in the CORDIC lecture, the system has two transmitters. The first is transmitting at the wanted carrier (IF) frequency 1 MHz. The other is an interferer signal, for example originating from a device operating in another radio channel; it transmits at a frequency of 3 MHz. The wanted signal has an amplitude of 1. The interferer has an amplitude amp\_interf that can be controlled by the testbench user. This amplitude is zero by default, which amounts to the interferer not being present. Interferers are quite common in radio communication. Many standards have multiple channels in some frequency band. The systems should be designed in such a way that communication in adjacent channels should not interfere too much.

The system setup acts as testbench for the GFSK Demodulator, the design of which is the topic of this project. As explained in the background document, the demodulator is built from a chain of 4 signal-processing blocks, as follows:



#### **Files and Directories**

Go to your home directory and fetch the files for this project with:

get-module gfs gfs

*Note:* The first argument of get-module is the project name, the second the name of subdirectory in your file system. So, by issuing the command several times with a different second argument, you will be able to make multiple copies of the distribution.

Three subdirectories of gfs will be created:

- arx: this subdirectory contains the source files in Arx and the *makefile* for creating their C++ and VHDL models. Note that each block in the demodulator chain is given by a separate Arx file and needs to be compiled separately. It is up to the C++ and VHDL testbenches to put the 4 Arx designs together.
- cpp: this directory contains the generated C++ files along with the files that are used to build the C++ testbenches. They are based on <u>IT++</u>.
- vhd1: this directory contains the generated VHDL files as well as files to support synthesis and simulation.

A multitude of files are involved in this project. They will be presented gradually, at the moment that they will be needed.

Change to subdirectory arx and run make. This ensures that the database is consistent by generating the C++ and VHDL versions of the Arx source files present in the subdirectory. At this moment, you don't need to understand contents of the source files.

Now change to directory cpp and run make in that directory as well. You can ignore warnings originating from the IT++ source files.

The cpp directory has IT++ code for both the modulation and demodulation as well as the C++ code corresponding to the four demodulator blocks. After running make 5 testbenches (5 compiled executables) are created from a single generic source file tb\_ber.cpp:

- tb\_ber\_itpp: This testbench does not instantiate any Arx blocks; the entire system is using IT++ models.
- tb ber mixer: This testbench instantiates the Arx *mixer* and keeps the other IT++ models.
- tb\_ber\_filter: This testbench instantiates the Arx *low-pass filters* next to the already instantiated mixer.
- tb\_ber\_demod: This testbench instantiates the *delay-and-multiply* block from Arx next to the blocks already instantiated in the previous testbench. It may be confusing, but the term *demodulator* is used sometimes to refer to just the delay-and-multiply block and sometimes to the entire chain from mixer to slicer.
- tb\_ber\_slicer: This testbench uses Arx code for all four blocks of the demodulator and is therefore the testbench to be used for the final design. The other testbenches are useful for intermediate

versions of the design.

Each executable will perform a *bit-error rate* (BER) simulation as follows. The noise level is given by the *signal-to-noise ratio* (SNR). The wanted signal has an amplitude of 1 and will not change throughout the entire project (as opposed to a real-life situation where the received signal can vary many orders of magnitude and *gain control* in the analog front end keeps the signal in a relatively small range). The SNR is computed with respect to that fixed wanted signal level. So, SNR = 0dB amounts to having as much signal power as noise power in the frequency band of interest. In a loop the SNR is gradually increased (the noise power is decreased) in steps of 1 dB until the BER drops below 1.0e-4 (one error in 10.000 bits). At a specific SNR level, the iteration is aborted either when a maximum number of errors (1000) is reached or a maximum number of bits have been simulated (100000). If an SNR level is found with BER less than 1.0e-4, the *sensitivity* level of BER = 1e-3 is estimated by interpolation.

*Note*: In this project description, the words *bit* and *symbol* are used as equivalent notions as GFSK transmits one bit per symbol.

As a sanity check, run all executables by typing the executable name after the shell command prompt. Each simulation will display an SNR level, the BER at that level, and between parentheses the number of bits used for the computation of the BER. If everything went well, all simulations will report a sensitivity level of about 10.8 dB. Throughout the entire project this value will be a reference for the quality of the design. Any design you make should keep the sensitivity within 0.5 dB of this value, so below 11.3 dB.

The testbench executables all support the same command-line options:

- -d (dump mode, no arguments). This file will dump all kind of intermediate data to file. Only use this option for a limited number of bits (so in combination with option -b) as you will create very large files otherwise.
- -s <float val> (SNR level). When specified, only one SNR level is simulated. There will not be a search for the sensitivity level.
- -b <int val> (number of bits). The simulation will stop after the supplied number of bits (symbols) instead of the default value of 100000.
- -o <int val> (slicer offset). This sets the *slicer-offset value* necessary for a correct synchronization, see also the background documentation. The supported values are 0, 1, 2 or 3. Optimal values for all 5 testbench versions are built in after being specified in file tb\_ber.h.
- -p <int val> (symbol offset/pipeline depth). This sets the number of symbols (bits) to be ignored at the beginning of the simulation due to the latency in the system. Optimal values for all 5 testbench versions are built in after being specified in file tb\_ber.h.
- -i <float val> (interferer amplitude). This controls the value of amp\_interf mentioned above.

Each of the 5 testbenches writes files in an output directory of its own, named output\_slicer, output mixer, etc. The following files are written:

- bits.err: This is the only file that is written without the dump-mode option. It writes BER-SNR pairs and could be used to plot the behavior of the BER vs. SNR.
- bits.in: This file contains the bits as transmitted in the wanted channel.
- bits.out: This file contains the received bits.
- dam.out: The output of the DAM block, fixed-point represented as floating point.
- lpf\_i.out: The *in-phase* output of the low-pass filter, fixed-point represented as floating point.
- 1pf q.out: The *quadrature* output of the low-pass filter, fixed-point represented as floating point.
- mixer\_i32.in: The mixer input (ADC output) in Arx integer format, to be used as input for VHDL simulations (optional).
- mixer.in: The mixer input (ADC output) in fixed-point format, represented as floating point numbers.
- mixer i.out: The *in-phase* output of the mixer, fixed-point represented as floating point.
- mixer q.out: The quadrature output of the mixer, fixed-point represented as floating point.
- mod.out: The modulator output, the received signal before applying the quantization due to the ADC, fixed-point represented as floating point.

• slc.out: The slicer output, the output bit stream oversampled by a factor 4, where a 0 is represented by -1. Using -1 instead 0 may be of visual help when plotting the DAM output and slicer output in the same graph.

#### **Exercise GFS-1: Familiarization with GFSK**

Study the background-information document in detail and try to match it to the IT++ models of both transmitter and receiver as given in files gfsk.h, gfsk.cpp, tb\_ber.h, and tb\_ber.cpp. The simulation operates bit by bit. For each bit, the modulation function is called. Due to Gaussian filtering which spans multiple bits (how many?), the modulator needs to keep track of its history. Also at the demodulator side, the simulation advances in steps of one bit and history matters e.g. for the FIR filters.

Now run the following simulation (SNR = 100 dB basically means that there is no noise):

```
tb_ber_itpp -d -b 1000 -s 100
```

Then, launch Matlab (command matlab) and in Matlab, plot the frequency spectrum of the modulator output with:

```
load output_itpp/mod.out
periodogram(mod,[],2^13,8e6)
```

Inspect the spectra of intermediate signals working your way step by step in the signal processing chain. Look at the ADC output, mixer outputs and LPF outputs. Explain what you observe referring to the theory.

Inspect the time signals dam.out and slc.out as well as the input bit stream. Do you recognize the input bit stream in the slicer output? How much latency does the system have (how much "garbage" comes out of the slicer before the first input bit is output)? How large is the amplitude of dam.out and how does this relate to the theory presented in the background document?

Repeat the simulation with an interferer amplitude of 1.0. Comment using the spectra as well as the time-domain plots.

#### **Exercise GFS-2: Bit-Error Rate Simulations**

The correct synchronization is essential for the BER performance. Two parameters that matter are the system latency, pipeline depth (for the bits stored on behalf of BER calculations) or *symbol offset* and the *slicer offset*. The optimal values for these have been hard-coded in file tb\_ber.h but can be overridden with the -p and -o command-line options. For testbenches tb\_ber\_itpp and tb\_ber\_slicer, investigate neighboring values to the optimal settings. In some cases, the BER will be degraded, in others the BER will collapse completely and no sensitivity level will be found. Give a few examples of both cases documenting the sensitivity numbers for each of the selected settings.

The lesson learned is that synchronization is crucial for performance. So, if you modify the design later on in a way that the latency is affected, you will need to find the optimal offset values again (inspecting the DAM and slicer outputs in the time domain may give a good clue).

# **Exercise GFS-3: Arx Implementation of Demodulator**

Inspect the Arx implementations of all four blocks in the demodulator and compare them to their models in IT++. Describe the main differences in a few sentences per block. Perform the following two simulations:

```
tb_ber_itpp -b 100 -s 100 -d
tb ber slicer -b 100 -s 100 -d
```

Plot the dam.out signals of both simulations in one graph. You should see two main differences. Mention and explain them.

For each Arx block estimate the number of flipflops used from the source code. Report and motivate the numbers.

# **Exercise GFS-4: Demodulator in Arx: VHDL Synthesis**

The distribution of this project comes with support for VHDL. An entity gfsk.vhd is provided that instantiates all blocks coming from Arx as well as a clock generator for the different clock rates necessary. Word lengths are centrally administered in a file pk\_gfsk.vhd. The clock-generator is a "quick-and-dirty" solution. In a real design, clock division would be implemented in a more robust way (not relevant for this project).

A testbench is provided for a standalone VHDL simulation. You can run a Questasim simulation if you are curious. As opposed to what you are used, no configuration is provided as there is just one architecture for each entity. The compulsory part of the VHDL work concerns synthesis. Synthesize the design using command srun generate-design and inspect the log file when synthesis is ready. The log file contains the size of each of the four blocks in the demodulator as well information on its number of flipflops. Create a table that shows for each block:

- its area.
- its number of flipflops,
- its average area per flipflop.

Is the number of flipflops a good indicator for the area of a block?

# **Exercise GFS-5: Fixed-Point Optimizations**

Execute the following simulations:

```
tb_ber_itpp -i 7
tb_ber_itpp -i 10
```

You will observe that the BER performance behaves well at interferer amplitude of 7 but not at 10. Now, edit file tb\_ber.h and increase the values for MIXER\_WORD\_LENGTH and MIXER\_IWORD\_LENGTH by one. The values concerned are already in the file and just need to be commented out. Recompile the testbenches by running make and repeat the above simulations. If everything went well, you should observe that the system can deal with interferer amplitude 10.

This is, however, not true for the testbenches that incorporate Arx designs. Obviously, robustness against an interferer was not taken into account when choosing the fixed-point formats in the Arx code. A redesign action is needed. In Arx, the fixed-point word lengths are provided in Arx code. Go to the Arx directory and modify their fixed-point parameters, making your way block by block from mixer to slicer and using the appropriate testbench in the cpp directory. Optimize the fixed-point parameters of each block and try to arrive in this way (close) to the smallest design which can deal with interferer amplitude 10. Each time you modify Arx code, you will need to run make in the arx and then the cpp directory. The goal is, of course, not to make word lengths larger than necessary.

Summarize the results of all your experiments. For each of the 4 blocks, present the observed sensitivity for the fixed-point parameters that you have tried including the parameter sets for which the sensitivity goal was no longer met.

#### Hints:

• The IT++ models are in floating point and are therefore much more robust. Work on one Arx file at a time and use the appropriate testbench to connect to the IT++ signal chain at the right point.

- To diagnose problems, think of inspecting dumped signals in the time and frequency domains. When relevant, include plots in your report.
- Keep the parameters as used in IT++ consistent with the generics used in Arx.
- Be aware that in the CORDIC implementation of the mixer, the phase is a fixed-point number the length of which can be optimized.

When you have optimized all fixed-point parameters, go the vhdl directory and synthesize the design. As you have changed the word lengths, you will need to modify the values in file pk\_gfsk.vhd to reflect the changes before synthesis. Check your log file for errors and warnings. If you are sufficiently convinced that the synthesis was successful, make a new table with the area and flipflop count per block, as you did for GFS-4.

#### **Exercise GFS-6: Polyphase Implementation of the FIR Filters**

The demodulator contains downsampling (by a factor 4) directly after low-pass filtering. This means that a *polyphase* implementation, exchanging the order of filtering and downsampling is possible (see also the lecture slides).

Consider the advantages and disadvantages of a polyphase implementation in this concrete case. You can involve the following elements in your reasoning: one-to-one implementation vs. scheduled solution, filter symmetry, multiplierless design, power-area-time trade off. Do not write any code at this time. What is your final recommendation regarding the polyphase implementation? Should it be used or not?

#### **Exercise GFS-7: Scheduled Solution**

The reference design is a one-to-one implementation where the sample frequency of 8 MHz is equal to the system clock frequency. Suppose that the technology available to you allows a system clock of 80 MHz and you can do arithmetic operations (add, multiply, etc.) in one clock cycle.

Sketch on paper a data path, a bit in the style of Project MAP, that:

- can perform the entire GFSK reference design for the receiver;
- is flexible enough to execute alternative algorithms of comparable complexity.

First estimate the complexity of this design: how many additions, shifts, multiplications, etc. would a data-flow graph of the entire receiver have? How would those numbers translate in a minimum of functional units (adders, multipliers, etc., FUs)? Involve the fixed-point formats to determine the word lengths that the arithmetic blocks should have.

By preference, the solution should look like the data path of a single VLIW processor which can handle the entire demodulation. Make use of multi-ported register files from which FUs can read their inputs and to which FUs can write their outputs. An extreme case of such a multi-ported register file would allow all FUs to read and write their data simultaneously from the same register file.

You do not need to be concerned with the controller. You can, for example, assume that the controller is a sequencer that, for each clock cycle, reads a pattern of control signals from an external memory (such a pattern can be considered an "instruction"). This means that the data path can be "programmed" by changing the contents of this external memory.

You do not need to provide all details. A rough sketch in combination with a motivation of your design choices and design dimensioning is sufficient.

#### **Exercise GFS-8: Free Design Assignment**

Make, depending on the time left, minor or major modifications to the Arx code for the GFSK receiver. Follow any of the suggestions below, a subset or a combination or just do something completely different.

- Replace the CORDIC by a dedicated mixer for 1 MHz; this is a simple modification which has the disadvantage of sacrificing the ability to compensate for variations in IF.
- Investigate alternatives for the implementation of the FIR filter: transposed form instead of direct form, the use of multiplier blocks (see reference [Lan02] on the main web page of this course), etc.
- Consider a polyphase implementation in combination with hardware sharing for the I and Q channels (a polyphase implementation will reduce the clock speed used in the filter by 4; if the clock is then doubled, the computations for the I and Q channels can be done on the same hardware in an interleaved fashion).
- Implement (part of) the architecture that you have proposed in Exercise GFS-7.

As mentioned earlier, the constraint on your design should be to keep the BER performance degradation within 0.5 dB with respect to the original design. Be aware that changing the number of registers in the signal path may affect the system latency. If this is the case in your design, you will need to find the optimal synchronization parameters.

You may need to modify the C++ testbench files. Try to keep modifications to a minimum and make sure that Arx blocks are directly connected to each other, so without performing intermediate computations in the testbench.

Once your design is ready, run synthesis in order to have an impression of the area and timing. You may synthesize the entire receiver, but you may also consider synthesizing each component separately.

For debugging Arx using generated C++, consult the debug "points of attention" mentioned in the description of Project MAP. As a last resort, you could try to debug the generated VHDL.

#### **Deliverables**

Write a short report always motivating your choices and explaining the way you have reached your answers. Particular points of attention:

- GFS-1: Short explanation of modulator and demodulator, illustrative plots from Matlab, answers to the questions in the text.
- GFS-2: Results of BER simulations.
- GFS-3: Answers to the questions, requested plot, requested numbers.
- GFS-4: The requested table on the synthesis results and your comments.
- GFS-5: All new fixed-point values supported by plots and BER simulation results. The table with synthesis results.
- GFS-6: The requested reasoning.
- GFS-7: The architecture that you have designed at the level of detail that answers the questions and a motivation of your design choices.
- GFS-8: A motivated presentation of your design, block diagrams, Arx code, C++ testbench code (only if modified), performance figures (both BER as well as speed and area after synthesis), evaluation of the obtained improvement.

# **Grading**

- GFS-1: 8 points
- GFS-2: 2 points
- GFS-3: 5 points
- GFS-4: 2 points
- GFS-5: 5 points
- GFS-6: 3 points
- GFS-7: 5 points
- GFS-8: 30 points (the number of points will depend on the correctness and sophistication of your solution; as a reference, the "dedicated mixer" solution, which is rather simple, will limit your number of points to 15).

Go (back) to <u>Sabih's Home Page.</u> *Last update on:* Mon Mar 20 23:15:10 CET 2023 by <u>Sabih Gerez</u>.