Comparing HLS with VHDL at the example of an FM Radio Receiver

Michael Wurm

MASTERARBEIT

eingereicht am
Fachhochschul-Masterstudiengang

Embedded Systems Design

in Hagenberg

im Juli 2021

Advisor:

DI (FH) Dr. Florian Eibensteiner

Declaration

I hereby declare and confirm that this thesis is entirely the result of my own original work. Where other sources of information have been used, they have been indicated as such and properly acknowledged. I further declare that this or similar work has not been submitted for credit elsewhere. This printed copy is identical to the submitted electronic version.

Hagenberg, July 15, 2021

Michael Wurm

Contents

Preface

Abstract

This should be a 1-page (maximum) summary of your work in English.

Kurzfassung

An dieser Stelle steht eine Zusammenfassung der Arbeit, Umfang max. 1 Seite. ...

Chapter 1

Introduction

1.1 Motivation

In the master's degree program Embedded Systems Design a wide range of topics relate to digital signal processing, chip design and software development. All of these disciplines find their application in an embedded system. Multiple different methods of implementation are introduced for each topic and their usage is tested in practical sessions. However, in the setting of a university course it is not always possible to look at the currently handled topic as a part of a larger, integrated system. Instead, it is often treated as a standalone part that fulfills its own, specific functionality.

Therefore, one of the main motivation points of this thesis is to combine several disciplines and multiple implementation methods in a common context of a single project.

In a more general perspective, the current electronics market situation is a huge motivational factor as well. Smart devices, smart sensors, Internet of Things (IoT) and many similar terms are well-known and are becoming more and more present in the publics' daily lives. In some degree, all of these devices represent embedded systems, since they all consist of processing units, combined with sensors and interfaces to communicate with their outside world. In the future, the number of devices and the range of applications in this area is going to grow from todays' point of view. This will require a lot of engineering work to advance the state of technology we have today and explore new areas of application.

Since the development of an embedded system combines so many displines, it is of advantage to have knowledge of as many as possible of them. However, not only understanding single parts, but to have an overview and understanding of the entire, integrated system and the interaction between those single parts is important in an embedded system.

Those major reasons lead to the choice of topic for this thesis - understanding all the single parts by themselves, while maintaining the understanding over an integrated embedded system that fulfills a task as a whole.

1.2 Choice of Application

In order to elaborate on the theoretical topics of this thesis, a comprehensive practical project is developed along with it. The project of choice is a digital FM broadcast radio receiver. It combines several disciplines that appear in the master's program Embedded Systems Design, such as digital signal processing (DSP), software development and system design and thus is a perfect candidate for this project.

FM broadcast radio is chosen as the input signal, because it is a common radio frequency signal that is available almost everywhere. Furthermore, the demodulated signal is an audio signal, usually music or speech, that can be listened to. This is subjectively more attractive than a generic data stream. Additionally, it is a comparably simple signal to demodulate, but still requires several techniques of signal processing to decode it.

1.3 Goals

The main goal of this thesis is to elaborate a system design for an embedded system, that combines all the knowledge of different disciplines and topics which are accumulated over the time of the university program, in a single project.

This includes the development of a system model in a tool like Matlab, the practical usage and implementation of DSP in an FPGA, the development of a digital receiver and the deployment on actual hardware.

The receiver implementation on the FPGA is to be done in two variants. The first variant is traditional, manually written hardware description language (HDL) in the programming language VHDL, as it has been done for years. For the second variant, high-level synthesis (HLS) is employed, to generate the low-level HDL from high-level C++ code. These two methods are implemented and compared on the basis of usability and some metrics.

In the view of a system design approach, another focus is put on a sustainable testing and development strategy for the entire system. Above mentioned implementation variants should optimally share as much as possible of testing and verification code, in order to save time and effort in implementation. Also, as many tasks as possible should be automated by scripts, to save time in manual labor for the engineers working on the development. All that enables an efficient development cycle of the product.

By the end of this thesis, the implementation of an FM radio receiver in multiple methods and abstraction levels should be achieved, as well as a good understanding of the overall system level design. Knowledge of each methods' preferred field of application, efficiency and application should be gained.

Chapter 2

Signal Processing Theory

This chapter describes techniques of signal processing, which are required to receive and decode an FM broadcast radio signal. The entire signal processing chain from the antenna, through the analog front-end, to the digital processing parts is explained. However, the main focus is put on the digital signal processing part, since this is a major part of the practical implementation of this thesis.

2.1 Overview

Wireless transmission systems usually transmit their signals over the air. Various frequency bands and different types of modulation are being used for this purpose. The choice of frequency, as well as the choice of the modulation type, have impacts on the transmission characteristics. Depending on these choices, positive effects can be achieved in the amount of data that can be sent over the transmission channel, for example. Another advantage is the ability to adapt the ongoing transmission to the changing characteristics of the channel. This is especially useful for mobile receivers, since the signals' path is inherently changing over time there, which leads to challenging conditions for the receiver. Additionally, varying signal strength, or multipath propagation may occur. Multipath propagation is caused by reflections of the signal, which - for example - can occur on a house facade, or a large vehicle that is passing by. The receiver then gets a signal, that is a summation of the direct path and the reflected path - a signal from multiple paths. Besides that, environmental impacts, such as rain or fog have an effect on the transmission channel and thus the received signal.

Many more factors could be listed, and elaborations over the various modulation types fill entire books. However, with all these different facts, there are things that they all have in common. On the transmitter side, all these signals are transmitted in a band-limited, high frequency signal. The band limitation needs to be given at the transmitting antenna, to minimize interference to other bands in the frequency spectrum. The frequency needs to be high, to radiate the signal by an antenna.

On the receiver side, once the signal reaches the antenna of a receiver, there are further similarities. The antenna signal needs to be amplified and filtered at first. Both steps depend on the antenna characteristics, but usually the received signal is of relatively low

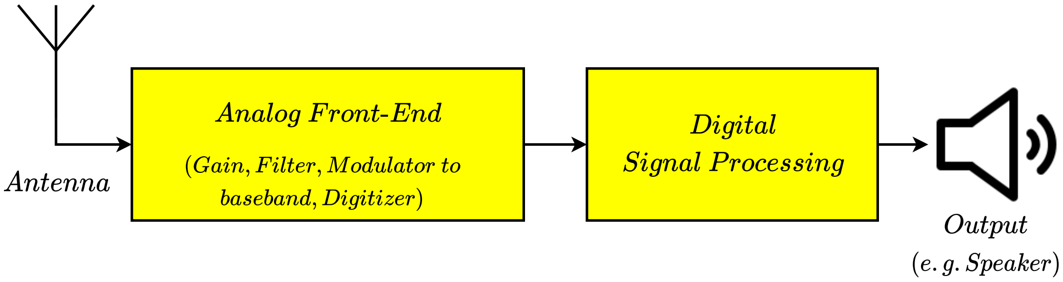

Figure 2.1: High-level block diagram of a signal processing chain.

power, and the antenna receives a wider range of frequencies than required. The filter therefor selects the frequency area that is of interest. Another reason for the filter is to avoid spectral replicas in the subsequent modulation, or down-conversion. Here, the front-end often down-converts to a low intermediate frequency (IF), instead of directly going to baseband. The intermediate frequency is then shifted down to baseband by a modulator later in the digital domain, which brings advantages in the downstream processing chain.

These three steps - amplification, filtering and down-conversion to an IF - are often done in the analog domain, since digital circuits cannot work with such high frequencies. After the analog signal is down-converted to baseband, it needs to be digitized. This is done by an analog-digital-converter (ADC).

After the signal is processed to digital baseband, the received signal can finally be decoded to get the actual data content. This decoding process depends on the respective modulation type that is used.

The block diagram in Fig.2.1 gives a high-level overview of a receiver structure.

2.2 Receiver Types

Multiple different receiver architectures exist and are implemented in receivers. Two common examples are explained in the following sections.

2.2.1 Superheterodyne Receiver

The superheterodyne receiver is often referred to as *superhet*, or *super* in the literature. The main characteristic of a superhet is, that it converts the radio-frequency (RF) antenna signal to an IF in a first stage, instead of going to baseband directly.

The IF is usually set to a fixed frequency, which results in the advantage, that the subsequent IF filter can be optimized, since it does not have to be adapted in frequency. Therefore, a superhet receiver is able to select narrow band signals, even if the bands' surrounding frequencies are being used by other signals. This is one of the superhets' main advantages. A disadvantage is that the signal noise around the image frequency

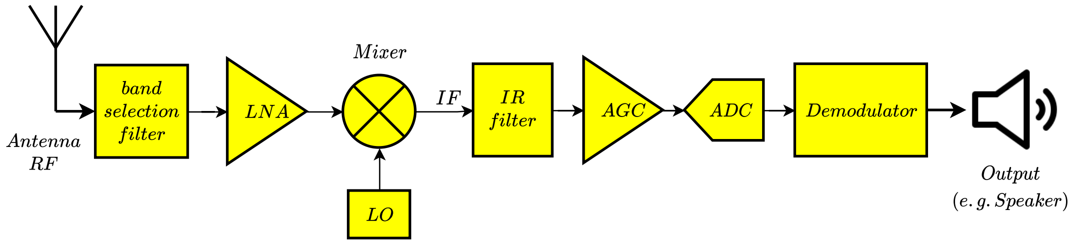

Figure 2.2: Block design of a superhetorodyne receiver [12, Fig.2.1]

band is translated into the IF output band. Thus, the filter requirements are relatively high, since they need to deliver a strong attenuation in order to suppress noise outside the desired band.

Fig.2.2 shows a block diagram of a superhet receiver. The antenna signal first runs through a band selection filter. It selects the desired frequency band and filters out any unwanted signals outside of this band. A low-noise amplifier (LNA) may be used to boost the signal strenght. The next step is the mixer, which is fed by a local oscillator (LO). This shifts, or down-converts, the antenna signal down to the IF. Afterwards, the IF signal is amplified and filtered again to remove the side-products of the mixer, namely the image replicas. Thus, it is also called an image rejection filter. The demodulator stage can now perform any further signal processing on the signal, until the required signal is retrieved [12].

The depicted block diagram may represent an FM broadcast receiver, where an audio output is generated from the antenna input.

2.2.2 Direct-Conversion Receiver

A direct-conversion receiver, as its name suggests, converts an RF signal to baseband directly. It can also be called homodyne, synchrodyne, or zero-IF receiver.

In order to achieve direct down-conversion, the local oscillator of the mixer needs to be set exactly to the center frequency of the desired signal.

The main advantage of direct-conversion receivers is, that their output is an image-free signal. This is due to the properties of the mixer. The strongest disadvantages are the LO leakage, which leads to a DC offset at the mixer output, and the flicker noise, also called *1/f*-noise, which induces strong noise at low frequencies [12]. These effects are not explained into more detail here, since this is not the focus of this thesis.

2.3 Down-Conversion

The process of down-conversion is a technique of amplitude modulation (AM). It is used to convert an RF signal to a signal of lower frequency, while preserving its information content. This is the explanation in the time domain. However, it may be easier look at the process in the frequency domain. In the frequency domain, the band-limited RF signal is shifted down to a lower frequency. This lower frequency can either be zero, i.e.

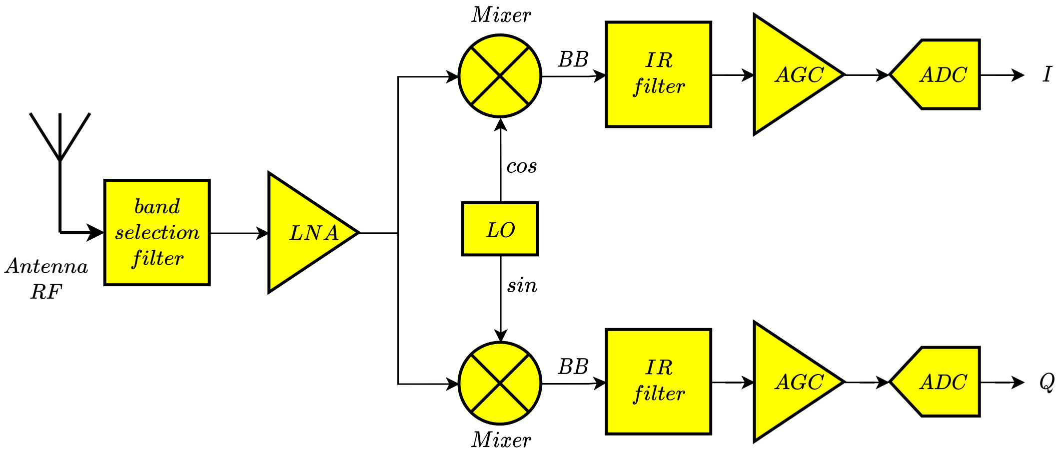

Figure 2.3: Block design of a direct-conversion receiver [12, Fig.2.3].

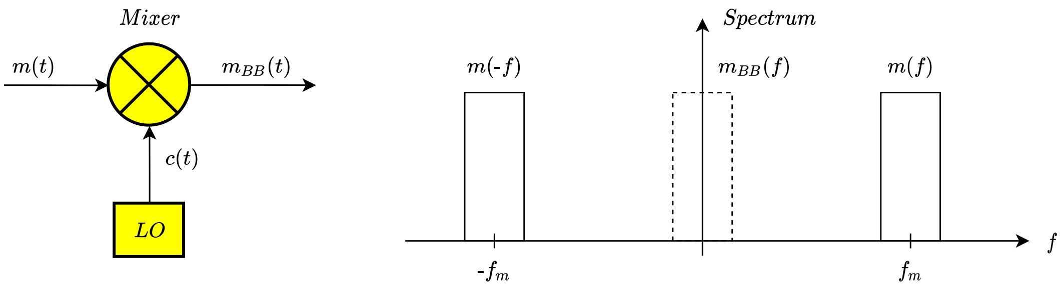

Figure 2.4: Block diagram and signal spectrum of down-conversion.

baseband, or an intermediate frequency, such as in the application of a superheterodyne receiver.

In order to achieve this, the incoming signal needs to be multiplicated with a carrier signal, that is generated locally in the receiver, by a local oscillator (LO). Depending on the frequency f_{LO} of this local carrier, the resulting signal will be shifted to baseband, or an IF. Note, that in the case of down-conversion, the local oscillator frequency f_{LO} is interchangeably denoted as carrier frequency f_c.

A block diagram, as well as the signal spectrum of the process is shown in Fig.2.4. In this example, the frequency of the LO is set to shift the signal from f_m to baseband.

2.3.1 Mathematical Description

The procedure that is used for down-conversion is an amplitude modulation (AM). The equation for a generic AM signal is $S_{AM}(t) = A_m m(t) \cdot A_c c(t)$, where $m(t)$ represents the message signal, and $c(t)$ is the high-frequency carrier. Their respective amplitudes are A_m and A_c. This equation is depicted as a block diagram and signal spectrum in Fig.2.4, where S_{AM} is denoted as the baseband signal m_{BB} of the original signal m.

The signal $m(t)$ is a baseband signal. It may be antenna signal that was filtered by a bandpass filter. The received signal is a real signal, and thus its negative frequency

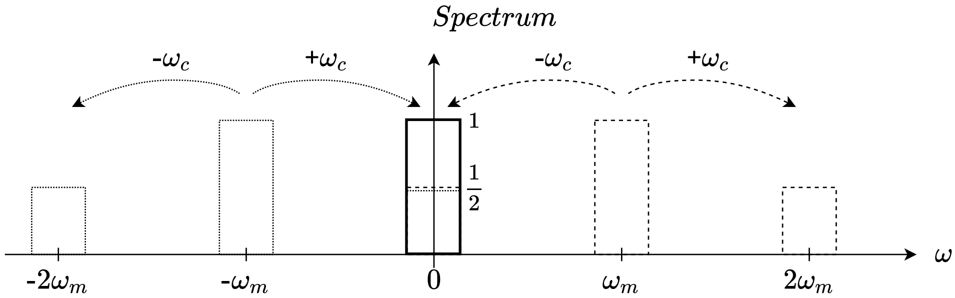

Figure 2.5: Signal spectrum, explaining the process of down-conversion.

spectrum is an exact mirror of the positive frequencies. This fact can be explained in two simple equations.

It is well-known that the real signal $cos(wt) = \frac{1}{2}\left(e^{jw} + e^{-jw}\right)$, which follows from $e^{jw} = cos(wt) + jsin(wt)$. Looking at the frequencies of the first equation, the negative and positive parts can be seen.

The process of down-conversion is described in the following formulas. The denotation of the subscripts is assuming a direct conversion to baseband.

$$m_{BB}(t) = A_m m(t) \cdot A_c c(t) \tag{2.1}$$

Assuming that $m(t) = cos(\omega_m t)$, $c(t) = cos(\omega_c t)$ and $A_m = A_c = 1$,

$$m_{BB}(t) = cos(\omega_m t) \cdot cos(\omega_c t) \tag{2.2}$$

follows. By applying the mathematical theorem

$$cos(\alpha) \cdot cos(\beta) = \frac{1}{2} \cdot \left[cos(\alpha - \beta) + cos(\alpha + \beta)\right] \tag{2.3}$$

the final formula of the conversion can be represented as

$$m_{BB}(t) = \frac{1}{2} \cdot \Big[cos\big((w_m - w_c)t\big) + cos\big((w_m + w_c)t\big)\Big] \tag{2.4}$$

The graphical representation in the frequency spectrum is shown in Fig.2.5. As mentioned above, the assumption is a direct conversion to baseband, so $w_m = w_c$. Looking at the positive original part, it is shown that only half of its energy is translated into the desired baseband. However, half of the energy of the original negative part is also shifted to baseband. The superposition of the two parts results in an amount of energy that is again equal to the original signal.

2.4 Sampling Theorem

Signal processing is often done in the digital domain, using digital signals. A digital signal is obtained by sampling an analog signal in a periodic interval, the so-called sampling frequency f_s.

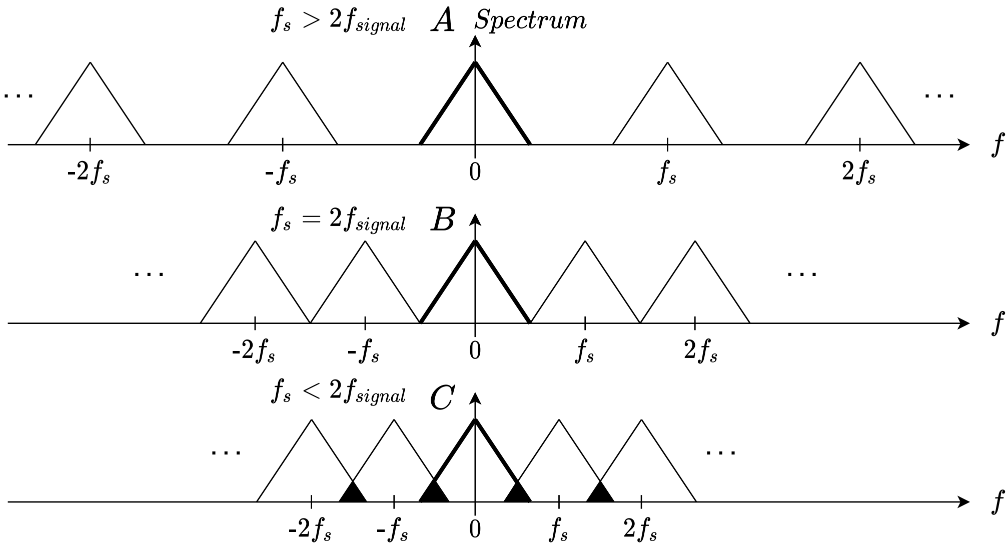

Figure 2.6: Frequency spectrum of the sampled baseband signal with its replicas at integer multiples of f_s [15].
A) $f_s > 2f_{signal}$: f_s is high enough and thus meets the theorem.
B) $f_s = 2f_{signal}$: f_s exactly meets the theorem.
C) $f_s < 2f_{signal}$: f_s is too low and thus creates overlaps of the replicas - aliasing.

The sampling frequency needs to meet the sampling theorem, in order to correctly represent the analog signal. The sampling theorem, by Nyquist and Shannon mainly consists of the following formula.

$$f_s > 2 \cdot f_{signal} \tag{2.5}$$

In words: The sampling frequency must be higher than twice the frequency of the signals' highest frequency content. Any digital signal processing technique must hold the specification of the sampling theorem [3, chpt. 4.2].

In case the theorem is violated, the effect of aliasing occurs on the generated signal. The sampling process introduces spectral replicas at integer multiples of the sampling frequency f_s. These replicas overlap, if f_s does not meet the theorem, which results in signal distortion. Fig.2.6 illustrates the effect of aliasing in the frequency spectrum [15].

2.5 Sample Rate Reduction

Sample rate reduction is also called downsampling. Downsampling is a way to reduce the amount of data that needs to be processed, without losing any of the signals' information. In some applications, the digital signal is oversampled by a large factor. However, if the frequency spectrum of the desired signal is a narrow band, a high oversampling rate is not necessary and the signal can be downsampled without losing any information content. To guarantee this, the Nyquist sampling theorem must still be met after the downsampling process.

A lower sample rate is especially useful for efficient filter design. Factors such as the transition bandwidth or the width of the passband are important there. Short transition bandwidths, or a narrow passband require a high filter order to fulfill the specifications. This translates into a high computational effort. In such cases, it is of advantage to downsample the input, in order to create better conditions for the filter design. Downsampling spreads the spectrum of interest, in terms of the relative bandwidth between passband and sampling frequency. Therefore, the filter can be designed with wider transition bandwidth and passband, which significantly reduces the required filter order.

As an example, assuming the signal of interest has a bandwidth of 20 kHz. The Nyquist theorem thus requires a remaining sampling rate of at least twice the signals' frequency, so 40 kHz. The signal is sampled with an oversampling factor of 10 - meaning a sample rate of 200 kHz. A downsampling factor of 4 is now applied, which results in a sample rate of 50 kHz. The resulting signal is now downsampled and can be processed at this lower data rate, while none of the original data content is lost.

The process consists of two parts: filtering and skipping samples. Let us assume downsampling by a factor of $M = 4$. In the first step, a low-pass filter with a normalized cut-off frequency of *1/M* is applied. This is required in order to avoid aliasing later in the process. The second step simply skips a number of M-1 samples in the sequence.

More detailed information can be found in [3, chpt. 6.9], [9, chpt. 4.1] and [15, chpt. 10.2.2].

2.6 Frequency Modulation in Broadcasting

Frequency modulation (FM) is a widely used standard to transmit data streams. The probably best known usecase therefor is commercial broadcast radio, where an audio stream is transmitted. Devices to receive these streams are available for low prices to the public.

This section describes the main properties of broadcast FM, such as the mathematical description, frequency bands that are used, or the specific frequency parts within a channel spectrum.

2.6.1 Mathematical description

The mathematical description of an FM baseband signal can be described with the formulas presented in this section. FM encodes the information content in its instantanious frequency. This means that the measured frequency at any moment in time represents a specific value of a transmitted message. For a general classification, FM belongs to the group of angle, or phase modulated signals (PM). The simple reason therefor is, that a frequency has a direct relationship to an angle, if the signal is seen on a unit circle.

The instantanious frequency f_i of an FM signal can be described as

$$f_i = f_c + \Delta f \cdot m(t) \tag{2.6}$$

where f_c is the carrier- or center frequency, Δf is the maximum frequency deviation and $m(t)$ is the information or message signal that is to be transmitted. Simply looking at this equation, the frequency deviation, which is sometimes also called the swing, varies in the range between $f_c \pm \Delta f$.

Considering the relationship between frequency and angle, as described above, and after some simple substitutions which will not be described into detail here, the equation for a generic frequency modulated wave is the following

$$s(t) = A_c \; cos\Big(2\pi f_c t + 2\pi k_f \int m(t)dt\Big) \tag{2.7}$$

where A_c is the amplitude of the resulting FM signal and k_f is the frequency sensitivity factor. This sensitivity factor has a direct relationship with the modulation index β.

$$\beta = \frac{\Delta f}{f_m} = \frac{k_f A_m}{f_m} \tag{2.8}$$

where f_m is the highest existing frequency, or the bandwidth of the information signal.

Formula (2.7) describes a frequency modulated signal with a generic message signal $m(t)$. A widely used example application of FM is broadcast radio, where audio streams are transmitted. An audio stream can be described as a cosine wave. Therefore, the information signal that is to be transmitted in broadcast radio can be represented as a cosine wave in the form of

$$m(t) = A_m \; cos(2\pi f_m t). \tag{2.9}$$

By inserting this message signal into the generic FM formula (2.7), the final equation for an FM signal transforms into the following form.

$$y(t) = A_c \; cos\Big(2\pi f_c t + \beta \; sin(2\pi f_m t)\Big) \tag{2.10}$$

Several formulas, as well as their derivations are described according to [11, pg.54-55].

2.6.2 Frequency Band

The frequency band that is used for FM broadcasting is defined worldwide and spans from 87.5 to 108 MHz. However, some countries only partially use the band [2, RR5-39]. The range is located within the so-called Very-High-Frequency (VHF) band, which is open to the public in terms of usage. This means, that any transmitter may use the specified frequency range freely. Austria allowed the legal usage in 2006 [4]. However, the transmission power needs to be limited, so that neighboring receivers are not disturbed in receiving any existing channels. The European Commision specifies the power limit as 50 nW of effective radiated power (ERP) [5]. Another limitation is, that a transmitter must be capable to transmit on multiple center frequencies within the broadcasting band. It must not be fixed to a single center frequency. Through this specification, a transmitter is capable of switching its transmission center frequency, in order to avoid interference with another transmitters signal.

In a practical usecase this means that a transmitter needs to select a channel or frequency range, that is not already occupied by an official transmitter. If a free range is found, the transmitter may start sending an FM signal with the mentioned power limitations [1].

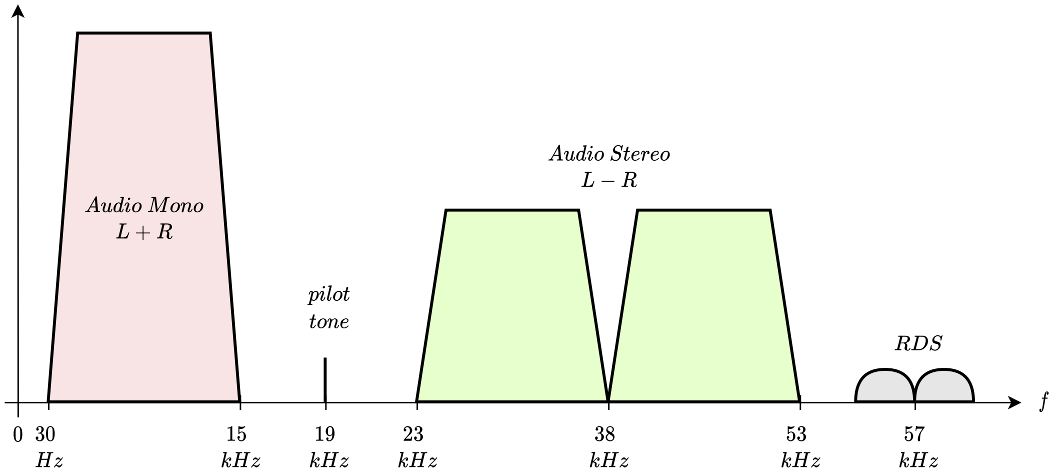

Figure 2.7: Allocation of frequencies in an FM channel [16].

2.6.3 Frequency Spectrum of a Broadcast FM Channel

The entire FM broadcasting band from 87.5 to 108 MHz divides into multiple channels which can be used. In Europe, the European Telecommunications Standards Institute (ETSI) sets the respective standards for the usage of these frequencies. The main specifications for a single FM broadcasting channel are explained in this section.

Each channel may allocate a bandwidth of 200 kHz. Within one of these channels, the maximum deviation from the center frequency shall not exceed 75 kHz. This leaves a guard band of 25 kHz on either side, to minimize interference with adjacent channels. Center frequencies may be allocated on multiples of 100 kHz. However, neighboring channels need to be considered therefore, since this may result in an overlap in the spectrum.

Fig.2.7 shows the frequency spectrum of a single channel, located at its carrier- or center frequency. For a better overview, only the positive spectum part is shown. The negative part is an exact mirror, because the transmitted and received signals from the antenna are real signals - they do not have an imaginary part. The spectrum consists of multiple parts. Therefore, it is also referred to as ‘multiplex signal’ in the literature. Several parts are described in the following paragraphs [1][10][19].

Mono Audio Part

The mono audio stream is located between 30 Hz and 15 kHz. This signal is built by the sum of the left and right audio channels. The upper limit at 15 kHz is chosen to maintain a sufficient spacing to the first subcarrier.

For audio streaming, as in FM broadcasting, this is not a limitation, since it is unlikely to have frequencies higher than 15 kHz in an audio stream. Also, this already reaches the upper limit of the human ears’ bandwith, so higher spectral parts will not be heard anyway.

Pilot Tone

The first subcarrier is allocated at an offset of 19 kHz from the center frequency. This subcarrier is also called the 'pilot tone', since it is a continuous signal. It is independent of any varying message signal content. The pilot tone is used for stereo audio demodulation. To regenerate a stereo audio signal, the left and right audio channels need to be recovered correctly.

Stereo Audio Difference

A signal that is constructed by the difference between the left and right audio channel is modulated on a 38 kHz subcarrier. This subcarrier is an integer multiple of the pilot tone at 19 kHz, for practical reasons. However, the 38 kHz carrier is suppressed and thus not visible in the received spectrum. The modulation technique that is used for this spectral part is called dual-sideband suppressed-carrier (DSB-SC). Even though the carrier is suppressed, it can still be recovered, because it is phase coherent with the 19 kHz pilot tone per definition.

The bandwidth for this difference-signal spans 15 kHz on either side of the subcarrier. It is used to generate a stereo audio signal, in combination with the mono signal. This process is explained into more detail in chapter 2.8.

Additional Services

Considering all these parts in the spectrum, there is still free bandwidth available to use up to the maximum channel bandwitdh of 100 kHz in one sideband. Because of that, additional services were added to the pure audio transmission, to provide additional data services and information.

Services that were implemented are the Data Radio Channel (DARC), which is mostly used in Japan and the USA, the Subsidary Communication Authorization (SCA) and the Radio Data System (RDS) [14]. Out of these, RDS is the most significant service in Europe. It is used to transmit additional information about the channel, such as the radio stations' name, the currently playing songs' title or traffic information.

2.6.4 Pre-Emphasis and De-Emphasis

TODO

2.7 Algorithms for Digital FM Demodulation

An FM signal that is received, for example from an antenna, needs to be demodulated in order to decode its actual data content. In general, an FM demodulator serves the purpose to transform a frequency modulated signal (FM) into an amplitude modulated signal (AM), so that AM signal processing techniques can be applied afterwards.

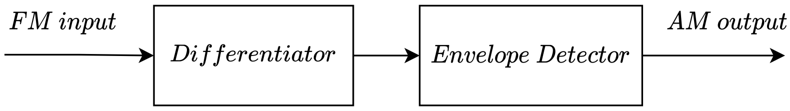

Figure 2.8: Frequency discriminator block diagram [17].

The radio frequency antenna signal usually needs to be amplified, bandpass-filtered, and down-converted to baseband, using subsampling and quadrature-mixing, or similar strategies. This part of the signal processing chain is assumed to be working correctly and is out of the scope of this chapter. The FM signal that is evaluated here is assumed to be a quadrature-mixed signal in baseband, which means that inphase and quadrature (I/Q) signals are available.

In the following sections, two digital FM demodulator variants are described.

2.7.1 Baseband Delay Demodulator

TODO
see /home/mike/Dropbox/Studium/ma_semester_4_SS21/ma/literature/FmDemodulator.pdf

2.7.2 Frequency Discriminator

A frequency discriminator generates an output that is directly proportional to the frequency of the input FM signal. In other words, it converts FM to AM. The discriminator consists of two main parts - a differentiator and an envelope detector. Different strategies can be implemented for both parts. A block diagram for a frequency discriminator with its two main parts is shown in Fig.2.8.

Differentiator

The differentiator already converts the FM signal into an AM signal, but still leaves an FM content. The transformation that happens here can be expressed in a formula, by simply differentiating Eq.(2.7). This operation, with some additional simple algebraic transformations, results in

$$\frac{ds(t)}{dt} = A_c 2\pi \Big[f_c + k_f m(t)\Big] sin\Big(\omega_c t + 2\pi k_f \int m(t)dt - pi\Big) \tag{2.11}$$

This formula shows the AM and an FM contents clearly. The FM part is described by the sine function, with its argument. The more important part is the AM part, which resembles the message signal. It is described by the term within the square brackets.

A time-domain example for this signal is shown in the top left diagram in Fig.2.9.

An important requirement for the differentiator is, that the input signal is of a constant amplitude [13]. The transmitted signal is subject to random additive noise on the transmission channel. The differentiator would deliver wrong results because of this noise, when substracting two consecutive samples, as described in Eq.(2.14). To achieve a constant amplitude, the received complex baseband signal $s(t)$ can be normalized to a value of one.

$$s_{norm} = \frac{s}{|s|} = \frac{A(n)\ e^{j\phi_{FM}(n)}}{|A(n)\ e^{j\phi_{FM}(n)}|} = e^{j\phi_{FM}(n)} \tag{2.12}$$

$$e^{j\phi_{FM}(n)} = |1| \tag{2.13}$$

In a hardware implementation, a differentiation is simply performed by substracting two consecutive samples, like

$$\frac{ds(t)}{dt} = s(t) - s(t - \Delta t) \tag{2.14}$$

where Δt is the inverse of the sample frequency.

Envelope Detector

The differentiated signal still consists of a FM part, as explained above, using Eq.(2.11). In order to remove this high-frequeny part and extract the low-frequency envelope, the signal needs to be fed into an envelope detector. A block diagram is shown in Fig.2.8.

The method implemented in Fig.2.9 is one of the most simple ones and is usually referred to as 'Asynchronous Half-Wave Envelope Detector' [18]. It consists of a thresholding unit to remove negative values and a conventional lowpass filter. In analog implementations, the thresholding unit is a diode. The cutoff frequency of the lowpass needs to be adapted to the envelope signals' maximum frequency. It needs to be able to follow the signal, but also sufficiently smooth out the rectified signal.

Different methods may be implemented for the envelope detector. For example, an 'Asynchronous Full-Wave Envelope Detector' can be used. Therefor, the previous method only needs to swap the thresholding unit with an unit that calculates the absolut value. In that architecture, a full-wave rectification is performed on the signal, which significantly improves the envelope detection accuracy [18]. The envelope is followed more exactly, because of the higher power that is available in the signal, since the entire signal is taken into account, and not only the positive half, as in the previous half-wave method.

2.7.3 Phase-Locked Loop

A Phase-Locked Loop (PLL) can also be used to transform an FM signal to an AM signal. A PLL is a feedback loop that is usually used to generate an output signal that has a fixed phase reference to its input. In the case of FM demodulation, the loop filter in the feedback branch is configured to be able to follow the frequency variations on the input, which is directly related to the encoded information. The output of the PLL directly delivers an AM signal, which corresponds to the transmitted information. [13]

FM demodulation with a PLL is not described into more detail, since this thesis is focussing on the concept of an entire system architecture.

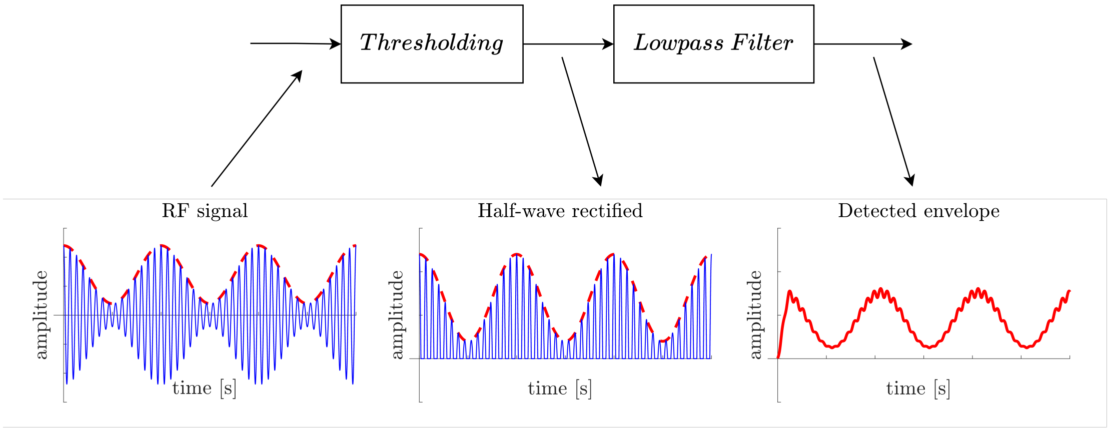

Figure 2.9: Time-domain signals in envelope detection [23].

2.8 Demodulation of Broadcast FM

This section describes the demodulation of the information content of an FM broadcast channel, as it is used in commercial FM radio broadcasting. The assumption here is, that the actual FM demodulation, as described in chapter 2.7 is already done correctly, including the de-emphasis of the signal.

So at this point the FM signal is converted to an AM signal. Thus, traditional AM DSP techniques can be employed to decode the FM broadcast channel.

A frequency spectrum of the multiplex signal of an FM channel is shown in Fig.2.7.

2.8.1 Mono Audio

The first part of the channel frequency spectrum is the mono audio signal. It is the summation signal of the left and right audio channel. Thus, it contains enough information to replay an audio stream in mono quality. A mono receiver can thus simply replay the so-called multiplex signal that is produced by the FM demodulator, without any further filtering or processing. The 19 kHz pilot tone, and any higher frequency components, will not be audible by most people, because it is outside of the range of a human ear. Besides that, any used speaker may also be unable to produce a frequency in this range.

Consequently, in order to demodulate this spectral part, a lowpass filter is required. The filter needs to have a cutoff frequency of 15 kHz, to allow the entire mono audio spectrum to pass. Sufficient attenutation must be available before 19 kHz, in order to suppress the pilot tone.

2.8.2 Stereo Audio

The demodulation of an FM channel to recover a stereo audio signal, requires a more sophisticated approach and multiple steps in signal processing. Two parts need to be combined in order to achieve stereo. The sum and the difference of the left and right audio channel signals - the mono and the stereo difference part, respectively. The following equations need to be applied, to compute a left and a right channel.

$$\begin{aligned}(L+R)+(L-R)&=(2)L\\(L+R)-(L-R)&=(2)R\end{aligned} \tag{2.15}$$

where $(L+R)$ represents the mono part and $(L-R)$ the stereo difference.

The block diagram in Fig.2.10 illustrates the signal processing chain for stereo audio. In the block diagram, $x(t)$ represents the multiplex signal from the FM demodulator. The signals $x_l(t)$ and $x_r(t)$ describe the left and right audio signal, respectively.

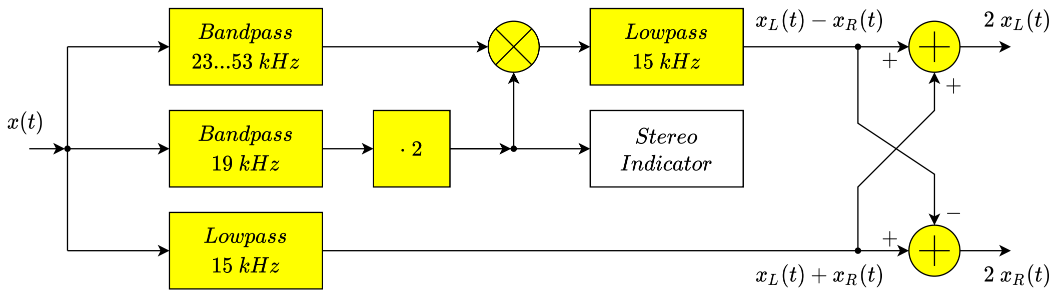

Figure 2.10: Block diagram of FM stereo audio demodulation [23].

The central branch selects the pilot tone at 19 kHz with a bandpass filter. This bandpass needs to have a frequency response that is sharp enough to have a sufficient attenuation at ±4 kHz, since this is where the mono and difference signals' frequency spectra are allocated. Afterwards, the pilot tone frequency is doubled to achieve a 38 kHz subcarrier. This guarantees a phase-coherency between pilot and subcarrier, which is required to correctly shift the difference signal down to baseband.

The frequency doubling can be implemented with multiple methods. The pilot tone can be modulated by simply multiplying it with itself. Another option is to use a PLL and configure it to generate the double frequency at its output. The PLL lock indicator can then be re-used as an indicator, whether a pilot tone is existent.

In the upper branch, a bandpass filter selects the difference signal with ranges from 23 to 53 kHz. It is then modulated to baseband by the previously generated 38 kHz subcarrier. Subsequently, a lowpass limits signal bandwidth to 15 kHz, which removes the generated modulation artifacts.

The lower branch lowpass-filters the summation, or mono signal, as described in paragraph 2.8.1.

The rightmost part in the diagram performs the combination of the mono and the difference signal, according to Eq.(2.15).

Chapter 3

High-Level Synthesis

3.1 Introduction

3.2 State Of The Art

3.3 Functionality (transform high-level code to HDL)

3.4 Language Support

3.4.1 C++

3.4.2 SystemC

3.5 Coding

3.5.1 Compiler Directives (#pragma's)

3.5.2 Data types

3.5.3 Functions

3.5.4 Loops

3.5.5 Conditional statements

3.6 Advantages / Disadvantages

easy and much faster to get complex HDL code, such as image/video processing, through libraries like OpenCV. However, it is less optimized, and a developer still needs to have a strong hardware background, to be able to understand what is being generated in hardware from x lines of code in HLS/Cpp.

Chapter 4

System Architecture and Concept

This chapter introduces the FM radio receiver project that is implemented as a major part of this thesis. It specifically talks about the system level approach that is taken in the implementation of the project. The main concept, as well as the architecture are explained.

4.1 Overview

In the development of a project that includes multiple functional parts, it is always of advantage to start with a system level overview. Major decisions can be made at this stage, before any implementation has begun. This can have a direct impact on the efficiency of the entire development, which concequently has an impact on the final quality of the product. A thorough system level design can also prevent from issues that would otherwise arise while the development is already ongoing. As an example it could turn out that one of the functional parts is not able to fit into the integrated system, in the way it is implemented. This leads to the necessity to re-engineer this entire functional block. As a result, it takes more effort to implement the functionality, which directly correlates with the development cost and the time-to-market, in order to deliver the final product. Summing up, a comprehensive system level architecture and concept can pave the way for a successful implementation of a project from start to finish.

In order to create a system level design, considerations around the high-level, final goal of the product are a good starting point. This includes the definition of its functional range. Once this is done, the target system or target hardware should be considered, because there are various ways to implement a certain functionality. All of these variants will eventually result in the same output, but will, however, heavily differ in their implementation effort, efficiency, or their applicability for the products' aim in general. Thus, things like the available hardware platform, implementation time, degree of optimization, output quality, and similar things are important factors at this stage. Of course, there are many more factors to consider when a product is to be developed. However, only a subset of these is covered here, since this chapter is focused on how this thesis' accompanying project was developed and is thus written in that context.

4.2 The project

The aim of this thesis is develop a system architecture and concept, that allows the implementation of an FM radio receiver in multiple different ways, while providing an elegant way to compare the different solutions. The final goal thereby is always be the same, that is: Listen to the music of a radio station that is transmitted over FM broadcast radio.

A very high-level overview of the entire system is given by the block diagram in Fig.4.1. It shows the front end, which processes the received antenna signal, so that it can be used in the FM radio receiver block. The radio receiver itself is implemented in multiple different variants. The produced audio output is sent to a speaker, since the ultimate goal is to listen to a radio stations' audio signal.

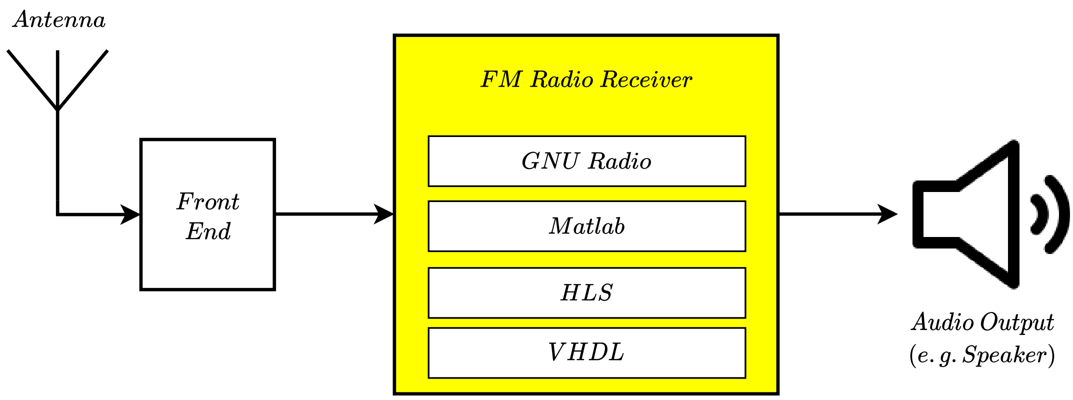

Figure 4.1: High-level overview of the system.

In the context of a system level design approach, several steps are necessary to find a solution that is ready to be implemented. As a very first step, the inner workings of an FM receiver need to be understood. This mainly regards the signal processing theory, to be able to decode the signal that is received by the antenna. Chapter 2 describes this part into detail.

As already mentioned above, multiple ways of technology and implementation can lead to a similar result. Thus, the next step is to define a set of possible ways which will actually be implemented. For this thesis, the decision was made to implement the FM radio receiver by using the following methods.

- **GNU Radio**
  GNU Radio is a free and open-source software development toolkit to implement a software-defined radio. The implementation is done by connecting functional blocks in a block design. This abstracts the inner workings of each functional block and thus provides an implementation approach on a very high level of abstraction. GNU Radio supports interfaces to external RF hardware [20].
- **Matlab**
  Matlab is a software platform that can work with matrices and compute numerical solutions. Matlab also includes various toolboxes, that support developers with

complex calculations and tasks on datasets. This includes a range of toolboxes for signal processing, such as filter designers or FFT functions, as well as convenient ways to visualize data [21].

- **FPGA: C++ High-Level Synthesis**
  In order to use an FPGA, a hardware design needs to be developed for it. This can be done using high-level synthesis. High-level synthesis (HLS) is explained in detail in Chapter 3.
- **FPGA: VHDL**
  Similar to the HLS approach, this is a variant to write a hardware design for an FPGA. The language VHDL is used in this approach.

In order to implement an FM radio receiver, all of the listed methods require deep knowledge about the neccessary details of signal processing. However, the implementation of the various different methods require more or less knowledge and effort on the exact underlying implementation of each signal processing part. This depends on their respective level of abstraction.

4.3 System concept

The overall system concept has to be developed in a way, to fit the respective projects' needs and requirements. There should be a clear structure, in order to help to create a working system successfully. This structure should include a description of the data source, the data processing blocks and stages, commonly used parameters, as well as a strategy to verify the correct functionality of the system. A good system structure tries to share as many parts as possible in the system. This is to minimize the points of failure, as well as to reduce the effort of implementation.

In the apparent projects' case, a single block of the system is implemented in multiple ways, which should, however, all compute an equal output. Thus, it is of advantage to define the data source to be common for all of these variants, to create a unified situation at the input. Additionally, the data source may be chosen to be recorded, to have a known, reproducable set of input data. This significantly eases the verification of the functionality in the final step, since all variants are expected to create the same output in the optimum, fully functioning case. Not only the data source is important to be common, but also the way of verification. In order to achieve that, the verification block needs to be able to compare multiple data outputs with the verification data, which is chosen to serve as the reference.

An example of a system concept and its architecture is developed and shown in Fig.4.2. Here, a common data source is provided by the block called *Front End*. It records data from an antenna and saves it to a file, in a format that the consequent blocks require at their input. Therefore, a reproducable input is ensured. Within the *FM Radio Receiver* block, all the different implementation variants are shown in parallel. They all work on this common input source and generate their respective output. This output data is again written to a file. Hereby it is important to note, that the output of the *Matlab* implementation is chosen to be the 'golden reference model' of correct

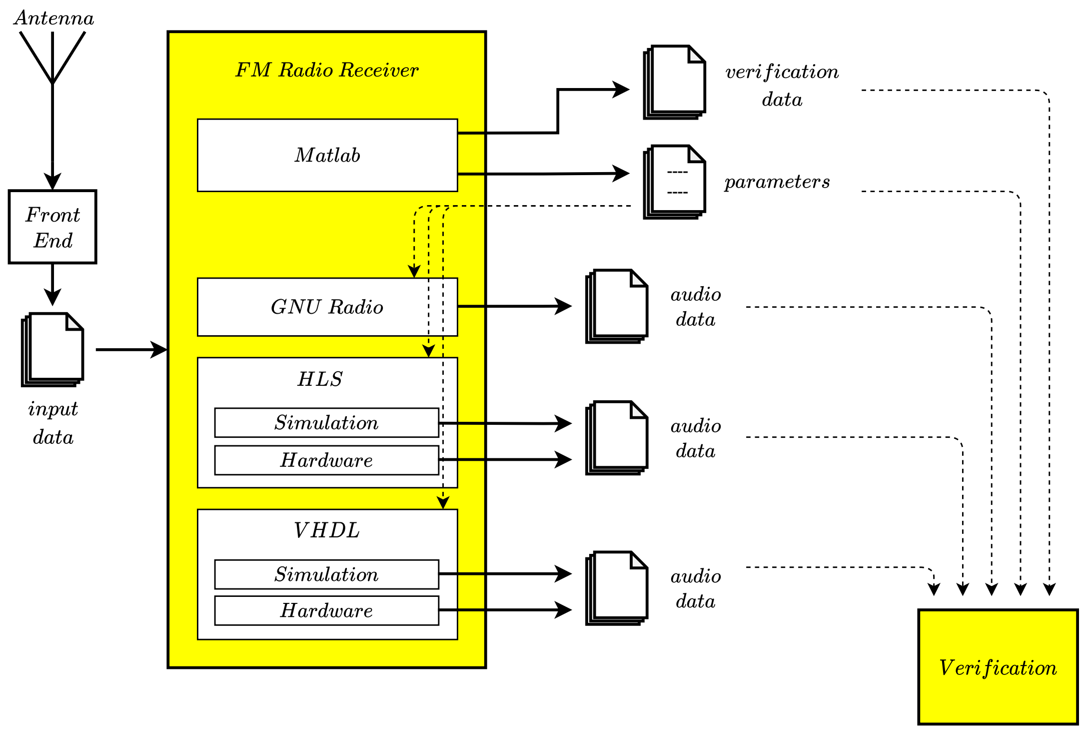

Figure 4.2: Block diagram of the overall system concept.

output. Its output is used as the verification data later in the process. This is because Matlab was used to research decoding algorithms and to develop the DSP chain, which is used in the GNU Radio, HLS and VHDL implementations. In addition to the verification data, the Matlab model also writes common parameters to a file. These parameters are directly used in the other implementations, to ensure a common setup. The final block is the *Verification* entity, which compares the output files of all the variants, with respect to the chosen model parameters.

A major point in this system concept is, that it works independently of the respective implementation methodology. This is especially important, since there is no possible way to run all these different variants in a single verification environment. Instead, each methodology writes an output file, which is a supported functionality that is available anywhere.

Chapter 5

Implementation

This chapter describes the most important details of each implementation variant.

5.1 Digital Signal Processing Chain

The theory behind the demodulation of an FM signal is already explained in Chapter 2. The demodulation of an FM broadcast channel specifically, is described in Section 2.8. In this section however, the DSP chain is described into more detail and with more information, as it is implemented in the various methods.

Fig.5.1 displays the detailed DSP chain as a block diagram.

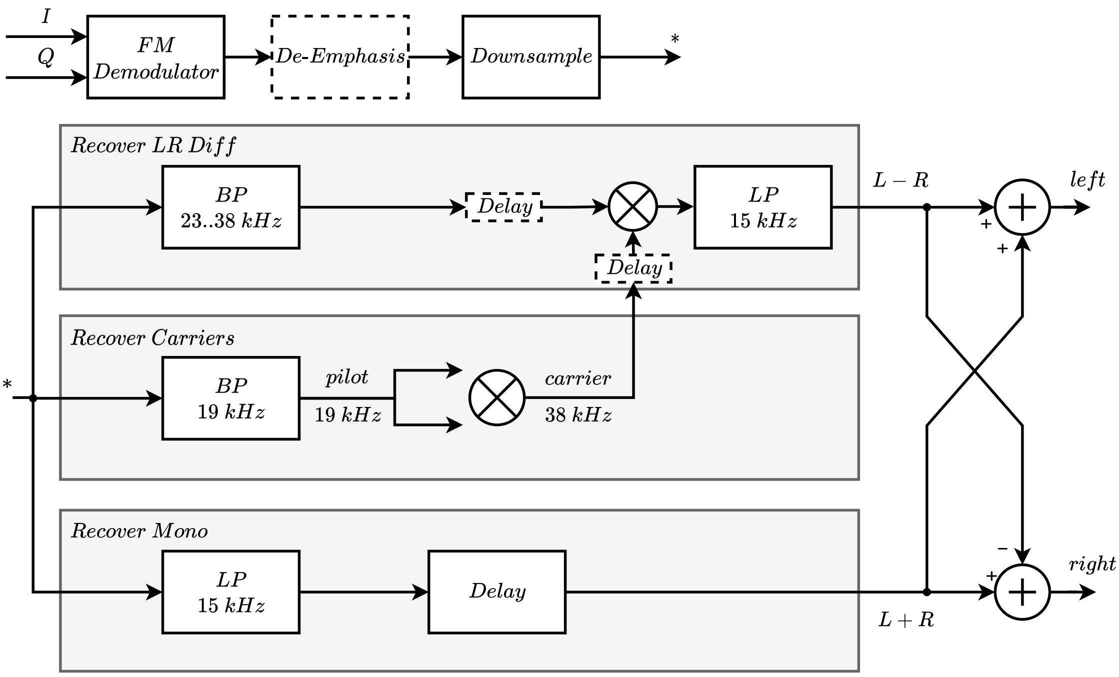

Figure 5.1: Detailed block diagram of the digital signal processing chain.

The following list explains implementation details about the respective DSP blocks.

- **Data Source**
  Matlab is used to generate the data source file, which is used as an input to the system. An audio file is read and modulated as an FM channel signal.
- **FM Demodulator**
  The FM demodulator is implemented as a Baseband Delay Modulator, as described in Section 2.7.1.
- **De-Emphasis Filter**
  The above described Matlab source generation does not include an Emphasis filter. Thus, a de-emphasis filter is not implemented as it is not required.
- **Downsample**
  Downsampling is implemented in a simplified version, without an anti-aliasing filter. It is not required, because of the following explanation. At this point the sampling frequency is set to 960 kHz. The input is band-limited at 75 kHz, which is the maximum frequency in an FM channel, as explained in Section 2.6.3. Thus, the input has an oversampling rate of 12.8, while downsampling is only performed with a factor of 8.
- **Filters**
  Several filters are implemented as FIR filters. The 15 kHz lowpass (LP) filter is instantiated twice, with the exact same coefficients.
- **Filter Group Delay Compensation**
  FIR filters introduce a delay on the output signal, with respect to the input. The group delay in number of samples can be computed with the following formula.

$$group\ delay = \frac{filter\ order}{2} = \frac{(N\ coefficients - 1)}{2} \tag{5.1}$$

  This is due to the property of the FIR filters' linear phase response [7]. Since phase is directly proportional to time, the term *group delay* is often used. This delay needs to be accounted for, depending on where the signal is used subsequently.

  Looking at the block diagram, the group delay needs to be compensated at two places.

  - **Between the branches *Recover LR diff* and *Recover Mono*:**
    The *Mono* branch only contains a single LP filter, while the *LR diff* branch contains two filters, namely a BP and an LP filter. The latter branches' signal is therefore delayed, with respect to the *Mono* branch.
    The issue here is, that both signals are added and substracted at the end of the processing chain, to compute the left and right channel signals, respectively. Thus, they need to be aligned in time at that point. Consequently, a delay block needs to be inserted in the *Mono* branch, to compensate the group delay of the BP.
  - **Between the branches *Recover LR diff* and *Recover Carriers*:**
    Both branches only contain a single BP filter. However, they have properties, like the transition bandwidth and the passband. Thus, their filter order may be different as well, which leads to a non-matching group delay. Since the two signals are multiplicated in the subsequent mixer, and an important factor

is the phase-coherency of the recovered carrier (as explained in Section 2.8), the group delay compensation becomes necessary here.

To overcome the issue at this point in the actual hardware implementation, the filters were manually 'tuned' to have a matching order.

- **Carrier Recovery**
  The 38 kHz carrier can be recovered from the 19 kHz pilot tone. It is important that the recovered carrier is phase-coherent with the received input, as described in Section 2.8. This can be achieved, by simply multiplying the pilot tone with itself, which is the way it is implemented here.

Several of the listed details are implemented, in order to achieve a successful demodulation of the FM signal.

5.2 Matlab Model

Matlab is used to create a model of the receiver, which is used as a reference for the other implementation variants. Here, DSP algorithms are tested and evaluated, to find a method that is suitable to implement in hardware, specifically in an FPGA.

The Matlab script consists of a transmitter, the receiver model, and additional code for analysis.

5.2.1 Transmitter

The transmitter is implemented to provide a reproducable source of data, which can be used to feed the receiver. Another option would be to use external hardware as a software-defined radio and record some antenna data. However, the main advantage of a locally implemented transmitter over a recording is, that the raw data input is known. Thus, the expected output of the receiver model is this raw data, that was previously modulated by the local transmitter. In the case of a recording, the original data is not known, which does not allow to actually verify the receiver.

In the implemented version, an audio file is modulated into the FM channel. The audio file contains data that represents speech with the words *'Left channel... Right channel...'*. In terms of audio channels, the words exactly match the respective channels' output - the first part is modulated on the left audio channel only, while the second part only targets the right audio channel. This is particulary useful for verification, especially to verify the functionality of channel separation. From the experience during development, the channel separation is a good indicator of whether the system is functioning correctly. This left-right indication also simplifies the verification process, because a developer can recognize it with their visual and acoustic senses. It is possible to audibly verify it by listening to the output. It is also possible to just visually inspect the output data, where a clear separation is visible between the channels.

The transmitter is implemented according to the structure that is depicted as a block diagram in Fig.5.2. To facilitate an easy understanding of the block diagram, it

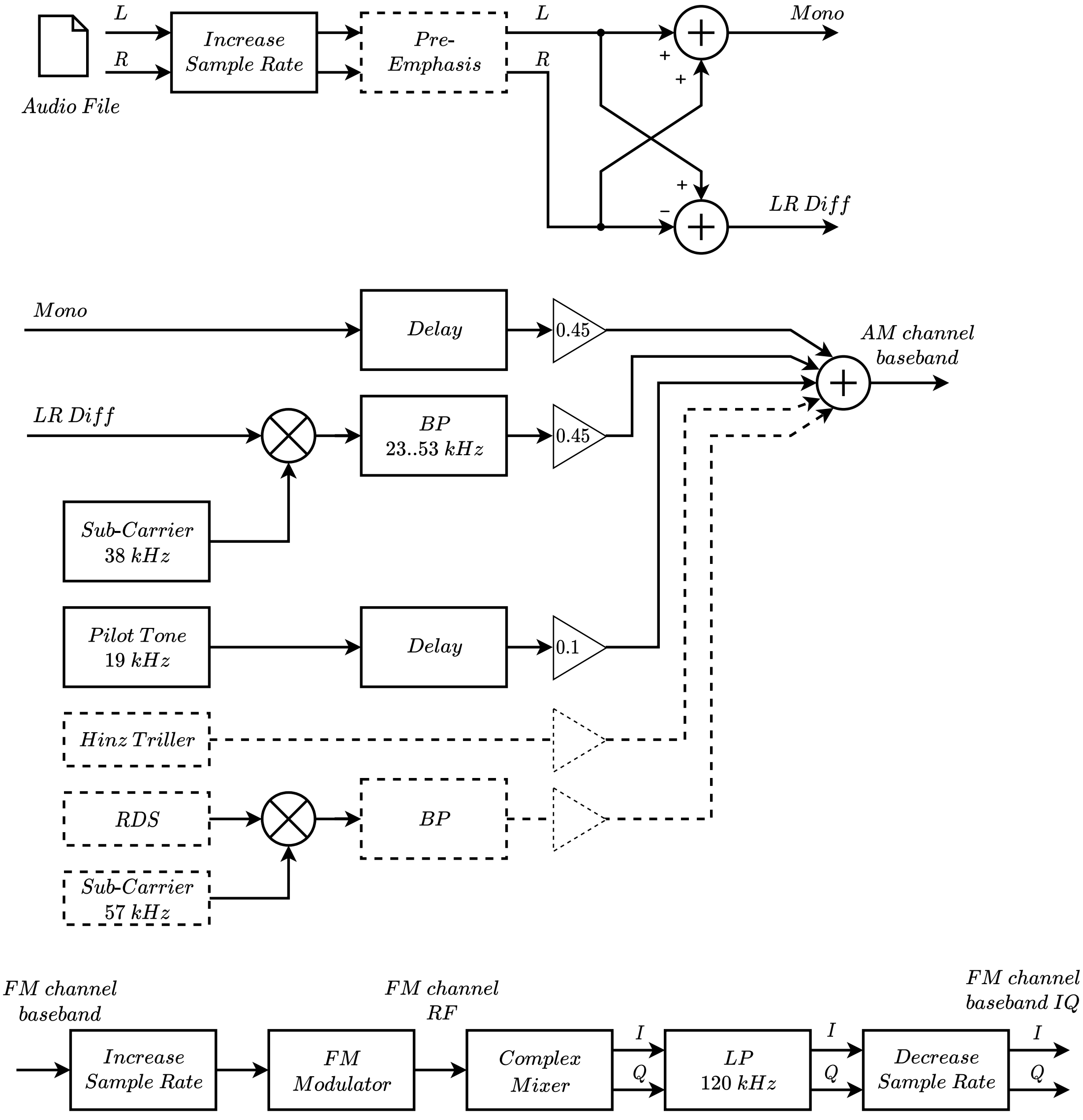

Figure 5.2: Block diagram of the implemented transmitter in Matlab.

is of advantage to keep in mind that it exactly resembles the frequency spectrum that is previously shown in Fig.2.7.

The transmitter reads an audio file, including a left and a right channel in the first step. The audio file contains data in a sample rate of 48 kHz. Next, the sample rate is increased by an oversampling factor of 20, to achieve a sufficient sample rate for the subsequent DSP procedures.

The achieved sample rate of 960 kHz could actually be chosen much lower, since an FM channels' maximum frequency is 75 kHz (see 2.6.3), and according to the Nyquist theorem the sample rate thus only needs to have 150 kHz (see 2.4). However, the higher rate was chosen, mainly because a higher number of signal samples beautify any plots that are created during analysis. Since a thorough optimization of the DSP chain is not the focus of this thesis, this was an acceptable trade-off.

The next step is the pre-emphasis filter, which is implemented, but is disabled in the receiver. This is done to reduce some complexity in the hardware implementation. The following stage generates the Mono and LR Diff signals, by simply calculating the summation and diffence of the left and right channels, respectively.

The next section in the block diagram represents multiple processes in parallel. The LR Diff part is shifted to 38 kHz in the spectrum. This is done by modulating the signal with its sub-carrier of 38 kHz. A subsequent BP filter serves to remove the artifacts of the modulation, such as the image replica. It band-limits the spectrum of the LR Diff part between 23 and 53 kHz. The BP filter introduces a constant group delay of the signal. In order to maintain the alignment of all signals, this side-effect needs to be accounted for and be compensated in the other signal branches. Thus, the Mono signal, as well as the Pilot Tone need to be delayed by the amount of samples defined by the BP filters' group delay. The subsequent amplifier blocks create the defined proportions of the sub-band, such as 90% for the audio parts, and 10% for the Pilot Tone [6].

One part in this block diagram is the so-called Hinz Triller, which was part of the Autofahrer-Rundfunk-Information service (ARI, German for Automotive-Driver's-Broadcasting-Information). It was used as an indicator for traffic announcements. It is now obsolete and is replaced by the more modern Radio Data System (RDS). However, the Hinz Triller is still transmitted in many radio stations and can often be heard before and after the traffic announcements [24]. In the implemented transmitter, the Hinz Triller is implemented, but is disabled, since it only represents a temporarily transmitted signal.

The Radio Data System (RDS) signal is not implemented in the transmitter, since it consists of a comprehensive protocol and multiple steps of modulation, such as differentially-coded BPSK [8] and is therefor too time-consuming to implement. Furthermore, it would not contribute any additional value to this project, since this is not the main focus.

At this point in the DSP chain, the signal resembles the frequency spectrum of Fig.2.7.

In the final chain of DSP blocks, the sample rate is increased again, to be able to frequency-modulate the signal. The resulting frequency modulated signal represents the signal that is received by an antenna - the RF signal of an FM channel. This signal is then processed by a complex baseband mixer, which results in inphase-, and quadrature components in the baseband. An LP filter band-limits the signal and removes artifacts of the mixer. As a final step, the sample rate is decreased again, back to the chosen sampling rate of 960 kHz.

This signal can now be processed in a digital FM receiver.

5.2.2 Pre-Emphasis and De-Emphasis

TODO: show effects in spectrum, show how to design the filters

5.2.3 Fixed Point Arithmetic

The target hardware platform of the FM radio receiver project is an FPGA. These are devices that have a limited number of logic cells, which can be used to implement logical and computational functions. Any kind of computation can be implemented within these logic cells. However, depending on its detailed implementation, the amount of required logic cells may differ by a large factor. In order to overcome this issue, mathematical operations are often implemented in fixed point arithmetic, rather than floating point. This is due to the fact, that fixed point operations can be implemented with much less logic cells than floating point operations. The reason therefor is, that the bitwidth of input and output numbers is previously known in fixed point. Floating point calculations do not have this assumption and thus need to be more flexible, which requires a larger amount of logic cells.

The Matlab model is therefore developed by keeping in mind this limitation. Matlab provides special datatypes for fixed point number representation. However, these were not used in this implementation. Instead, everything is calculated in floating point representation, using the standard *double* datatype. Then, in certain positions in the DSP chain, the numbers are rounded to the resolution of a fixed point datatype. A fixed point format of *2.14*, meaning 2 bit for integer- and 14 bit for fractional number representation is chosen. This is sufficient for the FM receiver and could potentially be lowered to a lower bitwidth. Again, an optimization of the DSP chain is not the focus of this thesis.

Summing up, the Matlab model is not implemented in fixed point arithmetic, but is implemented in a way, that is close enough to the hardware implementation. Therefore, it can serve as a reference for the hardware implementation.

5.3 GNU Radio

5.3.1 Introduction

5.3.2 Transmitter

5.3.3 Receiver

5.4 VHDL (no shortcut)

5.4.1 Testbench

5.4.2 Channel Selection (IF to Channel-BB)

5.4.3 Phase Detector

5.4.4 other Elements

5.5 High-Level Synthesis

5.5.1 Testbench

5.5.2 Channel Selection (IF to Channel-BB)

5.5.3 Phase Detector

5.5.4 other Elements

5.6 Common Testbench

5.6.1 Architecture (same tb for VHDL and HLS-generated HDL)

5.6.2 Framework cocotb, with ghdl compiler

Instantiate both HDL models in the testbench.
Display a direct comparison of outputs in graphs. This is practical, since the cocotb framework runs in python and graphs can be generated using Python's matplotlib.

Chapter 6

Deployment on Hardware

6.1 Hardware Platform

6.1.1 RTL2832u Dongle

6.1.2 ZedBoard

6.2 Block diagram

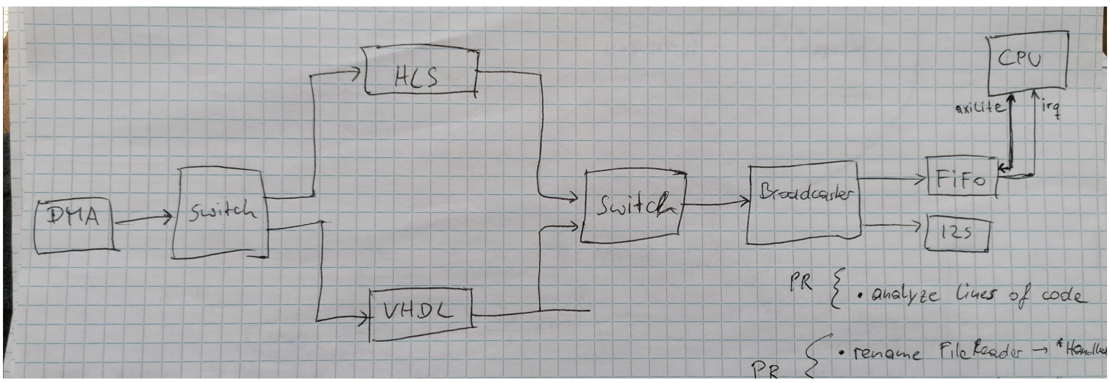

Figure 6.1: Block diagram of the Vivado implementation.

Chapter 7

Comparison and Results

7.1 Demodulated Signal

7.2 Functionality

7.3 Code Development

7.4 Simulation

7.5 Useability

7.6 Applicability

7.7 Hardware Utilization

7.8 Latency

7.9 Lines of Code

Appendix A

Technical Details

Appendix B

Supplementary Materials

List of supplementary data submitted to the degree-granting institution for archival storage (in ZIP format).

B.1 PDF Files

Path: /

- thesis.pdf . . . . Master/Bachelor thesis (complete document)

B.2 Media Files

Path: /media

- *.ai, *.pdf . . . . Adobe Illustrator files
- *.jpg, *.png . . . . raster images
- *.mp3 . . . . audio files
- *.mp4 . . . . video files

B.3 Online Sources (PDF Captures)

Path: /online-sources

- Reliquienschrein-Wikipedia.pdf [22]

List of Abbreviations

ADC	Analog Digital Converter
AM	Amplitude Modulation
AXI	Advanced Extensible Interface
BP	Bandpass (Filter)
CI	Continuous Integration
DC	Direct Current
DSB-SC	Dual-Sideband Suppressed-Carrier
ERP	Effective Radiated Power
FM	Frequency Modulation
FPGA	Field-programmable Gate Array
HLS	High-Level Synthesis
IF	Intermediate Frequency
IoT	Internet of Things
IP	Intellectual Property
IQ, I/Q	Inphase and Quadrature components
LED	Light Emitting Diode
LP	Lowpass (Filter)
LSB	Least Significant Bit
LNA	Low Noise Amplifier
LO	Local Oscillator
PM	Phase Modulation
RTL	Register Translation Level
RDS	Radio Data System
SDR	Software Defined Radio
SoC	System On Chip
VHDL	Very High Speed Integrated Circuit Hardware Description Language
VHF	Very High Frequency

References

Literature

[1] European Broadcasting Union (EBU). "Low-power FM modulators in Europe". EBU-R120-2007 (2007), p. 5. URL: https://tech.ebu.ch/docs/r/r120.pdf (cit. on pp. 10, 11).

[2] International Telecommunication Union (ITU). *Radio Regulations Articles, Edition of 2020*. eng. Vol. I-EA5. 2020. ISBN: 978-92-61-30301-3 (cit. on p. 10).

[3] Silvia Maria Alessio. *Digital Signal Processing and Spectral Analysis for Scientists : Concepts and Applications*. eng. 1st ed. 2016.. Signals and Communication Technology. 2016. ISBN: 3-319-25468-5 (cit. on pp. 8, 9).

[4] Innovation und Technologie Bundesministerium für Verkehr. "Frequenznutzungsverordnung 2013 - FNV 2013". BGBl. II Nr. 63/2014, Anlage 2 (2014), p. 99. URL: https://www.ris.bka.gv.at/Dokumente/Bundesnormen/NOR40219692/II_390_2016_Anlage_2.pdf (cit. on p. 10).

[5] European Commission. "Commission Decision on harmonisation of the radio spectrum for use by short-range devices". 2006/771/EC (), p. 11. URL: https://eur-lex.europa.eu/eli/dec/2006/771(2)/2019-08-13 (cit. on p. 10).

[6] Leonard Feldman. *FM Multiplexing for Stereo*. Vol. FMS-1. 62-13498 (cit. on p. 26).

[7] Orhan Gazi. *Understanding Digital Signal Processing*. eng. 1st ed. 2018.. Springer Topics in Signal Processing. 2018. ISBN: 981-10-4962-9 (cit. on p. 23).

[8] Liang Ge, EK Tan, and Joe Kelly. "Introduction to FM-Stereo-RDS Modulation" (). URL: https://www3.advantest.com/documents/11348/7898f05e-0a52-4e68-9221-3b8b75595436 (visited on 07/14/2021) (cit. on p. 26).

[9] E. S Gopi. *Multi-Disciplinary Digital Signal Processing : A Functional Approach Using Matlab*. eng. 2018. ISBN: 9783319574301 (cit. on p. 9).

[10] European Telecommunications Standards Institute. "Transmitting equipment for the Frequency Modulated (FM) sound broadcasting service; Harmonised Standard covering the essential requirements of article 3.2 of Directive 2014/53/EU". v2.1.1.ETSI EN 302 018 (Apr. 2017) (cit. on p. 11).

[11] Mohammad A Matin. *Communication Systems for Electrical Engineers*. eng. SpringerBriefs in Electrical and Computer Engineering. 2018. ISBN: 978-3-319-70129-5 (cit. on p. 10).

[12] Fábio Passos. *Automated Hierarchical Synthesis of Radio-Frequency Integrated Circuits and Systems : A Systematic and Multilevel Approach*. eng. 1st ed. 2020. 2020. ISBN: 978-3-030-47247-4 (cit. on pp. 5, 6).

[13] Franz Schnyder and Christoph Haller. “Implementation of FM Demodulator Algorithms on a High Performance Digital Signal Processor”. diploma thesis. HSR Hochschule für Technik Rapperswil, Elektrotechnik. URL: https://www.veron.nl/wp-content/uploads/2014/01/FmDemodulator.pdf (visited on 07/06/2021) (cit. on pp. 13, 14).

[14] Rohde & Schwarz. “Messungen an FM-Sendern für Abnahme, Inbetriebnahme oder Wartung”. Sep-13 - 7BM105_0D (). URL: https://cdn.rohde-schwarz.com/pws/dl_downloads/dl_application/application_notes/7bm105/7BM105_0D.pdf (visited on 07/05/2021) (cit. on p. 12).

[15] K.S Thyagarajan. *Introduction to Digital Signal Processing Using MATLAB with Application to Digital Communications*. eng. 2019, pp. 39–44. ISBN: 9783319760292 (cit. on pp. 8, 9).

Media

[16] *Basisbandsignal eines UKW-Rundfunkprogrammes*. URL: https://de.wikipedia.org/wiki/UKW-Rundfunk#Technische_Details (visited on 07/05/2021) (cit. on p. 11).

[17] *Frequency Discriminator Block Diagram*. URL: https://www.tutorialspoint.com/analog_communication/analog_communication_fm_demodulators.htm (visited on 12/16/2020) (cit. on p. 13).

Online sources

[18] Rick Lyons. *Digital Envelope Detection: The Good, the Bad, and the Ugly*. Apr. 3, 2016. URL: https://www.dsprelated.com/showarticle/938.php (visited on 07/06/2021) (cit. on p. 14).

[19] Electronics Notes. *Broadcast VHF FM Tutorial & Basics*. URL: https://www.electronics-notes.com/articles/audio-video/broadcast-audio/vhf-fm-frequency-modulation-basics.php (visited on 07/05/2021) (cit. on p. 11).

[20] GNU Radio project. URL: https://www.gnuradio.org/about/ (visited on 07/09/2021) (cit. on p. 19).

[21] GNU Radio project. URL: https://www.gnuradio.org/about/ (visited on 07/09/2021) (cit. on p. 20).

[22] *Reliquienschrein*. Sept. 2018. URL: https://de.wikipedia.org/wiki/Reliquienschrein (visited on 02/28/2019).

[23] Carsten Roppel. *Analoge Modulationsverfahren und Rundfunktechnik; Begleitmaterial zum Buch 'Grundlagen der digitalen Kommunikationstechnik, Übertragungstechnik - Signalverarbeitung - Netze'*. Aug. 24, 2010. URL: https://www.hs-schmalkalden.de/fileadmin/portal/Dokumente/Fakult%C3%A4t_ET/Personal/Roppel/Buch/Analoge_Modulationsverfahren.pdf (visited on 07/06/2021) (cit. on pp. 15, 16).

[24] Wikipedia. URL: https://en.wikipedia.org/wiki/Autofahrer-Rundfunk-Informationssystem,%20https://de.wikipedia.org/wiki/Autofahrer-Rundfunk-Information (visited on 07/09/2021) (cit. on p. 26).

Check Final Print Size

— Check final print size! —

width = 100mm
height = 50mm

— Remove this page after printing! —