Signal Processing to Demodulate FM Radio

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***Abstract*—This paper describes the necessary parts of digital signal processing, in order to be able to listen to a radio station that is broadcast in the FM frequency band. Fundamental knowledge about frequency modulation and its application in audio broadcast is presented. A single FM radio channel in the FM radio band consists of multiple parts in the frequency spectrum, which is an important factor to be able to demodulate an audio stream correctly. This demodulation itself can be done in different algorithms, of which a selection is introduced. The entire demodulation chain is implemented in a Matlab simulation script, to demonstrate the functionality of an FM radio receiver.**

***Index Terms*—Audio signal processing, FM radio frequency, FM demodulation, Digital FM receiver, Matlab, Simulation**

I. Introduction

Frequency modulation (FM) is a widely used standard to transmit data streams. The probably best known usecase therefore is commercial broadcast radio, where an audio stream is transmitted. Devices to receive these streams are available starting from affordable low prices to the public. Since this is such a common technology that so many people are using on a daily basis - and which is included in virtually every automobile - a more detailed look into this signal transmission system is very interesting.

This paper tries to give an overview about the receivers' side of the FM radio technology. A Masters' thesis will be elaborated based on this topic.

II. Frequency Modulation

Frequency modulation (FM) is a modulation method, in which the information content is encoded in the frequency of a signal. To be exact, the information is encoded in the frequencies' deviation from the carrier frequency. The technology is widely used in FM broadcast radio. One of its main advantages over amplitude modulation (AM) is the higher signal-to-noise ration (SNR). This is achieved, because of the constant amplitude in FM, which makes it less sensitive to additive noise in the transmission channel. Additionally, this fact makes it easier for a receiver to synchronize on the received signal [7].

A. Mathematical Description

The signal of an FM baseband signal can be described with the formulas presented in this section. FM encodes the information content in its instantanious frequency, and since frequency has a direct relationship to an angle, FM belongs to the group of angle modulated signals.

The instantanious frequency f_i can be described as

$$f_i = f_c + \Delta f \cdot m(t) \tag{1}$$

where f_c is the carrier 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 also called the swing, varies in the range between $f_c \pm \Delta f$.

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

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

where A_c is the amplitude of the resulting FM signal and k_f is the frequency sensitivity factor.

An example application of FM is transmitting audio streams. An audio stream can be described as a cosine wave. If the information signal that is to be transmitted is a cosine wave in the form of

$$m(t) = A_m \ cos(2\pi f_m t) \tag{3}$$

the final equation for an FM signal transforms into the following formula.

$$y(t) = A_c \ cos\Big(2\pi f_c t + \beta \ sin(2\pi f_m t)\Big) \tag{4}$$

The variable β is called the modulation index,

$$\beta = \frac{\Delta f}{f_m} = \frac{k_f A_m}{f_m} \tag{5}$$

where f_m is the highest existing frequency, or the bandwidth of the information signal. Several formulas, as well as their derivations are described according to [7] [8].

B. 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 [1]. The range is located within the so-called Very-High-Frequency (VHF) band. In terms of usage the band is open to the public. This means, that a 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) [3]. 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 sender. If a free range is found, the transmitter may start sending an FM signal with the mentioned power limitations [2].

C. Channel Frequency Spectrum

The entire FM band frequency range 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 a channel, 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.

Fig.1 shows the upper half-bands' frequency spectrum of a single channel. A sum of the left and right audio channel is located between 30 Hz and 15 kHz. This summation of left and right audio channel resembles the mono audio stream. The lower limit at 30 Hz is to prevent the transmission of a direct current (DC) part, which would require a large amount of power in transmission. This effect is one of the downsides of amplitude modulation (AM), where the carrier is located at the channels' center frequeny and requires a high transmission power. The upper limit at 15 kHz is chosen, to maintain a sufficient spacing to the first subcarrier. This subcarrier is allocated at an offset of 19 kHz from the carrier. It is also called the 'pilot tone', since it is a continuous signal. The pilot tone is used to be able to demodulate the left and right channel of an audio signal correctly, to generate stereo audio. A signal that is constructed by the difference between left and right audio channel is modulated on a 38 kHz subcarrier. This subcarrier is an integer multiple of 19 kHz for practical reasons. However, the 38 kHz carrier is suppressed and thus not visible in the received spectrum. It can still be recovered, since 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 III.

Considering all these parts in the spectrum, there is still bandwidth free 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) [5]. 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, currently playing songs' title or traffic information. The information about the frequency spectrum was found in [6].

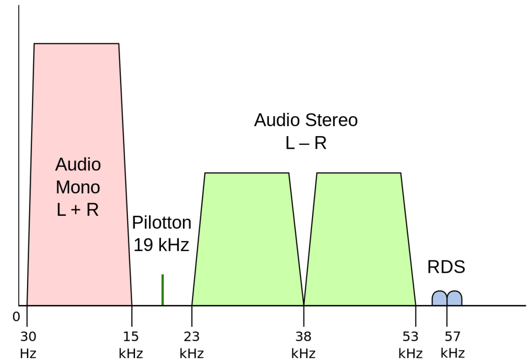

Fig. 1. Allocation of frequencies in an FM channel [9]

III. Demodulation of a Generic FM Signal

An FM signal that is received by an antenna needs to be demodulated, in order to decode its actual data content. 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 paper. 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. In general, an FM demodulator serves the purpose to transform FM to AM, so that AM signal processing techniques can be applied afterwards.

A. Frequency Discriminator

A frequency discriminator can be defined as a black box component, that generates an output that is directly proportional to the frequency of the input FM signal. Different strategies can be implemented in this black box component, the discriminator. Here, a differentiator with a subsequent envelope detector is used, which is shown in Fig.2.

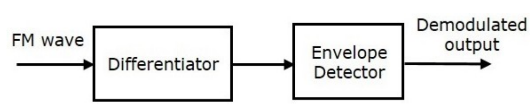

Fig. 2. Frequency discriminator block diagram [10]

This type of demodulator 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. This operation, with simple algebraic transformations, results in

$$\frac{ds(t)}{dt} = A_c 2\pi \Big[f_c + k_f m(t)\Big] sin\Big(\omega_c t + 2\pi k_f \int m(t)dt - pi\Big) \tag{6}$$

A differentiation in hardware is simply performed by substracting two consecutive samples, like

$$\frac{ds(t)}{dt} = s(t) - s(t - \Delta t) \tag{7}$$

where Δt is the inverse of the sample frequency.

It can clearly be seen, that the differentiated signal consist of an AM and an FM content. The FM part is described by the sine function. More importantly the AM part, which resembles the envelope of the signal, is described by the term within the square brackets. The resulting signal in time-domain is shown in the top left diagram in Fig.3. This signal needs to be fed into an envelope detector, as seen in Fig.2. It extracts the envelope by removing the high frequency portion. The method implemented here is one of the most simple ones and is usually referred to as 'Asynchronous Half-Wave Envelope Detector' [11]. 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 lowpass' cutoff frequency needs to be adapted to the envelope signals' maximum frequency, since 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. Therefore, the previous method only needs to swap the thresholding unit with an unit that calculates an absolut value. In that architecture, a full-wave rectification is performed on the signal, which significantly improves the envelope detection accuracy [11]. 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 method.

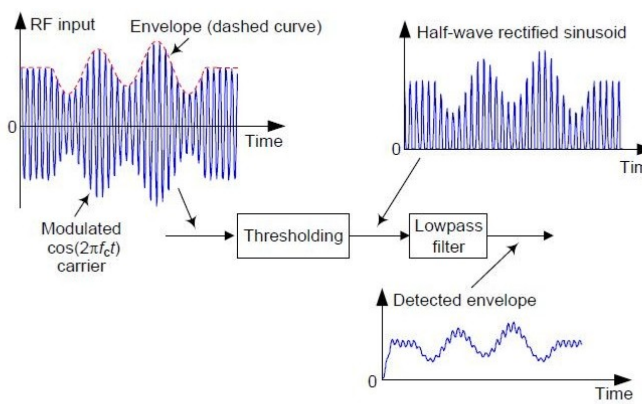

Fig. 3. Time-domain signals in envelope detection [12]

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.7. To achieve a constant amplitude, the complex signal $s(t)$ can be normalized to a value of one.

$$s_{norm} = \frac{s}{|s|} = \frac{A(n)\ e^{j\phi_{FM}(n)}}{|A(n)\ e^{j\phi_{FM}(n)}|} = e^{j\phi_{FM}(n)} = |1| \tag{8}$$

B. Phase-Locked Loop

A Phase-Locked Loop (PLL) can also be used to transform an FM signal to AM. 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 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]

Due to the limited space in this paper, FM demodulation with a PLL is not described into more detail.

IV. Demodulation of Broadcast FM

This chapter is describing 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 III is already done correctly. A frequency spectrum of an FM channel is shown in Fig.1 above.

A. Mono

The first part of the channel frequency spectrum is the summation signal of the left and right audio channel. A mono receiver can thus simply replay the so-called multiplex signal, which is generated by the FM demodulation described in chapter III. The 19 kHz pilot tone 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. This explanation applies to several higher frequency components in the spectrum as well. In case the multiplex signal should be stored it in a file with 44.1 kHz samplerate, a lowpass with a cutoff frequency of 15 kHz and sufficient attenutation before 19 kHz needs to be inserted before the decimation and file output.

B. Stereo

Demodulating an FM channel to recover a stereo audio signal requires a more sophisticated approach. In the channel frequency spectrum, there are two main parts of the audio signal - the sum and difference of the left and right audio channel signals. To combine these to a stereo signal, the following equations can be applied.

$$\begin{aligned} L &= (L + R) + (L - R) = (2)L \\ R &= (L + R) - (L - R) = (2)R \end{aligned} \tag{9}$$

The block diagram in Fig.4 illustrates the signal processing chain for stereo audio. In the block diagram, $x(t)$ represents the multiplex signal from the FM demodulator. Signals $x_l(t)$ and $x_r(t)$ describe the left and right audio signal, respectively.

The central branch selects the pilot tone at 19 kHz with a bandpass filter. This bandpass needs to have a frequency

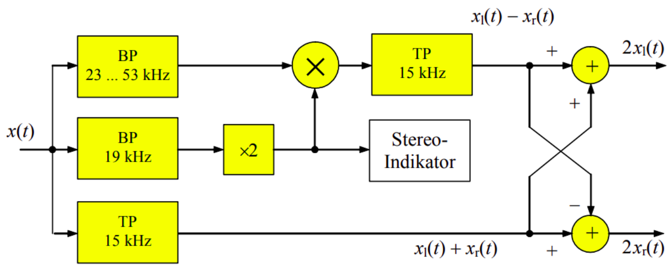

Fig. 4. Block diagram of FM stereo audio demodulation [12]

response that is sharp enough to have a sufficient attenuation at ± 4 kHz, since this is where the summation and difference signals frequencies are allocated. Afterwards, the pilot tone frequency is doubled via a PLL to achieve a 38 kHz to later demodulate the difference signal phase-coherently. The PLL lock indicator can 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 the signal to the audio signal bandwidth of 15 kHz. The lower branch lowpass-filters the summation signal. The rightmost part in the diagram performs the combination of summation and difference signals, according to Eq.9.

V. Matlab Simulation

Matlab is used to simulate an FM receiver. The objective is to demodulate a previously recorded data stream, so that one can listen to a mono audio signal. Several steps that are required therefore are described briefly in the following sections.

A. Data Acquisition

A data stream is recorded, using a low priced RTL-SDR dongle. The device can be plugged into USB on a host computer. It features an RTL2832U chipset. In combination with a host software, this basically assembles a software defined radio (SDR). The recorded data stream is stored as a file with interleaved IQ baseband data.

B. Demodulation Steps

The SDR is used to record a data stream with a sampling frequency of 1 MHz, at a center frequency of 98.0 MHz. A dataset of 10 seconds is recorded. The aim is to listen to the Austrian radio station OE3, which is broadcasted at a center frequency of 98.1 MHz in the region of Saalfelden.

1) Recorded spectrum: The file was recorded at a center frequency of 98 MHz, while the desired station is located at 98.1 MHz. This means, that the recorded spectrum is allocated at an intermediate frequency (IF) of 0.1 MHz.

2) Modulate to FM channel center frequency: Because the signal needs to be in baseband to be able to demodulate it, the spectrum needs to be shifted by 0.1 MHz. This is achieved by modulating with a complex signal with that exact frequency.

3) Decimation: The signal still has a sample rate of 1 MHz at this point. Keeping in mind the Nyquist theorem, and a single channels' bandwith of only up to 100 kHz, the sample rate therefore only needs to be at least 200 kHz. However, the final hardware implementation is targeting an FPGA, where processing of samples at 1 MHz is easily achievable. Therefore, a decimation is not done at this point.

4) Demodulation: Demodulation is done by normalizing the signal amplitude to one. Then, an FIR filter is used as a differentiator. Envelope detection is omitted for now, for simplicity and because of the mentioned 'natural' filter effects in section IV-A.

5) Verification: The correct functionality is only verified subjectively, by listening to the demodulated audio signal via the PCs' soundcard. The OE3 stations' signal can be heard clearly.

VI. Conclusion

FM is a well-known technology, that is established in peoples' daily lifes. Looking at it from a signal processing perspective is an interesting task, since it requires many principles of digital signal processing, but is still comparably simple to demodulate and implement.

References

[1] International Telecommunication Union (ITU), *Radio Regulation Articles, Edition of 2020*, Vol. I-EA5

[2] European Broadcasting Union (EBU), *Low-power FM modulators in Europe*, EBU-R120-2007, https://tech.ebu.ch/docs/r/r120.pdf, pg.5

[3] European Commision, *Commission Decision on harmonisation of the radio spectrum for use by short-range devices*, 13.08.2019, 2006/771/EC, https://eur-lex.europa.eu/eli/dec/2006/771(2)/2019-08-13, pg.11

[4] Bundesministerium für Verkehr, Innovation und Technologie, *Frequenznutzungsverordnung 2013 - FNV 2013*, BGBl. II Nr. 63/2014, Anlage 2, https://www.ris.bka.gv.at/Dokumente/Bundesnormen/NOR40219692/II_390_2016_Anlage_2.pdf, pg.99

[5] Rohde&Schwarz, *Messungen an FM-Sendern für Abnahme, Inbetriebnahme oder Wartung*, https://cdn.rohde-schwarz.com/pws/dl_downloads/dl_application/application_notes/7bm105/7BM105_0D.pdf, opened Dec. 17th, 2020

[6] Electronics Notes, *Broadcast VHF FM Tutorial & Basics*, https://www.electronics-notes.com/articles/audio-video/broadcast-audio/vhf-fm-frequency-modulation-basics.php, opened Dec. 10th, 2020

[7] https://fas.org/man/dod-101/navy/docs/es310/FM.htm, opened on Dec. 12th, 2020

[8] TutorialsPoint, *Analog Communication - Angle Modulation*, https://www.tutorialspoint.com/analog_communication/analog_communication_angle_modulation.htm, opened on Dec. 12th, 2020

[9] Figure: Baseband Signal of an FM channel, https://de.wikipedia.org/wiki/UKW-Rundfunk#Technische_Details, opened on Dec. 11th, 2020

[10] Figure: Frequency Discriminator Block Diagram, https://www.tutorialspoint.com/analog_communication/analog_communication_fm_demodulators.htm, opened on Dec. 16th, 2020

[11] Rick Lyons, *Digital Envelope Detection: The Good, the Bad, and the Ugly*, April 3, 2016, https://www.dsprelated.com/showarticle/938.php, opened on Dec. 16th, 2020

[12] C.Roppel, *Analoge Modulationsverfahren und Rundfunktechnik* Begleitmaterial zum Buch 'Grundlagen der digitalen Kommunikationstechnik, Übertragungstechnik - Signalverarbeitung - Netze', https://www.hs-schmalkalden.de/fileadmin/portal/Dokumente/Fakult%C3%A4t_ET/Personal/Roppel/Buch/Analoge_Modulationsverfahren.pdf, opened on Dec. 16th, 2020

[13] F.Schnyder, C.Haller, *Implementation of FM Demodulator Algorithms on a High Performance Digital Signal Processor*, 2002, https://www.veron.nl/wp-content/uploads/2014/01/FmDemodulator.pdf