Radio FM Signal Demodulation Using the RTL-SDR USB dongle

Electrical Engineering Department
DSP Laboratory (250704)
Instructor: Mohammadreza Bagheri Jazi

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1 Introduction to FM Modulation and Demodulation

Frequency Modulation (FM) is a technique where the frequency of a carrier signal is varied in accordance with the amplitude of the input audio signal. It is commonly used in FM radio broadcasting due to its resilience to noise.

Mathematically:

$$s(t) = A_c \cos\left(2\pi f_c t + 2\pi k_f \int_0^t m(\tau) d\tau\right)$$

Did You Know?

FM offers better signal-to-noise ratio than AM, making it ideal for high-fidelity audio broadcasts.

The message signal is denoted by $m(t)$, and the FM modulated signal is:

$$\begin{aligned} x_c(t) &= A_c \cos\left(2\pi f_c t + 2\pi k_d \int_0^t m(\tau)\, d\tau\right) \\ &= A_c \cos\left(2\pi f_c t + \varphi(t)\right) \end{aligned}$$

To convert it to baseband:

$$\begin{aligned} I(t) + jQ(t) &= A_c \cdot \text{LPF}\left[\cos(2\pi f_c t)\cos(2\pi f_c t + \varphi(t)) - j\sin(2\pi f_c t)\cos(2\pi f_c t + \varphi(t))\right] \\ &= \frac{A_c}{2} \cdot \text{LPF}\left[\cos(4\pi f_c t + \varphi(t)) + \cos(\varphi(t)) - j\left(\sin(4\pi f_c t + \varphi(t)) - \sin(\varphi(t))\right)\right] \\ &\approx \frac{A_c}{2}\left[\cos(\varphi(t)) + j\sin(\varphi(t))\right] = \frac{A_c}{2} e^{j\varphi(t)} \end{aligned}$$

So, we obtain:

$$\varphi(t) = \angle\left[I(t) + jQ(t)\right] = 2\pi k_d \int_0^t m(\tau)\, d\tau$$

Hence,

$$m(t) = \frac{1}{2\pi k_d}\frac{d\varphi(t)}{dt} \propto \frac{d}{dt}\angle\left[I(t) + jQ(t)\right]$$

Thus, for FM demodulation, we must extract the instantaneous phase angle. As a comparison, in AM demodulation we only measure the magnitude of the envelope.

To obtain the discrete-time signal, we use:

$$m[n] \propto \sin\left(\varphi[n] - \varphi[n-1]\right)$$

Use the above relation. (PL2)

Hint: To estimate

$$m(t) = \frac{d}{dt}\tan^{-1}\left(\frac{Q(t)}{I(t)}\right),$$

use the approximation:

$$\varphi[n] = \angle\left(I[n] - jQ[n]\right),$$

then:

$$m[n] \propto \varphi[n] - \varphi[n-1]$$

2 Familiarity with the SDR Module

2.1 Behavioral Model of the Module

To briefly describe this module: the RTL-SDR can be considered a block that, given a set of input parameters such as center frequency (f_c) and sampling rate (f_s)outputs a sampled complex baseband signal of the form:

$$\text{inphase}(I) + j \times \text{quadrature}(Q)$$

This signal is sampled at rate f_s and corresponds to the baseband representation of the received signal centered at frequency f_c.

2.2 Introduction to the RTL-SDR Module

The RTL-SDR (Software Defined Radio) is a low-cost USB dongle that can be used as a powerful and flexible RF front-end. Originally designed as a DVB-T TV tuner, it has been repurposed by the open-source community to serve as a general-purpose wideband software radio receiver.

With the help of software tools such as GNU Radio or SDR#, the RTL-SDR can receive and analyze radio signals in a wide frequency range (typically from 500 kHz to 1.75 GHz), making it ideal for learning about wireless communications, spectrum analysis, FM/AM demodulation, and more.

Below is an image showing typical RTL-SDR dongles used in the lab.

Figure 1: Standard RTL-SDR USB Dongle

3 Experiment Procedure: FM Demodulation Using RTL-SDR

3.1 Receiving the I/Q Signal from SDR#

To collect the raw FM radio signal:

- Connect the RTL-SDR to your PC and launch SDR#.
- Tune to a known FM frequency (e.g., 98.5 MHz).
- Select **WFM** mode.
- Enable I/Q output or recording.

3.2 Constructing I/Q Signal and Exploring its Frequency Content

The I/Q signal represents a complex baseband representation:

$$x(t) = I(t) + jQ(t)$$

It carries both phase and amplitude information and is essential for coherent demodulation.

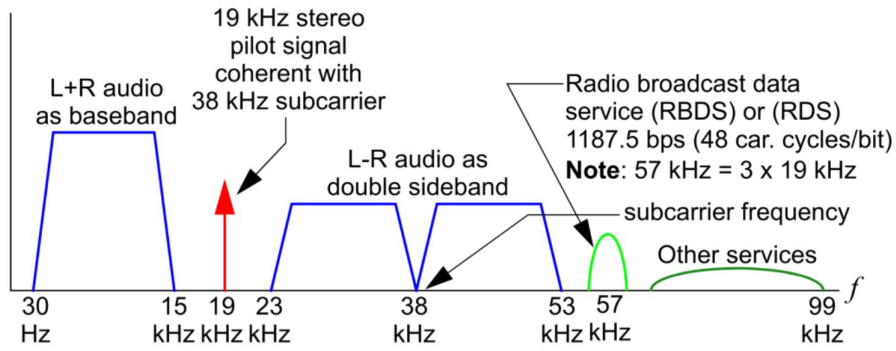

Figure 2: the frequency structure of an FM radio broadcast signal

3.3 FM Demodulation in MATLAB

Steps:

Step 1: Read the Recorded Signal

In the first step, we load the recorded IQ signal from a `.wav` file using `audioread`. The file should contain two channels: in-phase (I) and quadrature (Q) components of the sampled signal.

```
filename = 'test.wav';
[raw_signal, fs] = audioread(filename);
I = raw_signal(:,1);
Q = raw_signal(:,2);
complexSignal = I + 1i * Q;
```

Step 2: Low-pass Filtering

In this step, we design and apply a low-pass filter to isolate the desired FM station from the frequency-shifted signal. We use a FIR filter with an equiripple design.

```
    lpFilt = designfilt('lowpassfir', ...
    'PassbandFrequency', lp_cutoff, ...
    'StopbandFrequency', lp_cutoff * 1.5, ...
    'SampleRate', fs, ...
    'DesignMethod', 'equiripple');

    filteredSignal = filter(lpFilt, shiftedSignal);
```

Note:

Use linear-phase FIR filters to prevent distortion of audio signals.

Step 3: FM Demodulation

Now that we have isolated the desired FM signal, we proceed to demodulate it. The instantaneous phase of the complex baseband signal is extracted using the `angle` function. Taking the derivative (via `diff`) of the unwrapped phase yields the frequency deviation, which corresponds to the FM message signal.

```
    instPhase = unwrap(angle(filteredSignal));
    demodulated = diff(instPhase);
    demodulated = demodulated - mean(demodulated);
```

Step 4 (Optional): Decimation

To reduce the sampling rate to a standard audio rate (e.g., 44.1 kHz), we can optionally decimate the demodulated signal. This reduces computational load and prepares the signal for playback or further audio processing.

```
decimation_factor = floor(fs / audio_fs);
audio = decimate(demodulated, decimation_factor);
audio = audio / max(abs(audio));
```

Listing 1: Decimating the demodulated FM signal

Step 5: Listen to the Signal

In the final step, we play back the demodulated and decimated audio signal using MATLAB's `sound` function.

```
sound(audio, audio_fs);  % Play the audio at the target sampling rate
```

Listing 2: Playing back the FM audio

4 Working with Mono and Stereo FM Signals

4.1 Mono Audio Extraction

The mono signal occupies 015 kHz. After demodulation, use a low-pass filter with a cutoff of 16 kHz.

4.2 Suggested Extension: Stereo FM Demodulation

As an extension to this experiment, students are encouraged to modify their FM demodulation pipeline to support stereo audio.

FM stereo signals are more complex than mono and include:

- A **mono** (L+R) signal from 015 kHz,
- A **19 kHz pilot tone**, which indicates the presence of stereo,
- A **stereo difference** (LR) signal modulated as a double-sideband signal centered at 38 kHz.

To extract stereo audio:

1. Use a bandpass filter around 38 kHz to isolate the LR signal.
2. Use the 19 kHz pilot tone to regenerate a 38 kHz carrier for demodulating the LR component.
3. Combine the (L+R) and (LR) signals to recover left and right audio channels:

$$\text{Left} = \frac{(L+R)+(L-R)}{2}, \quad \text{Right} = \frac{(L+R)-(L-R)}{2}$$

This optional extension will deepen your understanding of signal modulation, filtering, and digital downconversion techniques in modern communication systems.

4.3 Steps for RDS Extraction

1. Bandpass filter around 57 kHz
2. Perform BPSK demodulation
3. Use Manchester decoding to extract bits

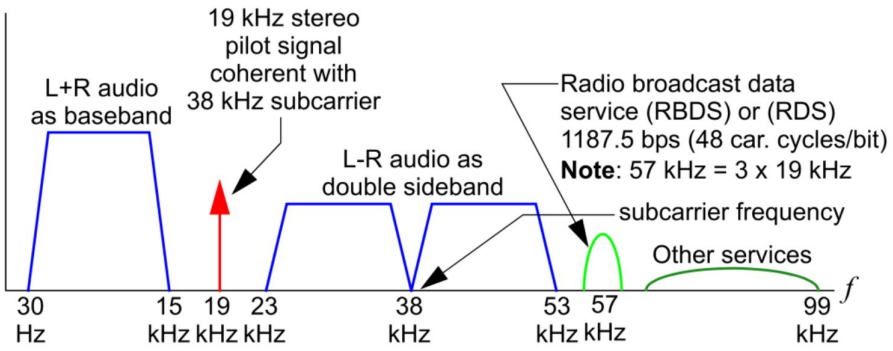

Figure 3: RDS Signal Spectrum (Centered at 57 kHz)