Pre Lab

1. What is the purpose of pre-emphasis/de-emphasis?

The purpose is to improve the Signal-to-Noise Ratio by reducing the effect of high-frequency noise. In FM transmission, noise power increases linearly with frequency since high frequencies have more noise. However, the power of real audio signals usually decreases at high frequencies. Pre-emphasis boosts the high-frequency components of the audio at the transmitter. De-emphasis attenuates the high frequencies at the receiver, restoring the original audio balance while simultaneously eliminating the high-frequency channel noise.

2. What modulation type is used for the L+R signal?

The L+R signal is unmodulated (Baseband) within the stereo multiplexing system. While the final signal transmitted over the air is FM, inside the stereo generator, the L+R component is not modulated onto a subcarrier. It is placed directly in the 0–15 kHz frequency range to ensure compatibility with older monophonic receivers.

3. What modulation type is used for the L-R signal?

The L−R signal uses DSB-SC (Double Sideband Suppressed Carrier) Amplitude Modulation. It modulates a 38 kHz subcarrier. The carrier is suppressed to save power and reduce potential interference, which is why a pilot tone is needed to recover it later.

4. How is the carrier generated for demodulating the L-R signal?

The carrier is generated by isolating the 19 kHz pilot tone and passing it through a frequency doubler. Since the L−R signal is DSB-SC (Suppressed Carrier), the receiver does not have the 38 kHz carrier required to demodulate it. The transmitter sends a 19 kHz pilot tone. The receiver filters this pilot out, doubles the frequency and uses that generated sine wave to demodulate the L−R signal coherently.

In this lab we designed and implemented a complete FM broadcast receiver chain starting from a softwaredefined radio (SDR) front end and finishing with audible mono and stereo audio playback. The Osmocom source was configured to a 264.6 kHz complex baseband sampling rate and tunes across the FM band from 88–108 MHz.

The first stage of the work focused on a mono FM receiver. In this design, the SDR output is passed through a 100 kHz low-pass filter to isolate a single FM channel before being fed into a PLL frequency detector that performs FM demodulation. The demodulated baseband is then processed by an FM de emphasis block (with a 75 μs time constant) to match broadcast pre-emphasis, resampled from 264.6 kHz down to a 48 kHz audio rate, and finally sent to an audio sink as shown in Figure 1. In parallel, the same demodulated signal is routed to a QT GUI frequency sink, which displays the power spectral density plot which is shown in figure 2. The PSD shows the L+R audio region, the 19 kHz pilot tone, and the higher frequency stereo subcarrier components even though the mono receiver itself only uses the L+R content.

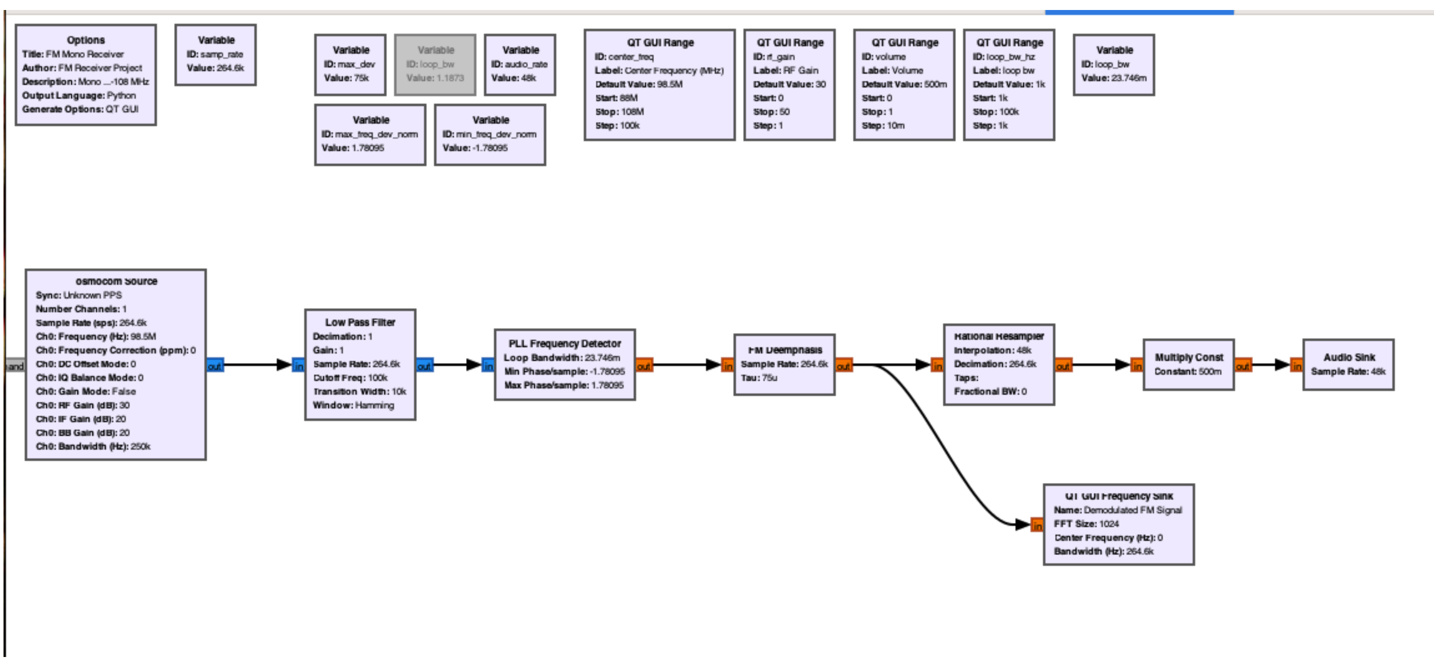

Figure 1: Block diagram of mono receiver

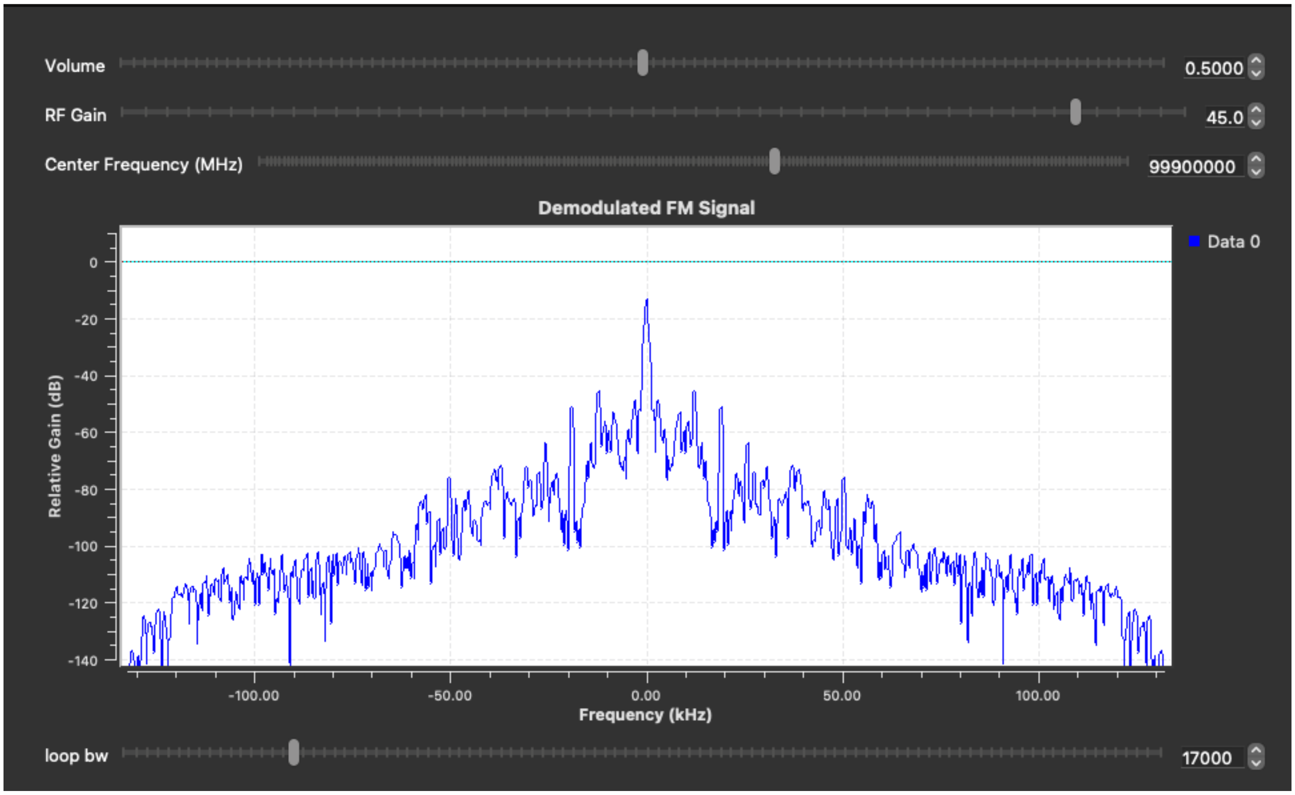

Figure 2: PSD of mono receiver

Then the design is extended to a full stereo FM receiver by actively processing all components visible in the demodulated spectrum. After de-emphasis, the composite baseband is split three ways. around One branch is low-pass filtered (0–15 kHz) to obtain a clean L+R (mono) signal. A second branch band-pass filters 19 kHz to isolate the pilot tone; this pilot is amplified with AGC and squared to generate a coherent 38 kHz carrier reference. The third branch band-pass filters the 23–53 kHz region containing the L−R double-sideband, suppressed-carrier (DSB-SC) stereo subchannel. By multiplying this L−R subchannel with the recovered 38 kHz carrier and low-pass filtering, we obtain a baseband L−R signal. Finally, a stereo matrix forms the individual left and right channels via L = (L+R) + (L−R) and R = (L+R) − (L−R), followed by appropriate gain scaling, de-emphasis, resampling, independent volume controls, and routing to a two-channel audio sink. This can be seen in figure 3.

Figure 4 shows the PSD plots of the receiver with both channels. At the output of the PLL frequency detector, the PSD of the composite demodulated signal is as expected and is picking up a broadcast FM baseband giving a broad low-frequency region extending from roughly 0–15 kHz corresponding to the L+R (mono) audio. There should be a sharp spectral line at 19 kHz showing the stereo pilot tone as well as higher-frequency components between about 23–53 kHz associated with the L−R double-sideband, suppressed-carrier stereo subchannel centered at 38 kHz. After stereo decoding and low-pass filtering, the PSDs of the left and right audio channels show pretty well confined energy within the 0–15 kHz band, with the pilot and L−R subcarrier components effectively removed.

Overall, the project progresses from a minimal mono receiver which is useful for understanding basic SDR interfacing, PLL demodulation, and audio post-processing to a fully featured stereo receiver that demonstrates coherent DSB-SC demodulation and stereo matrixing.

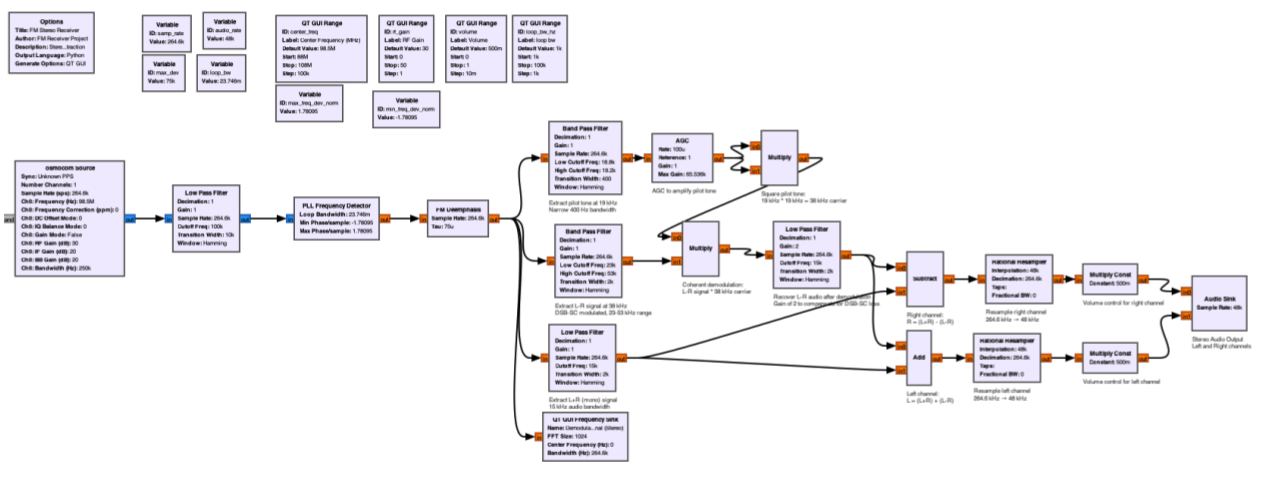

Figure 3: Block diagram of stereo receiver

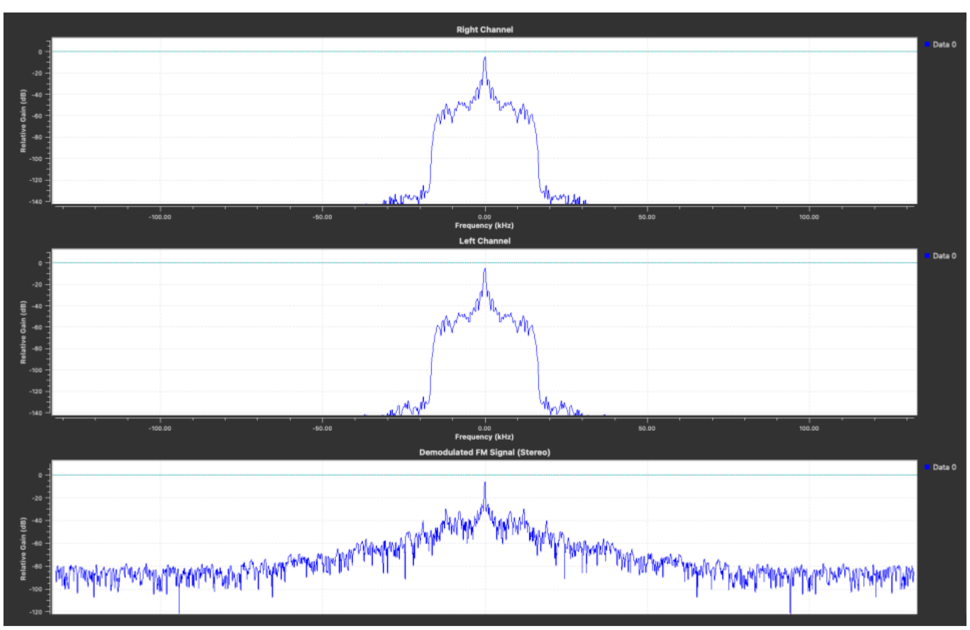

Figure 4: PSD from the stereo receiver