STEREOPHONIC FM BROADCASTING

Notes for EEL 4514 Communication Systems and Components
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1 Introduction

In monophonic broadcasting a single audio baseband signal is used to modulate a carrier; this signal originates from a single studio microphone. (If more than one microphone is used, their outputs are mixed to generate a single signal.) FM broadcasting was originally monophonic, and the FCC standards were established for mono. (Recall what the FCC standards are: a transmission bandwidth of 200kHz with maximum frequency deviation of $\Delta f = 75$kHz.)

Stereo uses two microphones (or two groups of microphones) to generate a left signal $L(t)$ and a right signal $R(t)$. These signals are to be fed into two identical speakers at the receiving end. With proper speaker positioning, the result is a more accurate reproduction of the soundstage than is possible with mono.

By the time engineers had figured out how to do stereo FM, mono FM had been around a long time and there were millions of mono receivers in use. Instead of changing the standards and forcing everyone to buy new stereo receivers, the FCC required that any proposed stereo scheme would have to be compatible with mono in the sense that any standard mono FM receiver would be able, without modification, to receive a mono version of a stereo transmission. Furthermore, the stereo signal is required to stay within the 200kHz transmission bandwidth limitation. Many stereo systems were proposed; the one in use today was adopted in 1961.

2 The Transmitter

The transmitter is shown in Figure 1. Note that the left and right signals are first bandlimited to 15kHz. The oscillator at $f_p = 19$kHz provides a *pilot carrier*, which will eliminate the need for a local oscillator at the receiver. The frequency doubler gives $f_{sc} = 38$kHz;

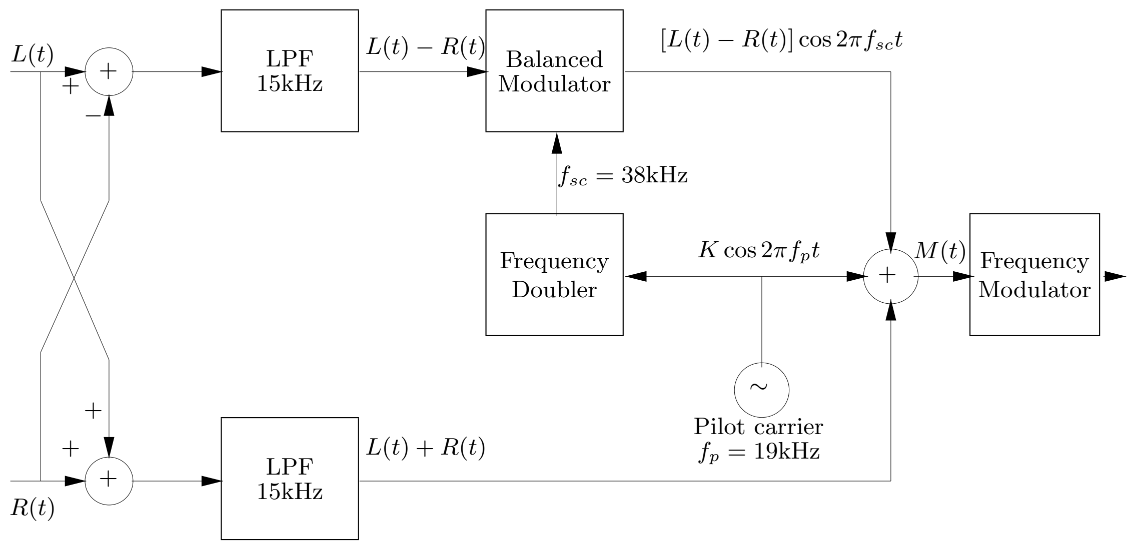

Figure 1: The Stereo Transmitter

this is called the *subcarrier*. The signal $L(t) - R(t)$ is applied to a balanced modulator; i.e., $L - R$ is used to generate a DSB-SC modulation of the subcarrier. The signal $M(t)$, which is used to produce the transmitted FM signal, consists of three parts: $L + R$, the modulation of the subcarrier by $L - R$, and the pilot carrier.

$$M(t) = [L(t) + R(t)] + [L(t) - R(t)] \cos 2\pi f_{sc} t + K \cos 2\pi f_p t.$$

The constant K determines the level of the pilot carrier relative to the other signals in M. A typical spectrum is shown in Figure 2. As we know, the maximum frequency deviation is set by the FCC to be $\Delta f = 75$kHz. For monophonic transmission, a baseband $B = 15$kHz results in a deviation ratio of

$$D = \frac{75}{15} = 5;$$

the transmission bandwidth then is

$$B_{FM} = 2(D + 1)B = 2 \times 6 \times 15 = 180\text{kHz}.$$

(Note that I used Carson's Rule, the validity of which is questionable when $D = 5$, but it gives us an idea of the bandwidth.) For stereo transmission, the baseband signal $M(t)$ has bandwidth 53kHz (see Figure 2), resulting in a much lower deviation ratio:

$$D = \frac{75}{53} = 1.42.$$

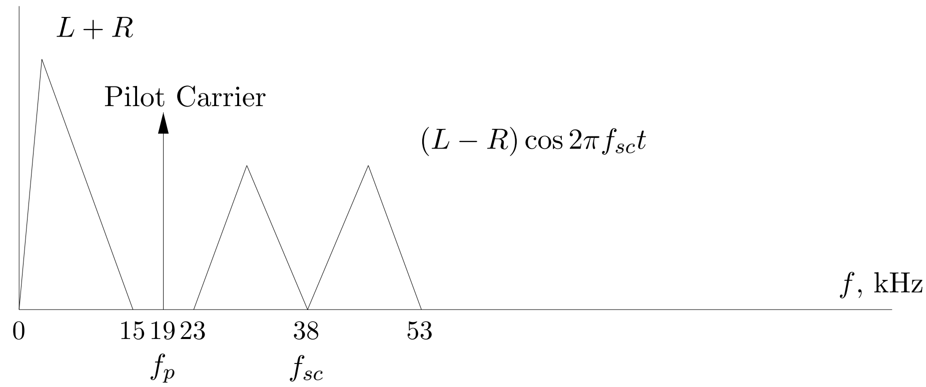

Figure 2: Spectrum of $M(t)$

The bandwidth now is (again, using Carson's Rule despite its drawbacks)

$$B_{FM} = 2 \times 2.42 \times 53 = 256.5\text{kHz}.$$

This exceeds the 200kHz limit; hence for stereo transmission, Δf must be reduced. (It is left to you to verify that $\Delta f = 47$kHz will meet the bandwidth limitation; this results in $D = 0.887$.) Note that in stereo transmission, we have a small deviation ratio D; i.e., stereo FM must be essentially $narrowband$ FM, and so we sacrifice the tremendous gain in output SNR that is possible with wideband FM. The main reason FM sounds better than AM is that the baseband signal bandwidth in FM is 15kHz, while in AM it is only 5kHz (remember that the AM transmission bandwidth is restricted to 10kHz).

3 The Receiver

The receiver first recovers $M(t)$ as the output of a frequency discriminator. The job then is to sort $M(t)$ into its component signals and thereby recover $L(t)$ and $R(t)$. The receiver is diagrammed in Figure 3. Note that we get f_{sc} at the receiver by filtering out the pilot carrier and using a frequency doubler; a local oscillator is not required. This is better than transmitting f_{sc} and trying to recover it directly by filtering.

This system is compatible with a mono receiver. In such a receiver, $L+R$ passes through the baseband filter, while the pilot carrier and the DSB-SC signal do not.

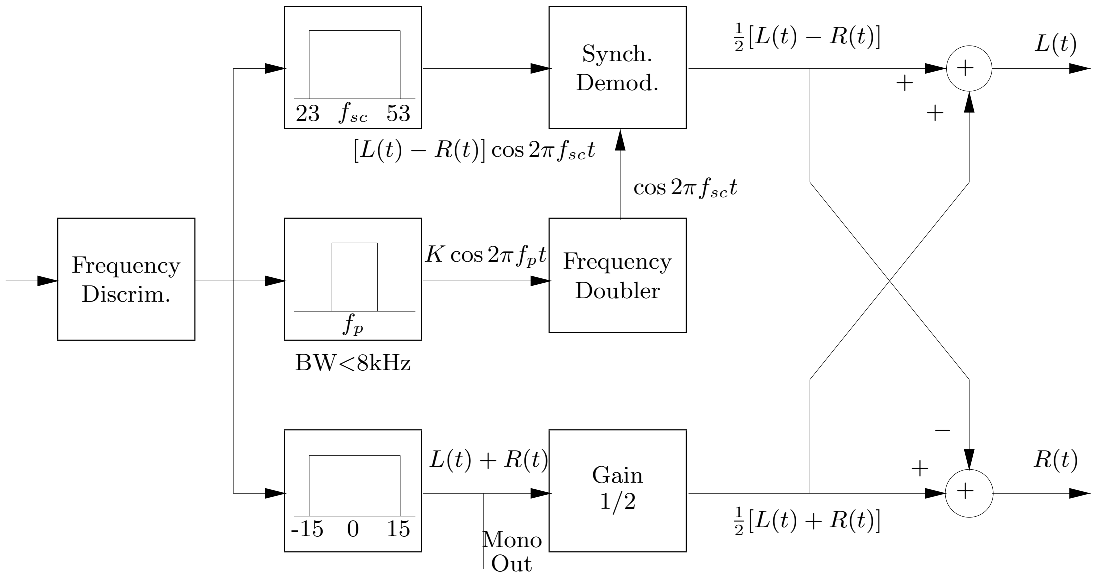

Figure 3: The Stereo Receiver

4 Sensitivity to Noise

A serious problem introduced by this stereo transmission system is that it is much more sensitive to noise than mono broadcasting. The problem is caused by the $L - R$ channel.

If we assume that the channel adds white noise, $\mathcal{S}_n(f) = N_0/2$, to the transmitted signal, the power spectral density of the noise out of the frequency discriminator will be

$$\mathcal{S}(f) = Kf^2,$$

where K is a positive constant depending on N_0, the carrier amplitude A_c, the attenuation of the channel, and the frequency deviation Δf produced by the transmitter. Therefore the noise powers for the two channels are

$$\begin{aligned} P_{N,L+R} &= 2\int_0^{15{,}000} Kf^2 df = 2.25 \times 10^{12} K, \\ P_{N,L-R} = &= 2\int_{23{,}000}^{53{,}000} Kf^2 df = 91.14 \times 10^{12} K. \end{aligned}$$

That is, the noise power on the $L - R$ channel is 40.5 times the noise power on the $L + R$ channel. Hence, we conclude that in a noisy environment, mono FM transmission would be preferred. This is also the reason that most home stereo systems have a switch labeled "mono"—if the noise is affecting the $L - R$ channel severely, the mono switch cuts out the $L - R$ channel and simply feeds the $L + R$ signal into the speakers, effectively turning the receiver into a mono receiver.