19. RADIO BROADCASTING

19.1 Introduction

The aim of radio broadcasting is to inform and to entertain the people. Consequently, the properties of the human hearing have to be taken into regard when the quality parameters of the radio broadcasting systems and that of the receivers are specified. Fluctuations of the air pressure with frequencies between 16 Hz and 20 kHz are sensed as sound by the human ear. Within this frequency range, the ear is able to sense sound intensities within the 100-120 dB wide range which lies between the threshold of hearing and the threshold of pain.

Great series of subjective measurements have been carried out to show that almost lifelike sound perception can be achieved by moderate sound parameters. Parameters of such a -high fidelity- system can be summarized as follows: 30 Hz -15 kHz frequency range, 40-60 dB dynamic range (the ratio of the loudest and the weakest sound intensity), about the same signal-to-noise ratio and nonlinear distortion less that 1%. Beside these parameters, the directivity of the hearing is also an important factor.

Because of economical reasons, high fidelity sound transmission cannot be considered as a general requirement. High fidelity sound transmission is used mainly for programs with a high degree of artistic impression. Analog FM and digital broadcasting systems are used for this purpose. The majority of programs, however, is broadcasted by analog AM systems with moderate quality parameters.

19.2. AM Broadcasting

19.2.1. Basic parameters

The frequency ranges internationally accepted for AM broadcasting are as follows:

Long Wave	(LW)	150 - 300 kHz,
Middle Wave	(MW)	520 - 1600 kHz,
Short Wave	(SW)	2.3 - 26.1 MHz
		(12 consistent bands, each 0.1 - 0.5 MHz wide).

AM-DSB systems are used for AM radio-broadcasting . As it is known, the bandwidth required by an AM-DSB transmitter is twice the maximum of the modulating frequency. This means that for high quality transmissions $B = 2\,f_M = 2\cdot 15 = 30$ kHz would be required for each radio station. Comparing this value with the bandwidth of the MW band, less than 40 stations could be operated within a certain geographic area.

For this reason, the maximum modulation frequency of AM radio broadcasts has been limited by international agreements to 4.5 kHz; i.e. a 9 kHz wide frequency band is used by one station so that about 120 stations can operate simultaneously within the same geographic area. Obviously, such a low value of the maximum modulating frequency does not make it possible to broadcast good quality music programs. (Let us remark that just for this reason many countries brake the agreement and use modulation frequencies as high as 12-15 kHz.) Owing to the particular features of AM systems, maximum a 30 dB signal-to-noise ratio and the same dynamic range can be achieved. Since the dynamic range of an orchestra is about

80 dB, the signals of musical programs have to be significantly compressed before transmitted which further reduces the available quality.

19.2.2. AM Transmitters

The AM-DSB signal is produced by a modulator fed by a high-frequency sinusoidal carrier and by the modulating audio-frequency signal. The carrier frequency is generated by a high-precision oscillator and carried to the modulator via a power amplifier. The audio-frequency signal, produced in the studio comes first through a switched distribution network and then it is connected via a modulator amplifier to the other input of the modulator. The AM-DSB signal appearing on the modulator output is amplified by a linear power amplifier. The high-power AM signal is then coupled to the antenna and transmitted.

One of the most important parameter is the nominal value of the carrier frequency. This should be kept very precisely constant which practically means that relative frequency deviations should be less than 10^{-4}-10^{-7}. Such an accuracy can be achieved only by thermally-controlled electromechanical oscillators.

The frequency determining element of such an oscillator is a quartz crystal so that the accuracy of the oscillator is determined by the crystal parameters, mainly by its linear thermal coefficient. The thermal coefficient depends on the direction the crystal slice was cut with respect to the crystal axes. Optimal cuts, e.g. the so-called DT cut, have a thermal coefficient as low as 10^{-9} which is good enough for the above application.

In spite of the great development in semiconductor technology, electron tubes are still used in the output stages since very great transmission powers are required (e.g. the Kossuth receiver, located at Solt operates with a 2 MW output power). In the course of the transmission, a very great amount of energy is dissipated, i.e. converted to heat even if the efficiency is relatively high.

For instance, about 20 - 40 kW out of 100 kW output power is converted to heat thus the cooling of the transmitter is of primary importance. The cooling methods (air-flow, water-flow, evaporation) always depend on the local geographic and climatic conditions and on the value of the transmitted power.

19.2.3. Basic Principles of AM Signal Reception

Selectivity, gain, demodulation and gain control are the essential functions to be solved in both AM and FM receivers. Several other signals and noises reach the antenna side by side with the signal we want to receive. The selectivity of a receiver is meant as its ability to select the desired signal out of the other signals. One the most important parameters of a receiver is its *near selectivity*, i.e. how the signals of the neigbouring frequency bands are suppressed by the receiver. For AM receivers, this is specified by the attenuation level at frequencies located 9 kHz from the received carrier frequency to the level of the received carrier. The distortion of the received signal caused by the selection is also an important parameter.

It is very important for a receiver to be able to select and demodulate the signal of the received station even if the input signal level is low. This is possible only if the receiver has great reserves of amplification. The receiver gain can be characterized by the signal-limited and the noise-limited sensitivity. The *signal- limited sensitivity* of AM receivers is defined as the effective value of that high frequency carrier which -if modulated in 30 percent by a 1 kHz sinewave- produces an audio frequency signal with 50 mW power at the output (loudspeaker) of the receiver. The *noise-limited sensitivity* is defined by the effective level of that high frequency carrier which produces an audio frequency output with 26 dB signal-to-noise ratio.

The amplification and the noise properties can be characterized even better by the demodulated audio frequency output level as the function of the high frequency input signal (modulated by a single sinewave) or by the demodulator output noise level as a function of the level of the unmodulated carrier. Such a diagram, typical for AM receivers, is shown in Fig. 19.1.

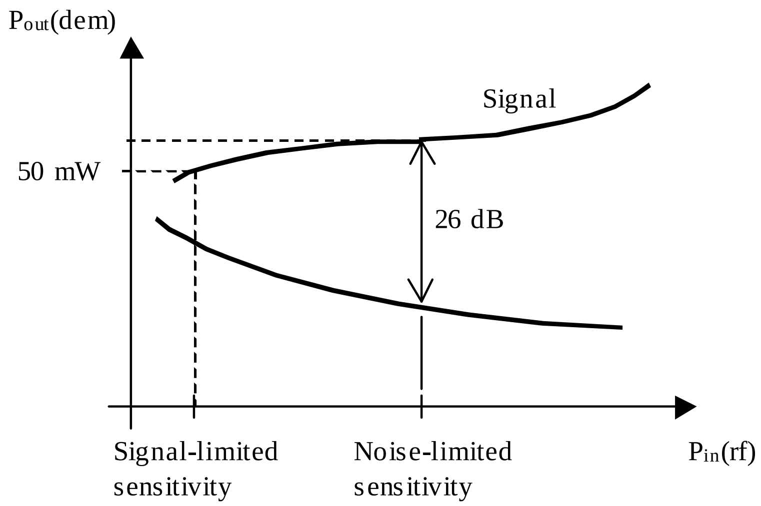

Figure. 19.1. Signal and noise level at the demodulator output

Another diagram typical to characterize a receiver is shown in Fig.19.2. This diagram illustrates how the nonlinear distortion varies as a function of the high frequency input signal (modulated by a single sinewave).

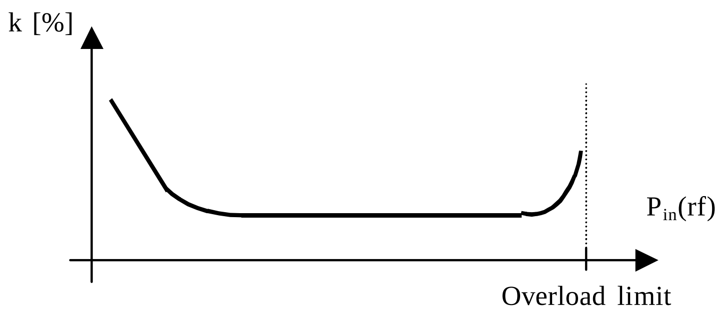

Figure. 19.2. Nonlinear Distortion of the Demodulated Signal

So far the question of the gain-control has not been touched. Both above diagrams are really very important since the magnitude of the high frequency input signal appearing at the output of the receiver antenna might vary even several orders. The reasons of that are the following:

- the distances between the receiver the various transmitters are different,
- the output power of the transmitters is different,
- some propagation losses are time-variant (fading),

- the distance between the transmitter and the receiver and shadowing by terrain obstacles may continually change (e.g. in the case of car radios).

A good quality receiver has to provide for proper selectivity and demodulation conditions regardless of great changes in the level of the signal reaching the antenna. Even more, the level of the demodulated signal has to be kept as constant as possible regardless of the level of the input signal. These requirements can be satisfied only if the gain factor of the receiver automatically varies as a function of the high frequency input signal. This function is accomplished by the so-called *automatic volume control* (AVC) unit.

19.2.4. AM Receivers

Although system considerations of AM receivers will be treated in this Chapter, the majority of the following conclusions is valid for the FM receivers as well.

Two different types of AM receivers exist: the *direct* and the *superheterodine* receiver. Being simpler, the direct receivers were constructed earlier. The block diagram of such receiver is shown in Fig. 19.3. The receiver antenna is connected to a tunable amplifier which is tuned to the required carrier frequency. The amplified signal goes directly to the demodulator and the demodulated signal is amplified by an audio frequency amplifier which feeds the loudspeaker. Disregarding the advantage of simplicity, direct receivers have many unfavourable properties:

- sensitivity significantly varies over the frequency range,
- the receiver is unstable (susceptible to self-oscillations),
- near-selectivity is bad and varies over the received range,
- the handling of the receiver is uncomfortable (the output loudness has to be changed frequently).

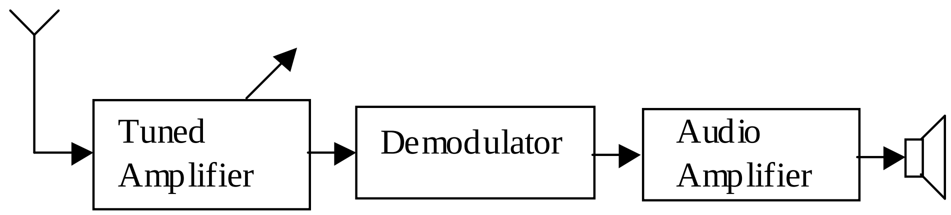

Figure. 19.3. Direct Receiver

Nowadays, direct receivers are not used at all, they have been mentioned only to present the historical development. In 1925, the superheterodine (frequency- transposition) principle was invented by Armstrong. Quality parameters of superheterodine receivers (or simply superhets) are incomparably better than those of the direct receivers. The block diagram of a superhet is shown in Fig. 19.4.

The main reason of the great quality improvement is the transposition of each input frequency to a predetermined constant frequency called the *intermediate frequency*. As the consequence of this transposition, the signal which has to be selected and amplified has a constant frequency.

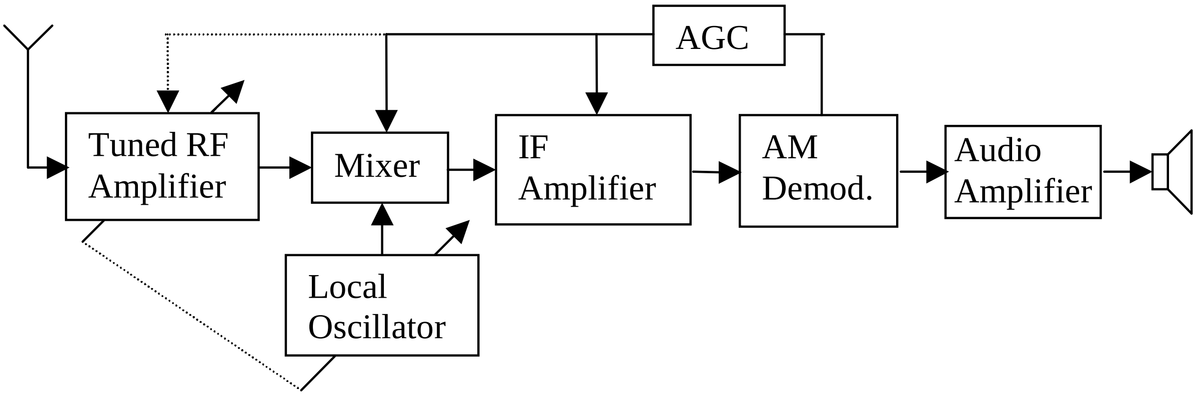

Figure. 19.4. Superheterodine AM Receiver

The required selectivity and gain are performed by the so-called intermediate frequency (IF) amplifier. Since it is much more easier to construct a stable, high-gain, selective amplifier for a single relatively narrow band than to meet the same specifications over a wide (entire) range of frequencies, all drawbacks of direct receivers have been eliminated at once by this invention.

The IF signal is generated by *mixing* which means the multiplication of the received modulated carrier by the (unmodulated) signal of a local oscillator. To provide sufficient input preselection and to have the same resulting frequency (IF) for different input carriers, the input circuit and the local oscillator have to be tuned in tandem. This is indicated by the dashed line between the input circuit and the oscillator.

The principle of mixing is shown in Fig. 19.5. If the modulated input signal with carrier frequency f_c and the unmodulated sinewave of the local oscillator f_o ($f_o > f_c$) are multiplied, the sum ($f_o + f_c$) and the difference ($f_o - f_c$) of these frequencies will appear at the multiplier output both retaining the original modulation content.

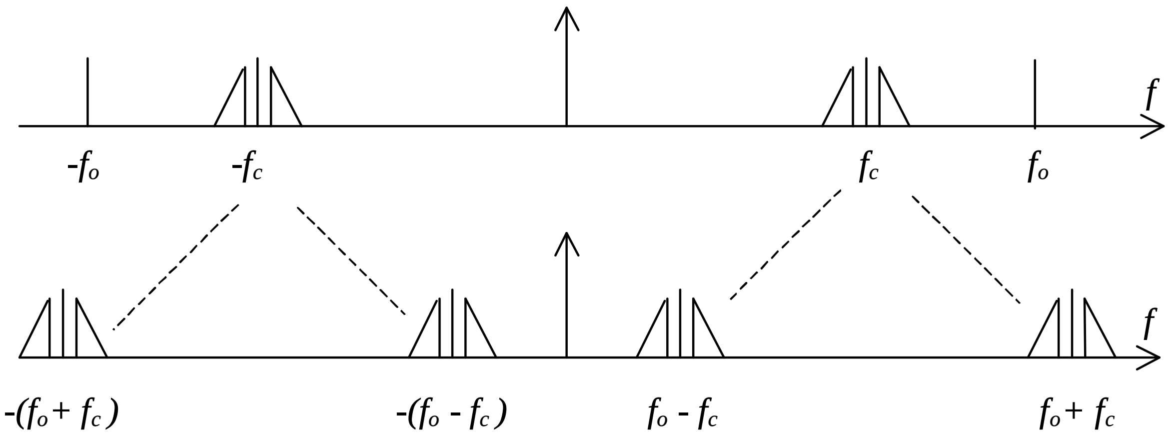

Figure 19.5. Principle of the conversion

So the frequency of the modulated signal has been transposed into two new locations, both of which can theoretically be used as intermediate frequencies Generally, the (f_o - f_c) component is selected at the input of the mixer and amplified by the IF amplifier.

Mixing can obviously be done with a local oscillator which has a lower frequency than the input frequency ($f_o < f_v$). This is called lower-band mixing while the previous case is the so-called upper-band mixing. Generally, upper-band mixing is used in AM receivers.

19.3. FM Broadcasting

19.3.1. Basic Parameters

FM radio broadcasting stations are operating in the very high frequency (VHF) range. Frequency bands internationally accepted and specified by the OIRT and CCIR recommendations for the European FM broadcasting are as follows: 65-73 MHz (OIRT) and 87-106 MHz (CCIR). In the USA, the range between 88 and 108 MHz is used. Some years earlier, the East European countries were governed mostly by the OIRT recommendations while the western countries applied the CCIR recommendations. Actually, the eastern countries are joining CCIR, too while maintaining OIRT transmissions (because of the great number of receivers used in this range).

As it was shown in Chapter 11., the FM signal requires a considerably greater bandwidth as compared with the maximum modulation frequency. Suppose to have a maximum modulation frequency of 15 kHz and using 50 kHz frequency deviation, a bandwidth of 180 kHz is required for a mono signal. This is one of the reasons why FM cannot be used economically in MW and SW bands.

The VHF range and FM are used for high quality audio broadcasting. This is not only due to the bandwidth available here but also to the signal-to-noise ratio and the dynamic range of FM transmission which are much better than those of AM transmission. Under optimal conditions, a 45-50 dB dynamic range can be achieved so that the signals need not be compressed as much as when AM is used.

Beside several favourable features, cost is the only drawback of FM transmissions in the VHF range. Since in this range the waves propagate line-in-sight (the transmitter and the receiver antenna have 'to see' each other), the area covered by one transmitter is small. This is so even in such relatively small countries as Hungary. To cover the whole country with the program, several transmitters have to be installed which is especially expensive if stereo programs have also to be transmitted since the receiving area of the stereo signal with the same signal-to-noise ratio as that of the mono signal is even smaller.

19.3.2. FM Transmitters

Some technical problems of transmission such as frequency stability and cooling have already been discussed thus only the particular features of FM transmitters will be considered here. The simplified block diagram of a conventional FM transmitter is shown in Fig. 19.6. Because of better frequency stability, *indirect modulators* are used in FM transmitters. An indirect FM modulator (see Fig. 19.7), however, can produce only an NBFM signal. To achieve a 50 kHz frequency deviation (OIRT standard), multiplication by about $n = 10^3$ is necessary. For such great values of n, a multistage multiplier has to be used which -for sake of simplicity- is represented by a single stage in the block diagram of Fig. 19.6.

Obviously, the oscillator frequency of the indirect modulator must also be multiplied by n. Since the carrier frequencies of different broadcasting stations are not necessarily multiples of n, the carrier frequency is set by an additional frequency transposition produced by the quartz oscillator and multiplier shown in Fig. 19.6. It is important to note that the stability of the carrier frequency is determined by the stability of both quartz oscillators (multiplication has no effect on the frequency stability). Therefore both oscillators have to be of high stability (10^{-7}-10^{-8}) which is achieved by putting the quartz crystals into a thermally-controlled oven.

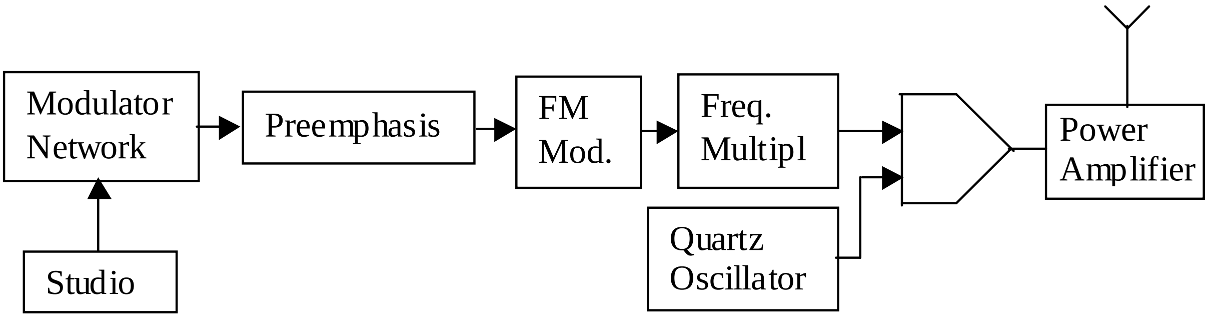

Figure 19.6 Block Diagram of an FM Transmitter

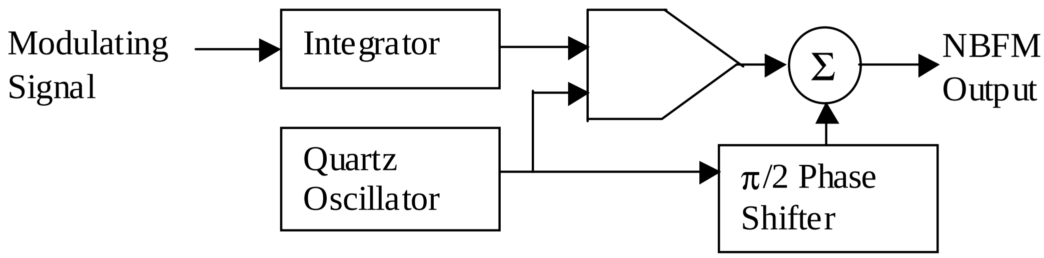

Figure 19.7 Indirect FM Modulator

19.3.3. FM receivers

The block diagram of a superheterodine FM receiver is very similar to that of an AM receiver. Some important and less important differences will be discussed on the base of the block diagram shown in Fig. 19.8.

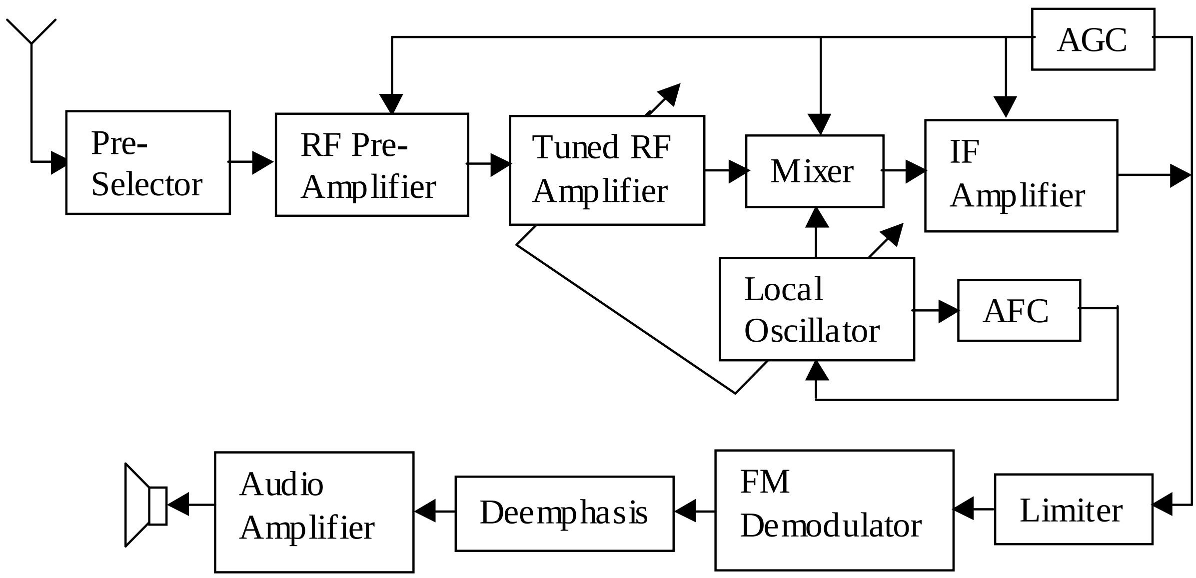

Figure. 19.8. Superheterodine FM Receiver

In the VHF range, selective antennas -such as the direct or the folded dipole of half wavelength- are generally used. The first stage of the receiver (the antenna is connected to) is a bandpass filter (and amplifier) with almost flat response within the VHF range so that it is unnecessary to be tuned. A preamplifier preceding the mixer is almost always used in FM receivers because of the low noise factor achieved in this way. Consequently, the noise-limited sensitivity significantly increases, too.

The structure of the mixer and the oscillator are the same as for the AM receiver. The only difference is the Automatic Frequency Control (AFC) circuit improving the stability of the local oscillator.

The design aspects and requirements of the IF amplifiers as well as the criteria of IF choice are also very similar to those of the AM receivers, of course the actual values of IF are different; while about 460 kHz IF and 9 kHz bandwidth is used in AM receivers, 10.7 MHz IF and about 300 kHz bandwidth (stereo!) are the typical values for IF stages of FM receivers. The only essential difference is that the phase response -or the group delay response- of FM IF amplifiers have also to be specified since great changes in these parameters may entirely degrade the binaural effect of the stereo transmissions. Without going into details let us remark that for good quality stereo transmissions the group-delay ripple should not exceed 2 s over the whole transmission band.

19.4. Stereo Broadcasting

To achieve a binaural effect, two independent signals have to be transmitted simultaneously, $L(t)$ for the left ear and $R(t)$ for the right ear. This must be solved so that the stereophonic broadcasting has to be *compatible* with the monophonic one, i.e. it must not interfere in any way with the existing monophonic broadcasting and it must stay within the same bandwidth, without the need to change frequency allocation of the stations.

Moreover, compatibility means that a person listening to a stereophonic broadcasting on a monophonic receiver can enjoy the program (as monophonic, of course) and inversely: programs broadcasted in mono have to be enjoyable on stereo receivers (as monophonic) regardless whether the stereophonic system has been implemented for AM or for FM broadcast.

19.4.1. FM Stereo Broadcasting

Several procedures for a compatible transmission of stereophonic signals have been worked out, but -with exception of some few cases- the so-called *pilot-carrier stereo MPX* procedure is the only one used in practice. The spectrum of the left and the right signals encoded by this procedure is shown in Fig. 19.9. For the sake of compatibility with mono receivers, the sum of two signals [$M(t) = L(t) + R(t)$] carrying the monophonic information is left untouched in the baseband (20 Hz - 15 kHz) and the difference of the signals [$S(t) = L(t) - R(t)$] is transposed above the audio frequency range not disturbing thus the operation of monophonic receivers.

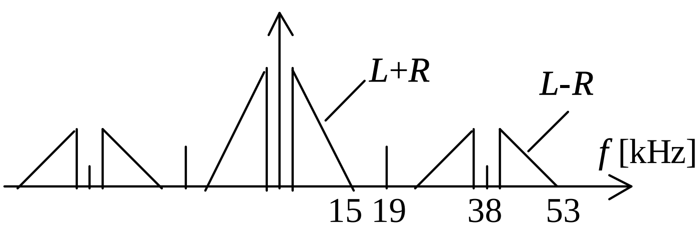

Figure 19.9. Spectrum of the Stereo MPX Signal

Basically, two different principles, the *time-division* and the *frequency-division* can be used for the pilot-carrier stereo MPX encoding. For the sake of its operational simplicity, the *frequency-division* encoder will be discussed here.

The block diagram of the encoder is shown in Fig. 19.10. The left and the right signals first go through the so-called preemphasis circuits, which in fact are two identical amplifiers and/or filters with increasing amplitude response towards the high-frequency end of the audio range. Then the signals are 'matrixed', i.e. added and subtracted, resulting in the $M(t) = L(t) + R(t)$ and the $S(t) = L(t) - R(t)$ signals.

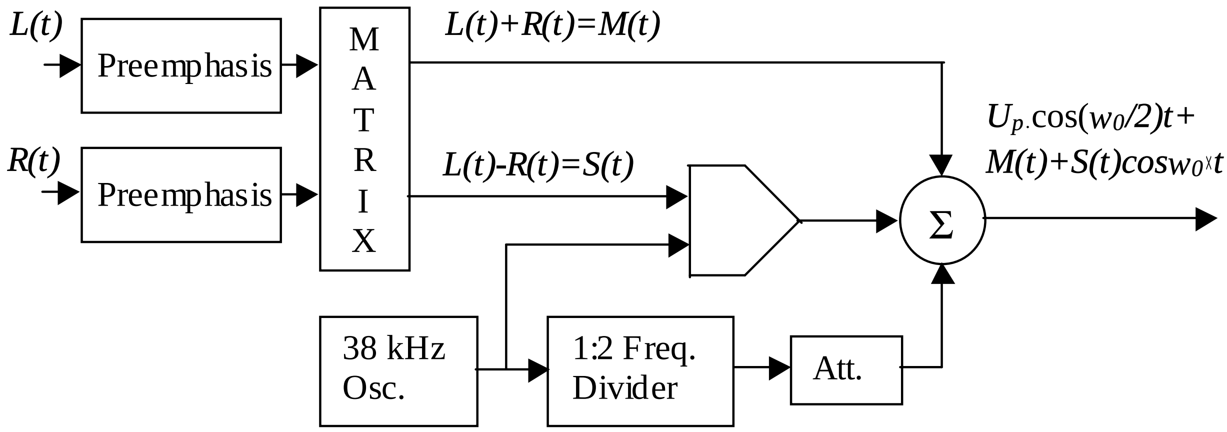

Figure 19.10 Block Diagram of a Stereo Encoder

To transpose the difference signal over the audio-frequency range (as shown in Fig. 19.9), it is multiplied by a 38 kHz subcarrier. The subcarrier is suppressed at the multiplier output so that an AM-DSB/SC signal is produced. To reproduce the left and the right signal with the proper phase at the receiver, the phase of the subcarrier has also to be transmitted somehow. For this reason the a pilot-carrier is generated. This is a 19 kHz signal derived from the subcarrier in such a way its frequency is divided by two and its amplitude is attenuated. The composite stereo MPX signal is obtained by summing the monophonic, the transposed difference and the pilot signals ($M(t)$, $S(t)\cos\omega$ and $U_p\cos(\omega/2)$).

To demodulate the composite stereo MPX signal, again, either the frequency-division or the time-division method can be used. Although the time-division decoder -being cheaper (easier to integrate)-is mostly used, the frequency-division decoder is presented here as the complementary part of the encoder discussed above. The block diagram of the decoder is shown in Fig 19.11.

The stereo MPX signal is first divided into three following bands:

- the monophonic component $M(t) = L(t) + R(t)$ is obtained at the output of the lowpass filter with 15 kHz cut-off frequency,
- the differential component $[L(t) - R(t)]\cos\omega_o t$ is selected by a bandpass filter with cut-off frequencies at 23 kHz and 53 kHz,
- the pilot signal, carrying the phase information necessary for decoding the $L(t)$ and $R(t)$ signals is filtered by a narrowband filter, tuned to 19 kHz.

To decode the stereophonic signal, the differential component $S(t)$ carried by the frequency ω_o (38 kHz) has to be mixed into the baseband. This is done by a multiplier, the second input of which is controlled by the pilot carrier being previously amplified and doubled in frequency. Following that, the left and the right signals are easily reconstructed in a matrix circuit. To compensate the preemphasis performed in the transmitter, the $L(t)$ and the $R(t)$ signals are passed through a deemphasis circuit, producing thus a significant improvement of the signal-to-noise ratio.

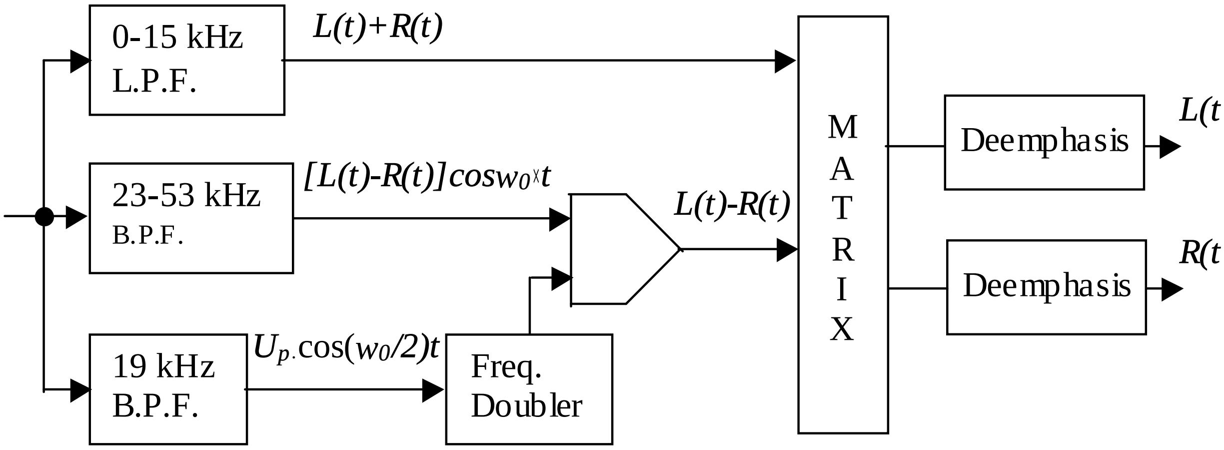

Figure 19.11 Block Diagram of a Stereo Decoder

19.5. Digital Broadcasting

To transmit an analog signal through a digital channel, first it has to be converted into digital. The main steps of the analog-to-digital conversion are shown in Fig. 19.12. The analog signal -actually a sound- is passed to the sampling circuit through a lowpass filter, whose cut-off frequency will depend on the required quality (and, of course, on the sampling frequency). For speech quality, a cut-off frequency of 3.4 kHz is used while the good quality transmission of music requires 15-20 kHz (depending on the sampling frequency). As determined by the sampling theorem for the undistorted reconstruction of the sampled signal, a sampling frequency at least twice as high as that of the bandwidth of the baseband analog signal is necessary. Consequently, this frequency is usually 8 kHz for speech communication while for good quality musical transmissions at least 32 kHz have to be used.

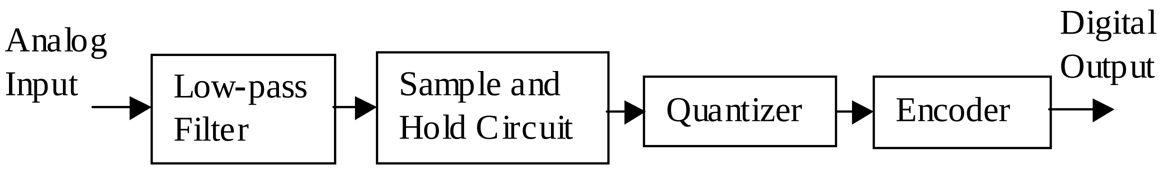

Figure 19.12 Block Diagram of an A/D Converter

Following the sampler, the signal discrete in time is ’rounded' by the quantizer. Quantization word-length of the quantizer depends on how great signal-to-noise degradation caused by quantization is allowed. As the quatization word length, 8 bits are used for speech transmission and 12-16 bits for music transmission. Using different code-compression procedures based on the redundancy of the signals, these values can be significantly reduced. Following quantization, the final length of the binary code is determined by the actual coding procedure performed by the encoder.

Because of the relative complexity of these systems, it is not possible to give a detailed presentation, instead of which we confine ourselves to an introduction and a brief description of the digital audio transmission systems.

The DSR (Digital Satellite Radio) system transmits 16 different good quality stereophonic programs. The sampling frequency is 32 kHz, the original sample length is 16 bit. To reduce the bit rate, this is companded to 14 bit before transmission.

The MAC-packet (Multiplexed Analog Components) has also been developed for DBS (Direct Broadcasting Satellite) systems. Although essentially intended for TV broadcasting, beside the TV accompanying sound, it is suitable to transmit other sound programs as well. In the C-MAC version, the samples of the sound channels are collected into packets of 751 bits and then -depending on the sampling frequency (32 or 16 kHz), on the source coding (14 bit-linear or 10 bit-companded) and on error-correction coding- 8 high quality mono or 4 high quality stereo or 16 average quality mono channels are simultaneously transmitted. The D2-MAC is a similar version, requiring less bandwidth but having only half the number of channels.

The DAB (Digital Audio Broadcasting) system is suitable both for terrestrial and for satellite broadcasting. The system uses a 3.5 MHz wide range to transmit 12 stereophonic signals of excellent quality. The sampling frequency is 48 kHz and the word length is 16 bit. The data are compressed before being transmitted. The compression ratio is 6:1 without causing any noticeable degradation of quality.

The NICAM (Near Instantaneously Companded Audio Multiplexed) system is also related to the TV broadcasting. Beside the conventional FM sound, a digitally encoded sound channel (32 kHz sampling frequency, 14-to-10 bit companded word length) is transmitted by this system.

Control Questions

1. What are the frequency bands used for AM and for FM broadcasting?
2. Functional block diagrams of AM receivers.
3. What are the advantageous properties of frequency transposed receivers?
4. Block diagram of a superheterodine FM receiver.
5. What types of digital audio broadcasting systems do exist?

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Abbreviations

AFC	Automatic Frequency Control AM Amplitude Modulation
AVC	Automatic Volume Control
CCIR	Comité Consultative Internationale des Radiocommunications
DAB	Digital Audio Broadcasting
DBS	Direct Broadcasting Satellite
DSB	Double Sideband
DSR	Digital Satellite Radio
FM	Frequency Modulation
LW	Long Wave

IF	Intermediate Frequency
MW	Middle Wave
MAC	Multiplexed Analog Components
NICAM	Near Instantaneously Compounded Audio Multiplexed
OIRT	Organisation Internationale de la Radiocommunication et Television
SW	Short Wave
VHF	Very High Frequency Range