FM STEREOPHONIC TRANSMISSION TECHNIQUE

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1. FUNDAMENTALS OF FM

1.1 The Beginning of FM

It is well know that, in amplitude modulation, upper and lower side-bands appear and a band-width of twice the modulated frequency will be normally required. As the noise tends to increase with the increase of band-width, in the initial stage of telecommunication, frequency modulation was considered to be used for the object of reducing the noise and secure as many as transmission channels as possible to save the band-width.

Although, it was simply considered that, to attain the foregoing objective, it would only be necessary to minimize the variation of frequency due to modulation. However, this is erroneous as side-bands would be generated over a wider band than in amplitude modulation, as logically clarified by Carson in 1922. As later experiments indicated poor S/N ratio, the FM system was discarded for a time.

It may be said that the development of the FM system we see today was launched in 1936 when Armstrong put forth his wide-band FM system and, proved that the S/N ratio could be improved by increasing the frequency deviation. As the band-width increases with increases with the increase of frequency deviation, FM can only be used above the VHF bands due to frequency allocation reasons. However, with the advancement of VHF technology after the World War 2, the usage of FM expanded to various fields.

The FM broadcast is considered to be a high fidelity broadcast, but the features of FM broadcast is in reducing interference and improving the S/N ratio. The S/N is of course an important element of high fidelity broadcast, but the direct advantage of FM is not because the frequency band is wide, but because it is in the VHF band, and the frequency band can be occupied wide! Further, in amplification modulation, in case the degree of modulation exceeds

100%, distortion will arise, but in case of FM, there is no such boundary. The noise will be less and the availability of occupying a wide dynamic range are considered to be the reason for high fidelity.

1.2 Logical Analysis of Frequency Modulation and Phase Modulation Wave Forms

Let us consider that high frequency current may be expressed as follows.

$$i = I_0 \sin\phi\ (t) \tag{1}$$

$\phi(t)$ here is the electrical angle of time t

The instantaneous angular frequency ω of this current and the instantaneous frequency f may be expressed by the following equation.

$$\omega = 2\pi f = \frac{d\phi\ (t)}{dt} \tag{2}$$

On one hand, the instantaneous frequency of that modulated by voice frequency f_m may be expressed by the following equation.

$$f = f_0 + f_d \cos 2\pi f_m t \tag{3}$$

f_0 is the carrier frequency when unmodulated and f_d the frequency deviation.

If equations (2) and (3) are combined, we have

$$\frac{d\,\phi\,(t)}{dt} = 2\pi\{ f_0 + f_d \cos 2\pi f_m t \} \tag{4}$$

Therefore:

$$\phi(t) = \int 2\pi\{ f_0 + f_d \cos 2\pi f_m t \} dt$$

$$= 2\pi f_0 t + \frac{f_d}{f_m} \sin 2\pi f_m t + \phi_0$$

$$= \omega_0 t + mf \sin \omega_m t + \phi_0 \tag{5}$$

$\omega_o = 2\pi f_o$, $\omega_m = 2\pi f_m$, mf: Modulation Index = $\frac{f_d}{f_m}$, ϕ_o: Electrical angle when $t = 0$.

As equation (5) is simple, we will delete ϕ_o and substitute in equation (1).

Then:
$$i = I_o \sin\{\omega_o t + mf \sin \omega_m t\} \tag{6}$$

If the above equations are expanded by using Bessel's Functions, they may be expressed as follows.

$$\begin{aligned} i = I_o[&J_o(mf) \sin \omega_o t \\ &+ J_1(mf) \{\sin(\omega_o + \omega_m)t - \sin(\omega_o - \omega_m)t\} \\ &+ J_2(mf) \{\sin(\omega_o + 2\omega_m)t + \sin(\omega_o - 2\omega_m)t\} \\ &+ J_3(mf) \{\sin(\omega_o + 3\omega_m)t - \sin(\omega_o - 3\omega_m)t\} \\ &+ \ldots\ldots\ldots\ldots] \end{aligned}$$

Here, $J_n(mf)$ is the "n'th Bessel Function of the First Order corresponding to argument mf and possesses the figures shown in Figure 1.

If we refer to Figure 1 and equation (7), it will be noted that the carrier frequency component of frequency modulation is not constant but varies with the modulation index and, at times, it will even disappear. A pair of upper and lower side-bands will be formed as in AM modulation and, even when modulated with a single frequency, a numerous of side-bands spaced equal to the modulation frequency will be generated as shown in Figure 2.

In this manner, the frequency spectrum the oretically has an infinite width, but when the modulation index is smaller than 0.5, the side-bands beyond the second ones will be so small that they may be disregarded as shown in Figure 1. This means, that the band-width may be considered to be practically the same as that of AM modulation.

If an FM side-band direct-viewing device is used, the side-band

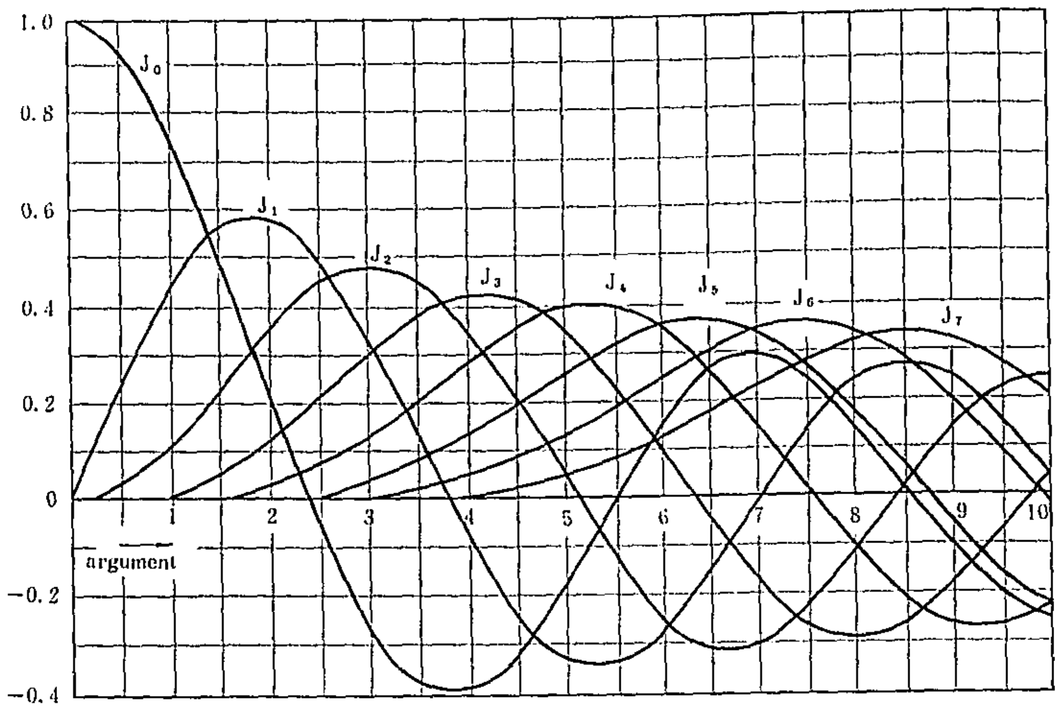


Figure 1. Besel Functions

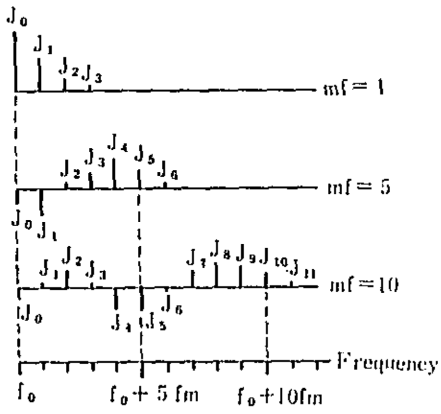


Figure 2. FM Upper Side Band Spectrum

configuration may be viewed directly on the oscilloscope. If modulation is then carried out with a constant frequency and, by varying the degree of modulation to obtain the extinction point of carrier-wave or the side-bands, the modulation index can be obtained by the following table.

Order of which Amplitude drops to Zero	J_0	J_1	J_2	J_3	J_4	J_5
First	2,405	3,832	5,135	6,379	7,586	8,780
Second	5,520	7,016	8,417	9,790	11,064	12,339
Third	8,654	10,173	11,620	13,017	14,373	15,700
Fourth	11,792	13,323	14,796	16,224	17,616	18,982
Fifth	14,931	16,470	17,790	19,410	20,827	22,220

Note 1. If the designated maximum frequency deviation of a certain transmitter is Δf, degree of modulation will be

$$kf = \frac{f_d}{\Delta f}$$

The modulation index will therefore be

$$mf = \frac{\Delta f}{f_m} kf$$

That is, the modulation index is in relation to the degree of modulation and in inverse relation to the modulation frequency. In some books, $\Delta f/f$ is taken as the modulation index.

Note 2. If the carrier-wave and side-waves take a negative value in Fugures 1 and 2, this would indicate that the phase will be reversal. But as a side-band direct-viewer shows only the size, the entire spectrum is in the upward position. Further, if we consider only the size, the side-band in equation (7) will be symmetrical in upward and downward, position in relation to the center frequency.

Next, if the electrical angle $\phi(t)$ in phase-modulation is expressed as

$$\phi(t) = \omega_o t + m_p \sin \omega_m t$$

and then substituted in equation (1), we will obtain the following equation.

$$i = I_o \sin \{\omega_o t + m_p \sin \omega_m t\}$$

Here, m_p = Modulation-index of phase-modulation = Phase shift (radian)

This simply indicates that the modulation index mf is substituted for m_p when compared with equation (6), and therefore, its arithmatic expression is exactly the same as in frequency modulation.

$$mf = \frac{f_d}{f_m} = m_p \qquad \text{Therefore: } f_d = f_m \ m_p$$

Namely, a phase modulation frequency of phase deviation m_p radian is equivalent to a frequency modulation wave of frequency deviation ($f_m \times m_p$) C/S. In other words, phase modulation is equivalent to a type of frequency modulation of which the frequency deviation is in direct relation to the modulating frequency. Therefore, by adding a network of which the output is in reverse relation with the frequency, to a modulated signal circuit, and provide phase modulation to it, a genuine frequency modulation will be obtained.

Let us next study about the frequency modulation wave expressed in equation (7) by means of vectors. When the modulation index is small, cut off the side-bands above the secondary order and, draw a vector only of the carrier and primary side-band, then it will become as Fig. 3(c). This vector resembles the vector (b) of amplitude modulation waves, but as in case of frequency modulation, the phase of the lower side-band L and upper side-band U differs 180 degrees, the composition of the upper and lower side-band wave will always become a vector of which the amplitude always varies at right angles to the carrier.

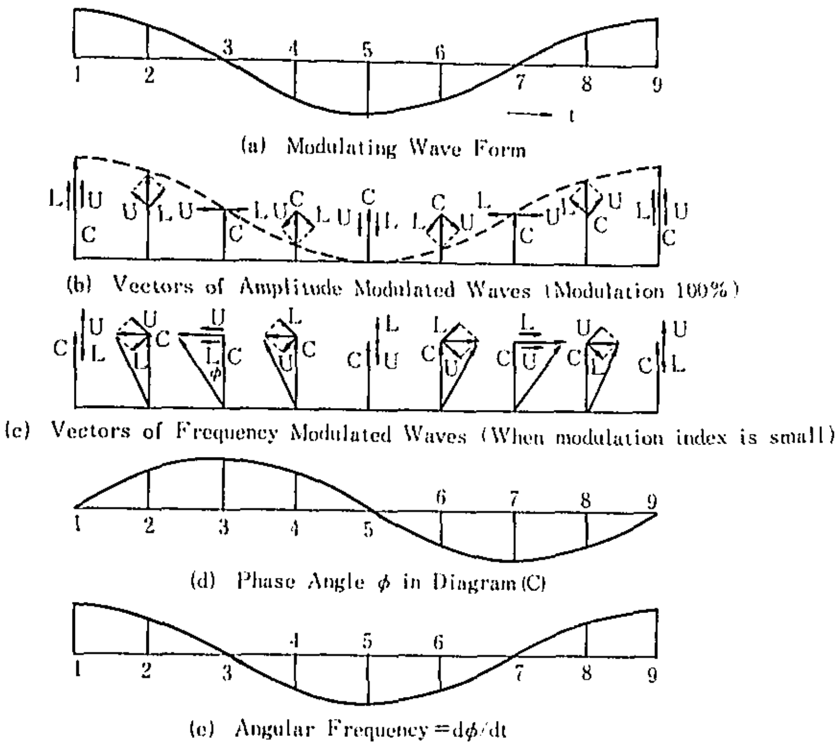


Figure 3. Vectors of Modulated Waves

C: Carrier Wave L: Lower Side-Band U: Upper Side-Band

The composition of the right angle vector and carrier wave C will shift ϕ phase from the carrier wave under no modulation. That is, it will accept phase modulation as Fig. (d) as well as frequency modulation of $\frac{d\phi}{dt}$ as shown in Fig. 3 (e). It should be noted that there is a 90 degree phase difference between phase modulation and frequency modulation, as far as the modulation signal is concerned.

1.3 Band Width Required for Frequency Modulation Waves

It is said that the side-bands in frequency modulation spread out infinitely, but, for practical purposes, it may be necessary to consider only the bands that include the bulk of the energy. The ratio of the side-band energy against the total energy, up to the N'th wave, which is modulated with a single sine-wave, is expressed as follows.

$$A = \frac{\sum_{-N}^{N} J_n^2\ (mf)}{\sum_{-\infty}^{\infty} J_n^2\ (mf)} \tag{10}$$

However, as it will become $\sum_{-\infty}^{\infty} J_n^2\ (mf) = 1$ with the Bessel Function equation, it will be indicated as follows.

$$A = \sum_{-N}^{N} J_n^2\ (mf)$$

According to Fusachika Miyata's calculations, if a figure equal to mf is used for N, A would be 0.95. In other words, if a band-width of $2f_d$ is taken, 95% of the energy will be transmitted.

The foregoing is a case of modulation with single sine-waves, but if modulated with square waves, the transmission of energy of this band will drop down to 80%. However, if modulated with square-waves, and side-bands up to $(mf + 1)$ are accepted, 93% of the energy will be included. Inclusion of side-bands up to $(mf + 1)$ order will be equal to use a band-width of $2(f_d + f_m)$ and this is considered to be the practical band-width.

As obvious from the Vessel Function curves in Figure 1, side-bands up to $(mf + 1)$ order are less than approximately 0.13 and up to $(mf + 2)$ order, are less than 0.05, it may be surmised that side-bands up to $(mf + 1)$ order will be sufficient t.

(Calculation Examples)

Maximum Frequency Deviation Δf = 75kHz
Highest Modulation Frequency f_m = 15kHz
Required Band Width B = 2(75 + 15)kHz = 180 kHz

Further, if we consider the primary side-bands only when degree of modulation is low, it will become necessary to secure a minimum band-width of $2f_m$ regardless of the degree of modulation. This will be of extreme importance when carrying out frequency multiplication of modulated frequencies.

In frequency modulation, the modulation index mf is in inverse relation to the modulation frequency. Therefore, as the frequency of the modulation frequency increases, the mf decreases and automatically restrains the expansion of side-bands and, if the maximum frequency deviation Δf is a of a constant value, the band-width will almost be constant.

In contrary to this, the disadvantage of this phase modulation is that as the frequency deviation is in direct relation to the modulation frequency, and the modulation frequency becomes higher, the energy tends to be transferred to the higher side-bands and the band-width will be expanding.

1.4 Improvement of Interference and Noise in Frequency Modulation

If an interference signal B of a frequency near a desired signal A is interfering, the end of composite signal C will travel along the circular path formed by beat angular frequency P_n. Since an amplitude modulation receiver senses amplitude variation C at this time, the interference ratio in amplitude modulation may be expressed as B/A, providing B is greater than A.

On one hand, in frequency modulation receivers, the amplitude limiter functions to limit amplitude variation C and thus senses frequency variations only. Frequency variation C is a differentiation of phase variations and will be as follows:

$$\omega_n = \frac{d\phi}{d_t}$$

Providing B is greater than A, phase will be

$$\phi \doteqdot \tan\phi = \frac{B}{A} \sin P_n t$$

As a result, frequency modulation from interference will be as follows:

$$\omega_n = \frac{d\phi}{d_t} = \frac{B}{A} P_n \cos P_n t$$

If we consider the size of the variations only, it will be

$$| \omega_n | = \frac{B}{A} P_n$$

However, as A itself is modulated by an optional modulation frequency with a frequency deviation of $\omega_d/2$, the interference ratio in frequency modulation will be as follows:

$$\left| \frac{\omega_n}{\omega_d} \right| = \frac{B}{A} \cdot \frac{P_n}{\omega_d} \qquad (11)$$

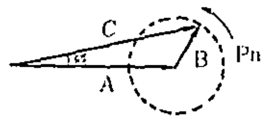


Figure 4. Interference

The improvement ratio will therefore be as follows.

$$\frac{\text{Interference}}{\text{Improvement Ratio}} = \frac{\text{Interference Ratio of Frequency Modulation}}{\text{Interference Ratio of Amplitude Modulation}}$$

$$= \frac{\frac{B}{A} \cdot \frac{P_n}{\omega_d}}{\frac{B}{A}} = \frac{P_n}{\omega_d} \backsim \frac{P_n}{mf} \qquad (12)$$

mf = modulation index

If we now consider the maximum audible beat frequency as being 15 kHz and the maximum deviation at 100% modulation to be 75 kHz at $\omega_d/2\pi$, interference will be reduced to 1/5 according to this equation.

We will next explain the problem of noise. Although noise is distributed in a certain frequency band, if we consider only one noise frequency, it will be the same as the previous interference. One need merely integrate the results for the bands passed.

Although there are 2 types of noise - impulsive noise and fluctuation noise - in the case of impulsive noise, it will only be necessary to integrate on the assumption that all of the noise elements will be of equal phase at a certain time $t = t_o$. Also, as noise actually becomes a problem within the audible frequency range at the output of the receiver, we need consider only that portion below maximum value P_{max} of the audible frequency. If we now carry out integration of noise within this range, the amplitude modulation noise area will be $(\frac{B}{A} \times P_{max})$ as shown in Figure 5, since B/A is unrelated to P_n in the case of amplitude modulation.

Next, in the case of frequency modulation, as may be discerned from the previous interference ratio equation, since it will be B/A when $P_n - \omega_d$ and will be in direct ratio with P_n, it will possess characteristics of the triangular noise spectrum as illustrated by line b in the same Figure. As in the case of amplitude modulation, it will also only be necessary to consider integration up to the maximum audible frequency value P_{max}.

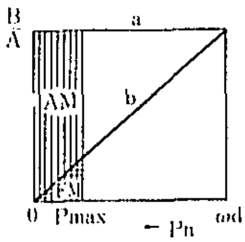


Figure 5. AM and FM Noise Characteristics

The noise area in frequency modulation will then be

$$\left(\frac{B}{A} \times P_{max}\right) \times \frac{P_{max}}{\omega_d} \times 1/2$$

The noise voltage ratio in the case of impulsive noise may be expressed as follows.

$$\text{Voltage Ratio of Impulsive Noise} = \frac{\text{Frequency Modulation Noise}}{\text{Amplitude Modulation Noise}}$$

$$= 1/2 \ \frac{P_{max}}{\omega_d} \backsim \frac{1}{2m_{f_0}} \qquad (13)$$

Here, m_{f_0} = deviation ratio

$$= \frac{\text{Maximum Frequency Deviation}}{\text{Maximum Modulation Frequency}}$$

In fluctuation noises, all of the components are not of equal phase but are calculated in the form of $\sqrt{\int e^2 dP_n}$. In other words, in amplitude modulation, it will be as follows:

$$NA = \sqrt{\int_o^{P_{max}} \left(\frac{B}{A}\right)^2 dP_n}$$

$$= \frac{B}{A}\sqrt{\left|P_n\right|_o^{P_{max}}}$$

$$= \frac{B}{A}\sqrt{P_{max}} \qquad (14)$$

In frequency modulation it will be as follows:

$$NF = \sqrt{\int_o^{P_{max}} \left(\frac{B}{A} \cdot \frac{P_n}{\omega_d}\right)^2 dP_n}$$

$$= \frac{B}{A} \ \frac{1}{\omega_d}\sqrt{\int_o^{P_{max}} P_n^2 \ dP_n}$$

$$= \frac{B}{A} \cdot \frac{1}{\omega_d} \sqrt{\frac{1}{3}\left| P_n^3 \right|_o^{P_{max}}}$$

$$= \frac{1}{\sqrt{3}} \cdot \frac{B}{A} \cdot \frac{\sqrt{P_{max}^3}}{\omega_d} \qquad (15)$$

Therefore, the voltage ratio of fluctuation noise will be

$$\frac{NF}{NA} = \frac{1}{\sqrt{3}} \cdot \frac{P_{max}}{\omega_d}$$

$$= \frac{1}{\sqrt{3}} \cdot \frac{1}{mf_0} \qquad (16)$$

As these forms of noise are typical cases of both extremes, in the case of general noise, the improvements in S/N ratio of frequency modulation over that of amplitude modulation will be the intermediate voltage ratio value of $1/(\sqrt{3} \sim 2)\ mf_0$. The ratio between the S/N ratio (FM) in frequency modulation and the S/N ratio (AM) in amplitude modulation is called the "frequency modulation wide-band S/N gain" and the noise considered here is fluctuation noise.

With degree of modulation as 1 in amplitude modulation and taking the maximum frequency deviation $\Delta\omega/2\pi$ of one half the intermediate frequency band-width B in frequency modulation and, moreover, with degree of modulation kf at maximum of 1, the signal ratio will be

$$\frac{\text{Frequency modulation signal}}{\text{Amplitude modulation signal}} = 1$$

The wide-band S/N gain of frequency modulation will therefore be expressed as follows.

$$\frac{(S/N)_{FM}}{(S/N)_{AM}} = \sqrt{3}\frac{\Delta\omega}{P_{max}} = \frac{\sqrt{3}}{2} \cdot \frac{B}{P_{max}} = \sqrt{3}\ m_{fo} \qquad (17)$$

Due to the above relations, S/N gain will increase in the frequency modulation system.

(Calculation Examples)

Maximum Frequency Deviation Δf = 75kHz

Maximum Modulation Frequency f_m = 15kHz

Deviation Ratio $m_{f_0} = \frac{75}{15} = 5$

Wide Band S/N Gain $= 5\sqrt{3} = 8.67$

$= 18.8\text{dB}$

1.5 Limit of S/N Improvement in Frequency Modulation

Frequency modulation is affected less by interference and noise than amplitude modulation, this is true only when A is greater than B. If B is equal to A or when A becomes greater than B, the phase variation of composite waves C or C' will generate extremely large frequency modulation, as shown in Figure 6, and noise output will increase.

As a result, in case of frequency modulation, when the interferring or noise signal B is considerably smaller than the desired signal A, it would be effective. It B approaches A, the effect will decrease, and when A = B, the degree of improvement will be reverse. This point is called the S/N improvement threshold. When the peak value of the signal voltage A and noise voltage B are equal, this is said to be in relation to the detector input of receiver sets, and there is a slight difference in regard to input of antenna, namely the receiver.

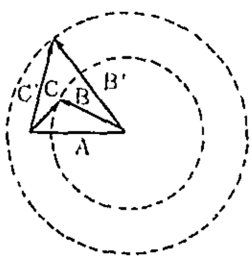


Figure 6. When interference is strong

Speaking about noise, as in frequency modulation receivers, if it has a wide $2\Delta\omega$ ($\Delta\omega$ = max. angular frequency deviation) band-width, a greater number of noise spectrums will enter the detector than in case of $2 P_{max}$ (P_{max} = angular modulation frequency) of a amplitude modulation receiver.

As the band-width is wide in a frequency modulation receiver, the peak value of noise at the detector input is greater than that in amplitude modulation. In the case of fluctuation noises, the extent of this difference is in proportion to the square root of the band-width and in the case of impulse noises, it is in direct ratio with the band-width. Nevertheless, as the band-width in frequency modulation is $2\Delta\omega$ and the band-width in amplitude modulation is $2 P_{max}$, the peak value of fluctuation noise in frequency modulation will be $\sqrt{\Delta\omega/P_{max}} = \sqrt{mf_0}$ times that in amplitude modulation, and $\Delta\omega/P_{max} = mf_0$ times for impulse noise.

Therefore, when the noise portion of the S/N ratio at the input of the receiver is multiplied by this multiplying factor, A = B will be the point where the product is 1 and this will be the S/N improvement threshold.

In the case of fluctuation noises, therefore, the improvement threshold will be the point where the S/N ratio at the receiver input is $\sqrt{mf_0}$, and where the S/N ratio at the receiver input is mf_0 for impulse noises. That is, the larger the deviation ratio mf_0, the greater must be the signal (carrier wave) at the S/N improvement threshold. As may be discerned from the foregoing relations, when $mf_0 = 1$, since there are no corrections in the noise improvement threshold, S/N is continually being improved to a greater extent in frequency modulation than in amplitude modulation as shown in the same diagram. The degree of improvement is $\sqrt{3}\,mf_0$ in the case of fluctuation noises and $2\,mf_0$ in the case of impulse noises, with mf_0 equalling 1. The improvement is very slight with $\sqrt{3}X = 5$ dB and $2X = 6$ dB respectively.

When mf_o increases and becomes $mf_o = 4$, as shown in the same diagram, the wide band S/N gain will increase to $\sqrt{3}\, mf_o = \sqrt{3} \times 4$ times = 17 dB, or $2mf_o = 2 \times 4$ times = 18dB when the S/N ratio at the receiver input is large. However, as S/N ratio $\sqrt{mf_o}$ (for fluctuation noise) at the receiver input is the S/N improvement threshold, as is mf_o for impulse noises, output S/N will become poorer than in amplitude modulation if the input S/N drops below this threshold. That is, when $mf_o = 4$, the input S/N is $\sqrt{mf_o} = 2$ = 6dB in relation to fluctuation noises and over $mf_o = 4$ = 12dB for impulse noises and, as may be discerned from the diagram, the output S/N is improved rather than the input S/N. The following facts are revealed from these results. When sufficiently high output S/N is desired such as in broadcasting, it will be desirable to have maximum deviation factor providing the input S/N is above the improvement threshold.

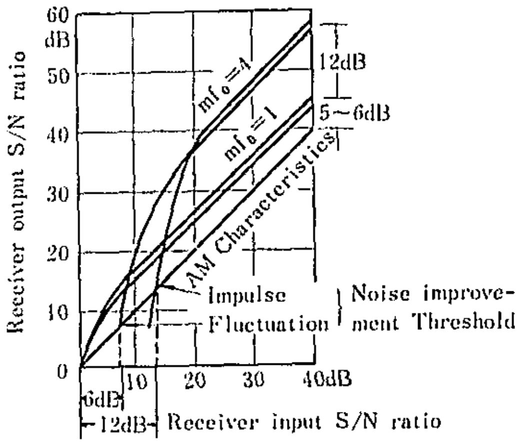


Figure 7. S/N Improvement Threshold

However, for portable transmission etc. where maximum dis-range is desirable even at the expense of some quality, output, and S/N, it would be desirable that the deviation factor be near the value $mf_o = 1$.

1.6 Pre-emphasis and De-emphasis

Although S/N is improved by wide-band S/N gain $\sqrt{3}\ m_{fo}$ in FM, as may be seen from the noise spectrum in the input in Figure 5, noise increases in relation to the frequency and the S/N ratio becomes poor in the high frequency ranges.

Transmission by first emphasizing modulation in the high bands to prevent this is called pre-emphasis and the function in the receiver to de-emphasize the high-band with completely opposite characteristics is called de-emphasis.

The overall transmission and receiving signals are given flat characteristics by this operation, but as noise decreases equal to the amount of de-emphasis, the overall S/N ration is improved.

Further, as energy distribution is low in the high frequency component of the signal, there will be no fear of over-modulation when pre-emphasis is applied.

Pre-emphasis frequency characteristics may be expressed as follows.

$$\sqrt{1 + (\omega\tau)^2} \tag{18}$$

Here, τ: Time constant of pre-emphasis

The time constant of pre-emphasis in FM broadcasts is designated as 50 μs and its characteristics are shown in Figure 26. Straight line "b" in Figure 8 indicates noise distribution in the output of the FM detector and its size is $\frac{B}{A} \cdot \frac{P_n}{\omega_d}$ as previously explained. Curve "c" indicates de-emphasized noise distribution and its size is

$$\frac{B}{A} \cdot \frac{P_n}{\omega_d} \cdot \frac{1}{\sqrt{1 + (\tau P_n)^2}} \tag{19}$$

If we calculate $\sqrt{\int e^2\, dP_n}$ of noise, we have

$$\text{NF (de-emphasis)} = \frac{B}{A} \cdot \frac{1}{\omega_d} \sqrt{\int_0^{P_{max}} \frac{P_n{}^2}{1 + (\tau P_n)^2}\, dP_n}$$

$$= \frac{B}{A} \cdot \frac{1}{\omega_d \sqrt{\tau^3}} \sqrt{\tau P_{max} - \tan^{-1} \tau P_{max}} \qquad (20)$$

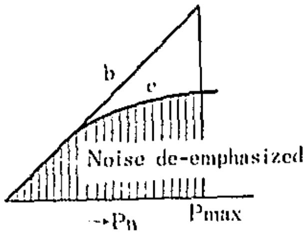


Figure 8. S/N Improvement by De-emphasis

If we compare this to noise output equation (15) without de-emphasis,

$$\text{Noise improvement by de-emphasis} = \frac{N_F \text{ (without de-emphasis)}}{N_F \text{ (with de-emphasis)}}$$

$$= \frac{\tau P_{max}}{\sqrt{3}} \cdot \sqrt{\frac{\tau P_{max}}{\tau P_{max} - \tan^{-1} \tau P_{max}}}$$

$$= 3.22 \text{ times } \left[\begin{array}{l} \cdot \tau = 50 \times 10^{-6} s \\ P_{max} = 2\pi \times 15 \times 10^{3} \end{array} \right]$$

$$= 10.2 \text{ dB.}$$

In other words, the S/N ratio is improved by 29 dB with the addition of wide-band S/N improvement of 18.8 dB.

1.7 Distortion in Frequency Modulation

Although distortion occurs in amplitude modulation systems due to non-linearity of vacuum tubes etc., in frequency modulation, the signal is not transmitted by means of amplitude and non-linearity in the circuit is eliminated by maintaining the amplitude at a fixed value by means of a limiter. However, distortion will occur when phase characteristics are not linear even if the transmission circuit is linear and, this type of distortion is called linear distortion. When the relation between angular frequency ω and the phase angle of the transmission line is shown as in figure 9, the first differential of the phase characteristics $\tau = \frac{d\phi}{d\omega}$ indicates the time-lag of the wave-form in the frequency element in this vicinity. Therefore, if the phase characteristics are linear within this band as shown in Figure 9(a), the time-lag will be constant as in Figure 9(b).

This means that, as all of the side bands within this band are equally delayed, the entire composite wave form is delayed and the wave form shows that distortion will not be generated by itself.

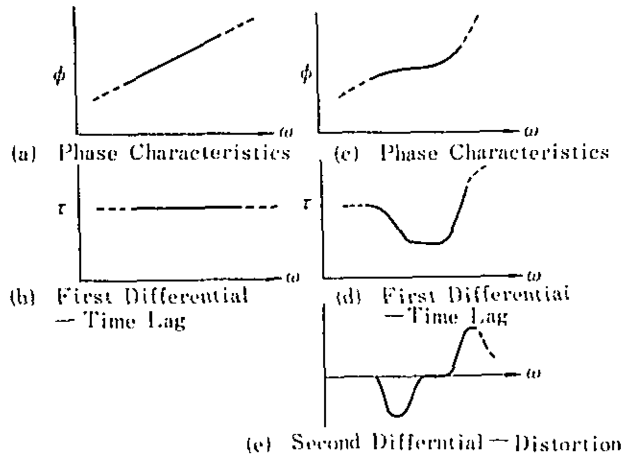


Figure 9. Linear Distortion

Next, as differences in time lag will exist due to the frequency when the phase characteristics are bent within the band as in Figure 9(c), some of the side-bands arrive early while others arrive late. It may therefore be assumed that signals obtained by composing and demodulating these frequencies will be different from the original. The second differential values of the phase characteristics as shown in Figure 9(e) will be the measure expressing the distortion. The amplitude characteristics and phase characteristics of electrical circuits are closely related as may be evidenced by the fact that, when the inclination of the amplitude characteristics is positive, phase is delayed and, when negative, phase is advanced. (Refer study material Radio Engineering (A) Volume 6 "Radio Broadcasting Transmitters" Section 4.5). It will therefore be desirable to make the amplitude characteristics within the band as flat as possible to maintain good phase characteristics.

As non-linear distortion tends to occur near the edges of the band-pass range of tuned circuits in FM, it will be necessary to have sufficiently wide band width as distortion will be compounded when passing through many tuned circuits. Also, if the characteristics of the discriminator in the receiver does not have a sufficiently wide linear range, linear distortion will tend to occur. Special importance is attached to linear distortion in multiplex communication as this will be the cause of crosstalk.

1.8 Multiplication and Frequency Conversion in Frequency Modulated Waves

(1) Expressing frequency modulated waves up to their peak phase as

$$i = I_0 \sin(\omega_0 t + m_f \sin\omega_m t + \phi_0) \qquad (21)$$

if this is multiplied "m" times, it will be

$$i' = I'_0 \sin m(\omega_0 t + m_f \sin\omega_m t + \phi_0) \qquad (22)$$

$$= I'_0 \sin m\omega_0 t + m.m_f \sin\omega_m t + m\phi_0)$$

This equation reveals that, when multiplied "m" times, not only is the frequency multiplied "m" times.

(2) If we mix an oscillation frequency of $I_1 \cos \omega_1 t$ to the frequency modulated wave obtained in equation (21) and frequency conversion carried out, it may be expressed as follows.

$$i'' = I_0 \sin(\omega_0 t + mf \sin\omega_m t + \phi_0) \cdot I_1 \cos\omega_1 t$$

$$= \frac{I_0 I_1}{2} [\sin \{(\omega_0 + \omega_1)t + mf \sin \omega_m t + \phi_0\}$$

$$+ \sin \{(\omega_0 - \omega_1)t + mf \sin \omega_m t + \phi_0\}] \qquad (23)$$

That is, when frequency conversion is carried out, 2 frequency modulated carrier waves are generated - one being the sum of the frequency modulated wave and the added frequency and the other the difference between these two. The modulation index and phase angle however, remain unchanged.

The foregoing phenomenon is frequently applied to actual transmitters and receivers.

2. STEREOPHONIC BROADCASTS

2.1 The History of Stereophonic Broadcasts

When one listens to an orchestra in a music hall, one senses expansion of the sound source, spatial separation and harmony in addition to the high and low, strong and weak and tone quality of the sound. In other words, one has the sense of presence. However, even with todays high level transmission reproduction technique one cannot sense expansion in sound source, spatial separation etc. as all sound emerges from speakers in monophonic systems. One therefore does not have the sense of presence as when at a music hall.

Stereo sound was therefore developed for further high level transmission of sound in an effort to break this monophonic barrier.

This however, was not a recent discovery but has a fairly old history.

It is said that stereo was first developed when Edison invented the gramophone in 1881 and carried out a public test at the Paris Opera House with a 2-circuit stereo reproducer using a right and left receiver head-set. Although no special advancement in stereo technique was noted for the next 50 years, A.D. Blumenlein carried out researches in 1929 on a 2-circuit stereo sound system employing 2 microphones and 2 speakers; and in 1934, announced the method of recording left and right stereo signals in a single groove of the record. Full scale research on stereo sound commenced from this year with Bell Company of America also conducting experiments on 3-circuit stereo systems.

Physiological and psychological research on binaural hearing became very popular from about this time.

In the broadcasting world on one hand, interest heightened on ever better sound quality with the popularization of radio broadcasts. E.H. Armstrong, the advocate of FM broadcasting, conducted experiments in stereo broadcasting in 1938 using an FM multiplex modulation system with left and right signals. However, it was not until after the second world war that full scale movement was made in this respect.

After the war, the practical application of stereo sound grew sharply with the growth of tape-recording and the advancement of record pressing technique. Stimulated by this situation, the broadcasting world evoked the response of the public with France carrying out experimental broadcasting of medium wave 2-wave systems in 1950 and America and Japan following in 1952. In December of 1952, Japan Broadcasting Corporation commenced experimental broadcasts on its No. 1 and No. 2 medium wave network and started regular broadcasts from November of 1954.

With advancements in stereo transmission technique, the true value of FM hi-fidelity broadcasts was subsequently widely recognized and research was promoted on compatible single-wave FM stereo broadcasting systems. In 1953 M.G. Crosby introduced the sum and difference system and in 1957, stereo recording in single grooves was standardized. The tendency in the broadcasting world was towards standardizing stereo broadcasting systems and in 1959, CCIR decided on a compatible stereo broadcasting system as their research problem. (Refer supplement 1.)

From 1958, the FCC conducted surveys of stereo broadcasting in America and solicited data from the various persons concerned. In response to this the Electronics Industrial Association formed the National Stereophonic Radio Committee, NSRC, which carried out experiments on various systems and reported the results to the FCC. Based on the data submitted, the FCC established a stereo broadcasting system in April of 1961 and the world's first FM stereo broadcasting station was born the following year. (Refer Supplement II(3)).

On one hand, research was commenced on the single wave stereo broadcasting system in Japan, at about the time that the Japan Broadcasting Corporation commenced experimental FM broadcasts in 1957. Subsequently, in 1961 the Radio Technical Deliberation Council received a request from the Minister of Posts and Communications to designate a standard stereo broadcasting system and technical standards. In response to this request, various experimental studies were carried out with the cooperation of the organizations concerned and in June of 1963 the results were consolidated and a report submitted on a standard stereophonic broadcasting system. Based on this report necessary laws and regulations were enacted and amended in 1968. It was also in 1968 that a policy was decided on in Japan to reorganize sound broadcasting and, regular FM broadcasting commenced at that time.

2.2 Fundamentals of Stereophonic Sound

2.2.1 Feature of Stereophonic Sound

It is said that the purpose of stereophonic sound is in realizing the sense of presence that could not be satisfactorily realized with the conventional monophonic reproductions.

According to research results on the elements from which this sense of presence could be realized, it was determined that this could be divided into the effect of directivity and the sense of reverberation. It is said that, if suitable amounts of these two elements are included, the sense of presence may be realized.

To analyze the effects of stereo phonic sound systems it will therefore be necessary to first know-how the directivity of the sound and its reverberation are being reproduced. Physically, the basis of the entire system is in first knowing how the sound from a sound source of a certain direction is being reproduced.

2.2.2 Position and Sense of Sound

If we consider the function of orientation of those of us who have normally experienced sound in natural surroundings, the sound source position is determined by first determining the horizontal and vertical direction from which the sound is heard and also the distance.

Orientation	Corresponding physical volume
Distance	Direct sound strength, curvature of wave front Ratio between direct and dispersed sound, difference in tone quality
Direction	Direction of advancement of direct sound wave front

(a) Sensation of distance

Although the sensation of distance to the sound source is not very accurate, it is considered fairly accurate if within approximately 1 meter. Although the sound source is actually at a distance in most cases, in these cases it is believed that a good part of the sound heard is reflected sound and that distance is judged by comparing the strength and sound quality with those memorized from past experience.

(b) Sensation of direction in the vertical plane

Considering the position of the ears, it may be easily imagined that the sense of direction in the vertical plane is poorer than that in the horizontal plane. However, our ability to actually differentiate these to a certain extent is believed to be due to the fact that an overall judgement is being made based on minute changes in sound quality resulting from diffraction effects of the body and from the reflections from the ground or floor.

(c) Sensation of direction in the horizontal plane

In relation to sensation of direction in the horizontal plane, considerably detailed measurements have been made and research is also being advanced. Although the physical mass carrying information as to the direction of the sound from a certain sound source is in the direction of the advancing wave front, the mechanics by which we receive these may be considered the difference in time and strength at which these wave fronts reach our ears.

However, it has still not been determined as to what the relation is between these two methods and the sense of direction of advancement of the wave front. Calculations and measurements have, however, been made on the relation between the direction and difference in arrival time from the sound source, and the difference in sound pressure between the two ears.

(i) Time difference in arrival

As shown in Figure 2.2.1, if the portion above the neck, that is the head, is considered to be a cylinder with a radius of "a" cm and with its center as "0"; and if we consider the ears to be located at the ends of the diameter, calculations may be made of the time difference in arrival to both ears of horizontal plane waves from a direction $\alpha°$ from the front of the face.

If "c" is the velocity of sound, the time difference of the waves reaching both ears will be as follows.

$$\Delta t = a \left(\frac{\pi}{180} \alpha + \sin\alpha \right) / c$$

If calculation is made with the diameter of the head as 21cm, it will be as shown in Figure 2.2.2.

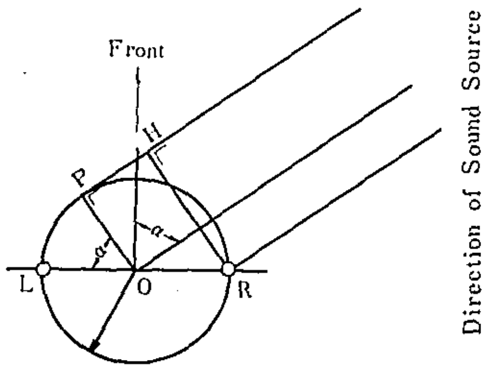


Figure 2.2.1

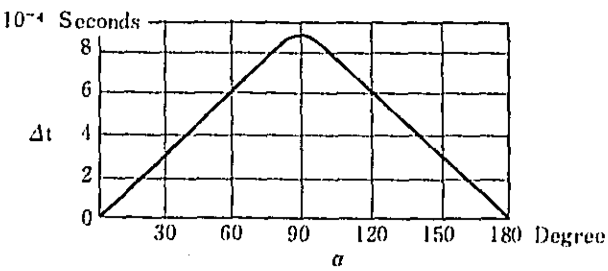


Figure 2.2.2
Relation between Δt and α

(ii) Difference in strength

Due to diffraction around the head, the sound pressure at both ears carry different values according to the direction of arrival of the sound waves. Since this sound pressure difference is due to diffraction, it will differ according to the sound frequency. In the low sound ranges where the wave lengths are long compared to the diameter of the head, there is practically no difference in sound pressure between the ears, for the sound arriving from any direction whereas a difference of approximately 10 dB may be perceptted in relation to the short wave lengths in the high sound range.

Fig. 2.2.3 shows the relations between the direction of speaker and difference in average sound pressure level at both ears, when speakers, voice arrives from various directions in the horizontal plane.

Figure 2.2.4 shows the relations between the direction of arrival of pure sound of various frequencies and difference in sound pressure level at both ears.

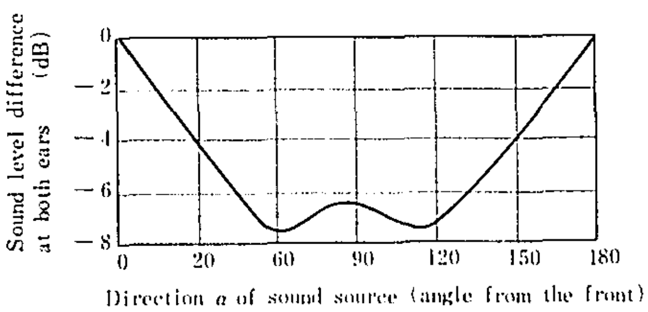


Figure 2.2.3 Direction of the speaker and difference of sound strength level at both ears

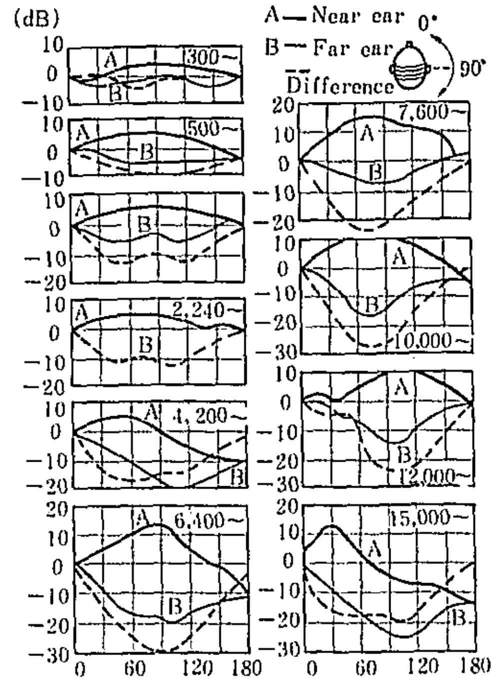


Figure 2.2.4 Direction of pure sound-source within horizontal plane (degrees) (Silvian and white)

(iii) Accuracy of direction orientation

Much research has been made in relation to the accuracy of direction orientation within the horizontal plane. Of these, according to Mill's research, the difference limen of direction orientation within the horizontal plane is related to the direction of the sound source and the sound frequency; and if the sound source is at the front and the lowest sound emitted is 500 to 700Hz, an approximate 1° change may be sensed.

When the sound source shifted from the front to the side and approached a position directly to the side, the difference limen increased sharply and measurements were not possible at this position.

For a certain fixed direction of sound source, the difference limen of direction will be minimum at 250 ∿ 1,000 Hz, and, increase gradually and, between 1,000 ∿ 1,500 Hz, it will increase sharply to the maximum value. It will decrease again to the minimum value between 3,000∿6,000 Hz and, increase to the second maximum value at about 8,000 Hz.

Judgement of the direction will be very inaccurate and above 3,000 Hz a slight concha effect will appear.

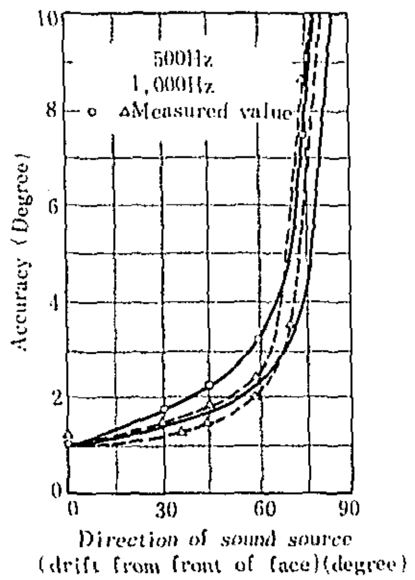


Figure 2.2.5(a) Direction of sound source (drift from front of face) (Degree)

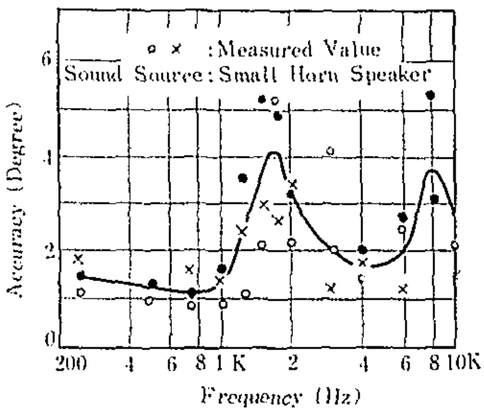


Figure 2.2.5(b) Frequency (Hz)

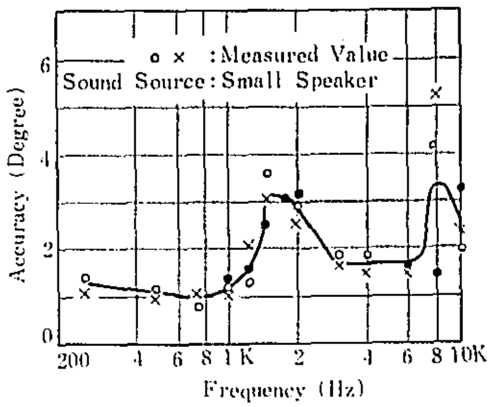


Figure 2.2.5(c) Accuracy of directional orientation within a horizontal plane (Mills)

2.2.3 2-Circuit Stereophonic Sound Field (1)

(1) Features of a 2-circuit stereophonic sound field

Let us make a rough comparison between a natural sound field and a man-made stereophonic sound field. Figure 2.2.6 shows a natural sound field in which the sound source is advancing towards the listener.

The first portion of the sound reaches the right ear first and then the left ear. It is in transient condition during this interval followed by sound waves passing in successive order.

On one hand, if microphones were placed at the position of both ears and their output reproduced by left and right speakers respectively, it may be considered that the results would be as shown in Figure 2.2.7.

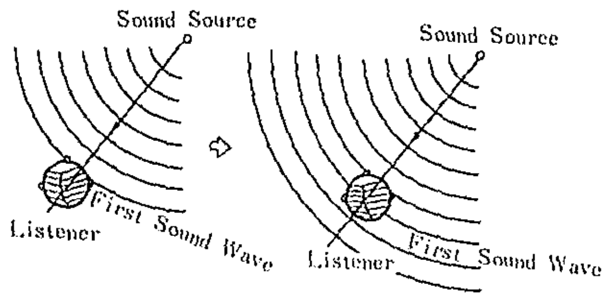


Figure 2.2.6

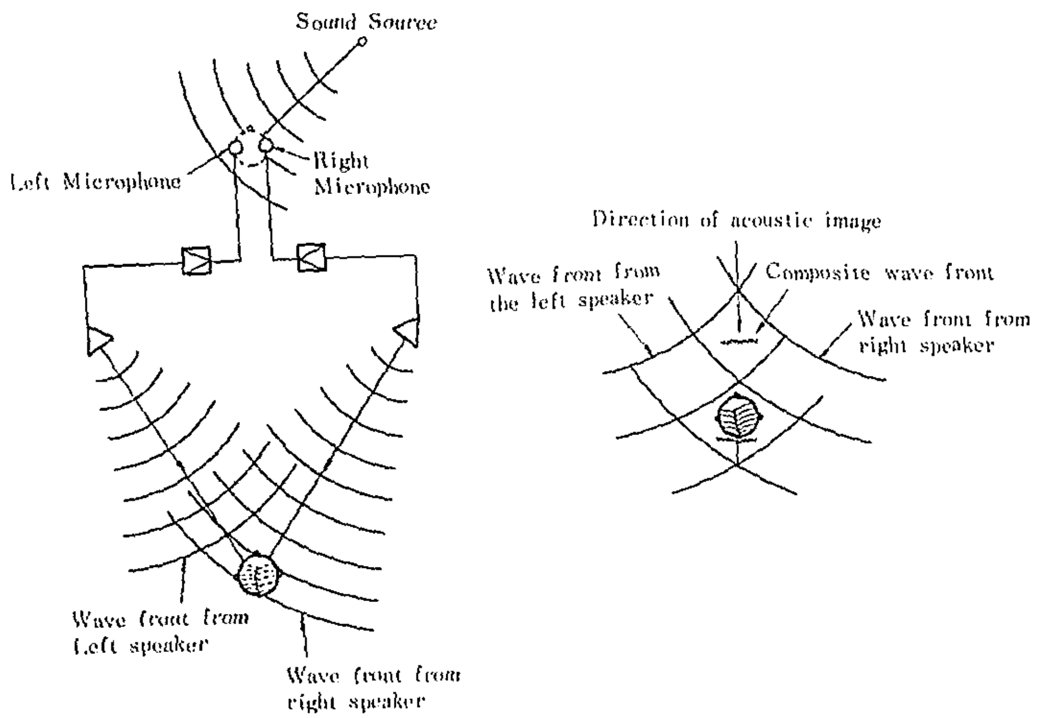


Figure 2.2.7

As the "actual sound source" is the left and right speakers, the sound field is created by the 2 progressive sound waves from the speakers as illustrated in the diagram.

Although it will be necessary to pay close attention to the fact that the transient conditions near the sound build-up-time in stereophonic sound fields is greatly different from that in the sound fields we normally experience. At present, it is not clear as to the influence this transient condition has on the sensation of direction. However, as with regard to the quasi-stationary condition of the bulk of the sound source, excluding the sound build-up portion, it will only be necessary to consider the already overlapped condition of the waves from the speakers. Therefore, the nature of the stereo sound field may be clarified by studying the progressive condition of the composite wave front while taking into consideration the foregoing transient condition.

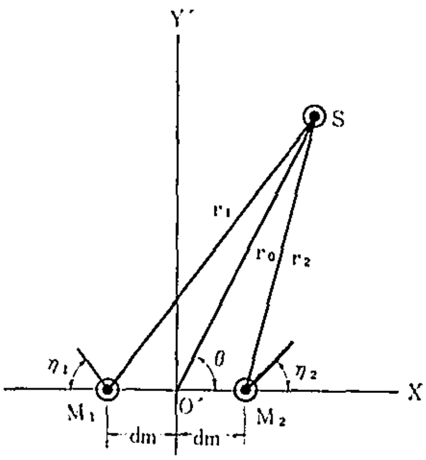


Figure 2.2.8(a) Sound Source Side

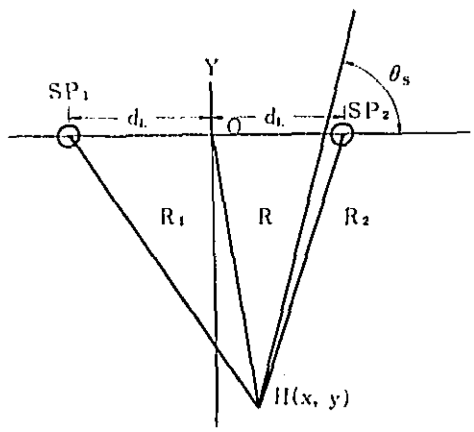


Figure 2.2.8(b) Listener Side

If, in the above diagrams

$$\alpha = 1 - \frac{dm}{\Upsilon_0}\cos\theta$$

$$\beta = 1 + \frac{dm}{r_0} \cos\theta$$

$$\phi = dm \cos\theta + \frac{R1 - R2}{2} + \frac{\varepsilon}{2k}$$

ε: Phase difference between the left and right circuits

k: Wave length constant $k = \frac{2\pi f}{c}$ d: Sound Velocity

r: θ Direction sensitivity ratio of both microphones

$$r = \frac{1 + \cos(\theta - \eta_1)}{1 - \cos(\theta + \eta_2)}$$

With undirectional cardioid microphones. The direction of progress of the composite waves may then be calculated from the following equations. (H) here is the counter-clock-wise angle as viewed from the front of the "X" axis.

$$\tan\theta = \frac{\frac{ky}{2}\left(\frac{1}{R_1}+\frac{1}{R_2}\right)\left\{\frac{\alpha^2}{R_1^2}+\frac{r^2\beta^2}{R_2^2}+2\frac{\alpha\beta r}{R_1 R_2}(\cos^2 k\phi - \sin k\phi)\right.}{\frac{k}{2}\left(\frac{x+d_L}{R_1}+\frac{x-d_L}{R_2}\right)\left\{\frac{\alpha^2}{R_2^2}+\frac{R^2\beta^2}{R_2^2}+\frac{2\alpha\beta r}{R_1 R_2}(\cos^2 k\phi - \sin_2 k\phi)\right\}} *$$

$$* \frac{\frac{2\alpha\beta r y}{R_1R_2}\left(\frac{1}{R_1^2}-\frac{1}{R_2^2}\right)\sin ky \cos ky + \frac{ky}{2}\left(\frac{1}{R_1}-\frac{1}{R_2}\right)\left\{\left(\frac{\alpha}{R_1}\right)^2+\left(\frac{\gamma\beta}{R_2}\right)^2\right\}}{-\frac{2\alpha\beta\gamma}{R_1R_2}\left(\frac{x+d_L}{R_1^2}-\frac{x-d_L}{R_2^2}\right)\sin k\phi \cos k\phi + \frac{k}{2}\left(\frac{x+d_L}{R_1}-\frac{x-d_L}{R_2}\right)\left\{\left(\frac{\alpha}{R_1}\right)^2+\left(\frac{\gamma\beta}{R_1}\right)^2\right\}} \quad (2.1)$$

Equation 2.1 is the wave length time constant 'k' or, in other words, is the function of the sound source frequency. This means that, when the sound source frequency varies, the direction of orientation changes.

In natural sound fields, one does not sense any movement in the sound source despite changes in the sound source frequency.

This type of abnormality may arise in 2-circuit stereo sound fields. However, if the conditions $\sin k\phi \fallingdotseq k\phi$, $\cos k\phi \fallingdotseq 1$ may be considered

$$k\phi = kdm\cos\Theta + k\frac{R_1 - R_2}{2} + \frac{\varepsilon}{2} \ll 1 \qquad (2.2)$$

orientation will no longer be changed by the frequency as the portion relating to k in equation (2.1) will disappear.

Under the assumption that the special conditions in equation (2.2) are met and the listener listens to 2 speakers from a position ($R_1 = R_2$, $x = 0$) equidistant from the speakers and using a coaxial microphone ($dm = 0$) under sound recording conditions, and moreover, with no phase difference between the two circuits ($\varepsilon = 0$), equation (2.1) will become

$$\Theta = \tan^{-1}\frac{y(1+\gamma)}{d^L(1-\gamma)} \qquad (2.3)$$

and will become unrelated to frequency.

(3) Relation between sound orientation and level and phase difference between the channels.

Considering recording conditions with microphone spacing at zero for the symmetrical listening positions on the 'Y' axis. Also, substituting $\theta s = \pi/2 - \theta$ and expressing the right side as positive and the left as negative with the direction of the sound positioned at the front at 0°, it will be as follows.

$$\tan\Theta_S = \frac{\frac{-2r}{kR_0}\sin\varepsilon + (1-\gamma^2)}{1 + 2\,r\cos\varepsilon + \gamma^2}\tan\theta_0 \qquad (2.4)$$

$\tan\theta_0 = \frac{-y}{dL}$ and θ_0 is one half the predicted angle of the left and right speakers from the listening position.

(i) When no phase difference exists

The calculated results of the relation between γ and θ_S in Figure 2.2.9 are shown in Figure 2.2.10 as measured values.

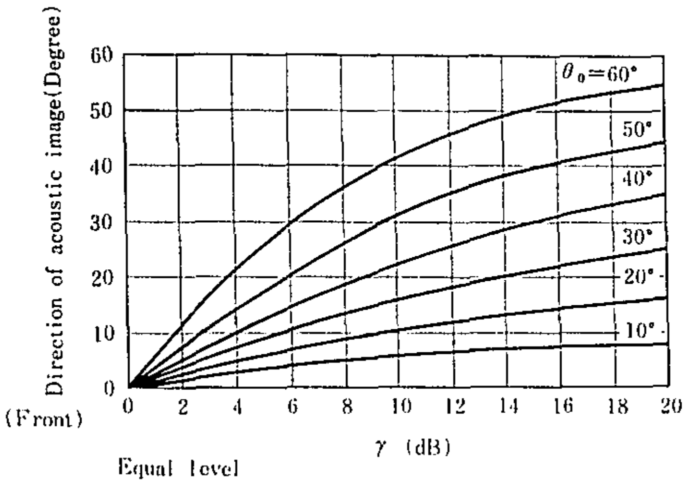
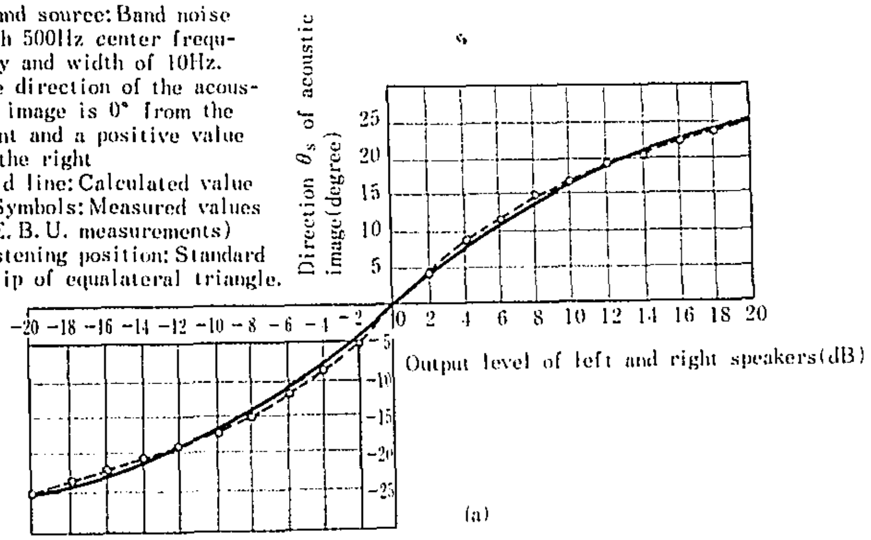


Figure 2.2.9 Direction of r and Acoustic Image

1. Sound source: Band noise with 500Hz center frequency and width of 10Hz.
2. The direction of the acoustic image is 0° from the front and a positive value to the right
3. Bold line: Calculated value
4. o Symbols: Measured values (E. B. U. measurements)
5. Listening position: Standard - Tip of equalateral triangle.



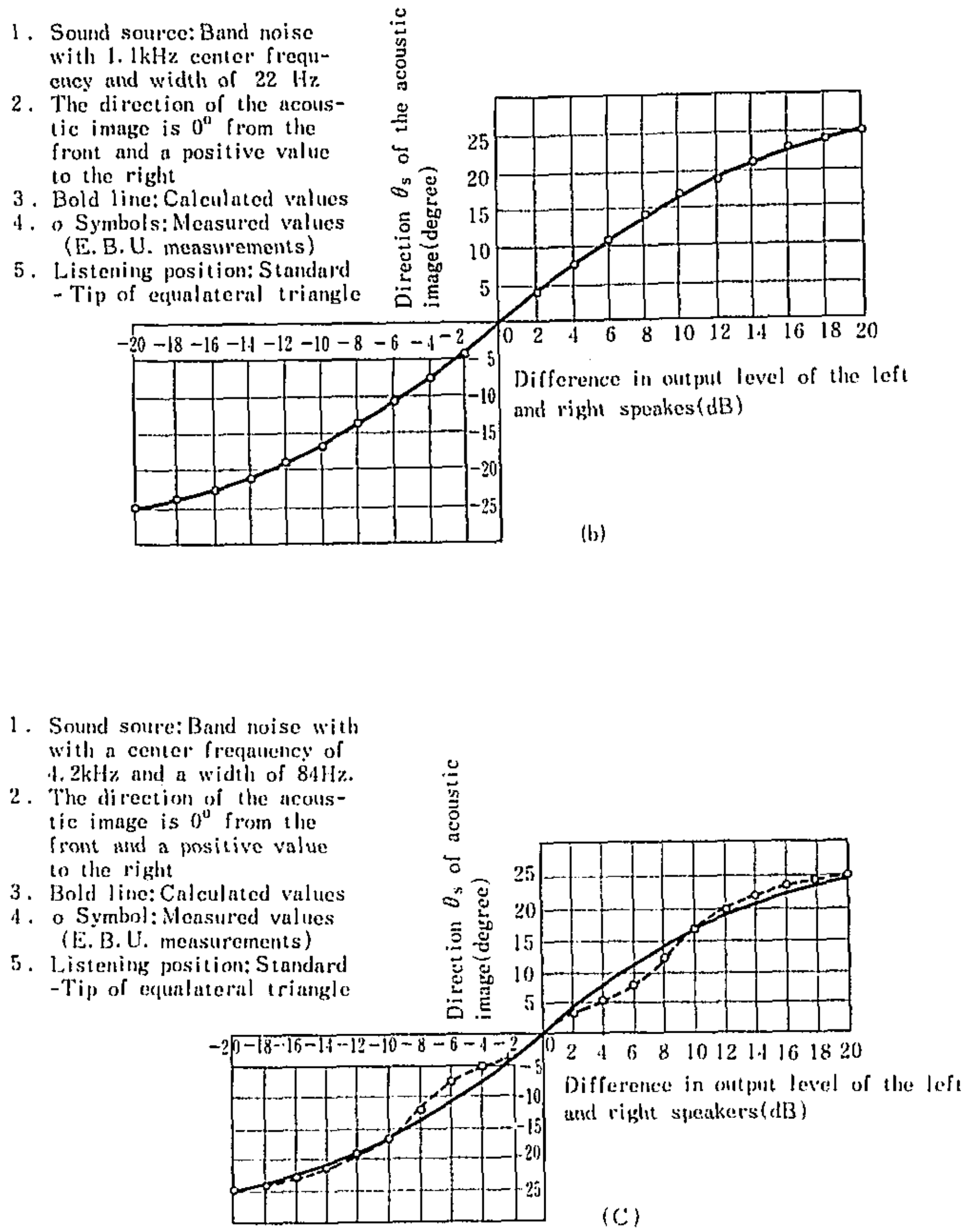


Figure 2.2.10 Output ratio of the left and right speakers and measured values of the calculated acoustic image position

(ii) When phase difference exists

$$\frac{c}{\pi R_0 f} \ll (\gamma > 1), \quad \frac{c}{\pi R_0 f} \gg (\gamma < 1)$$

$$\tan \theta_S = \frac{1 - \gamma^2}{1+2\gamma \cos\varepsilon + \gamma^2} \tan \theta_0 \tag{2.5}$$

As an example, if calculations are made with the left and right speakers and the listening position at the tips of an equalateral triangle, it would appear as in Figure 2.2.11.

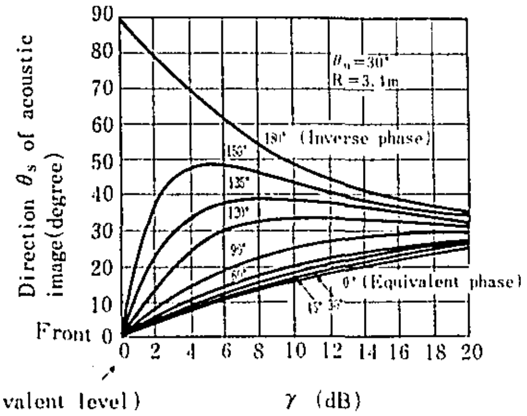


Figure 2.2.11 Direction of acoustic image and difference in level and phase

When frequency decreases. θ_S will be influenced by advances and delays in phase and the resulting deviations are shown in Figure 2.2.12.

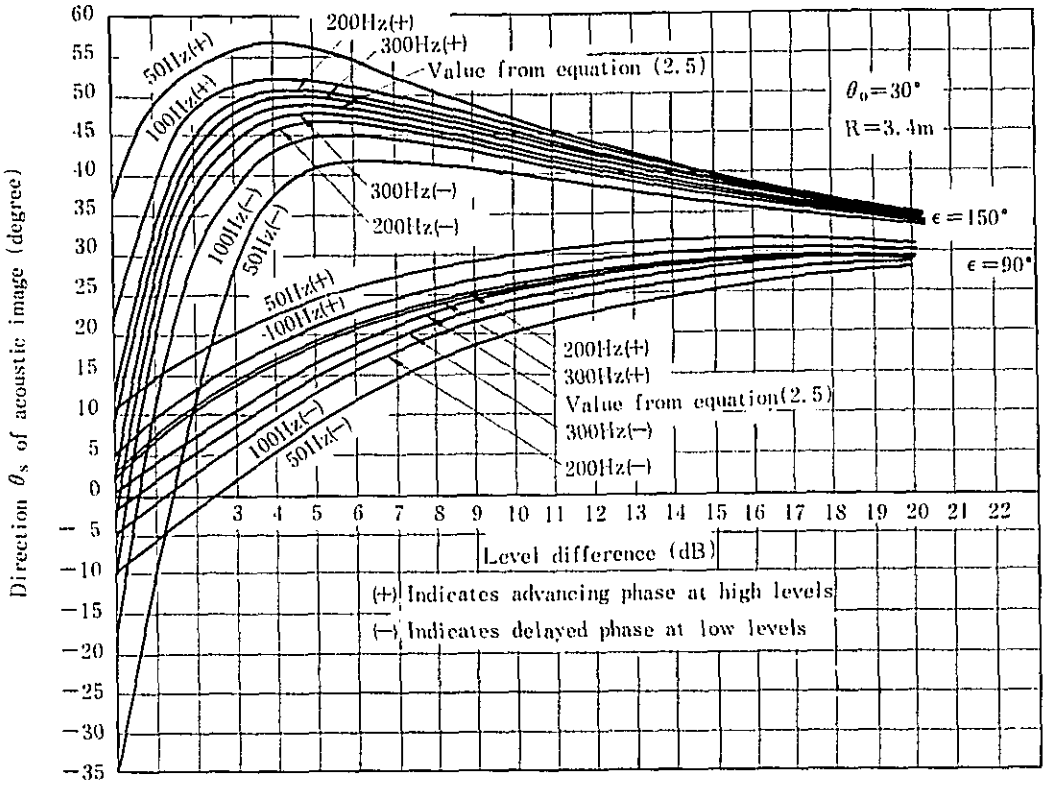


Figure 2.2.12 Direction of acoustic image and difference in level and phase

Theoretical and measured values under various conditions are shown in Figure 2.2.13 and it may be noted that they are well matched.

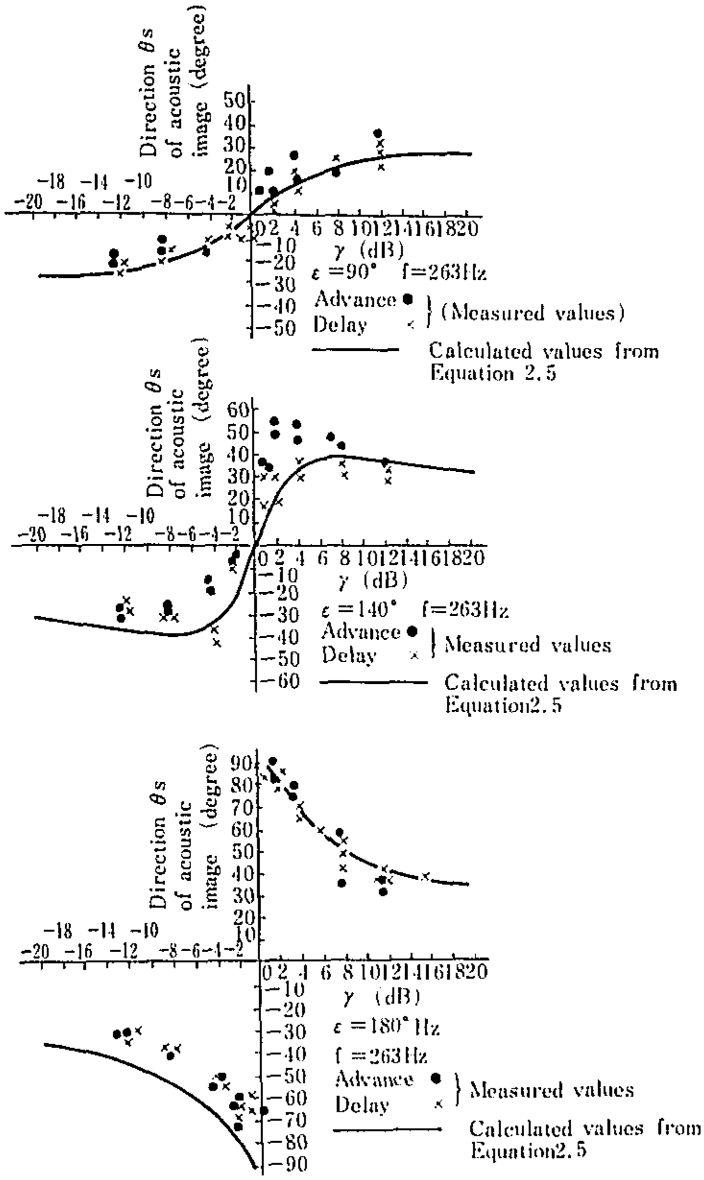


Figure 2.2.13

2.2.4 Stereo Effects and Sound Parameters Peculiar to Stereo (1)

(1) Crosstalk

When crosstalk is caused between left and right channels, the sound stage (range of distribution of the apparent sound source in stereo reproduction) is narrowed and stereo effects diminished.

According to test results of detection limits (Figures 2.2.14 and 2.2.15) when crosstalk is applied between channels, practically none could be detected with crosstalk under -30dB. If the crosstalk is under -20dB, it would be satisfactorily for stereophonic broadcast.

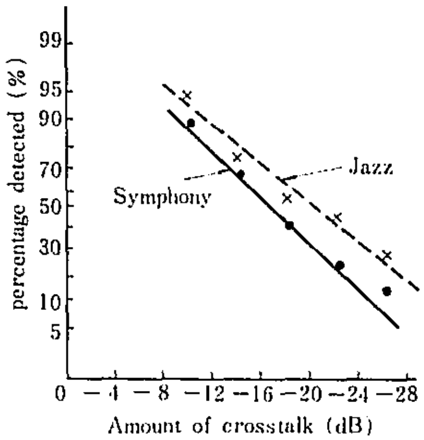


Figure 2.2.14
The percentage of difference detected with and without crosstalk (NHK)

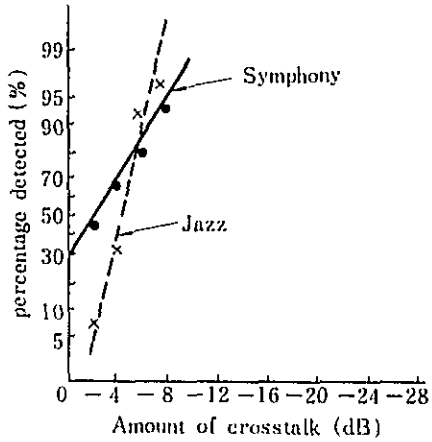


Figure 2.2.15
The percentage of difference detected with crosstalk of 0 dB and less than 0 dB (NHK)

(2) Difference in level between channels

When differences in levels develop between channels, the position of the acoustic image moves and the stereo effects of the sound source are obstructed. Although from the results of level fluctuation between channels in Figures 2.2.16 and 2.2.17, a target value within 0.5 dB should be considered. Since the detection limit will naturally be a high value when level fluctuations arise gradually, in practice, the standard value may be set slightly higher than the above value.

(3) Phase and time difference between channels

When phase and time differences develop between channels, not only will the normal direction of the acoustic image change but the acoustic image will be flurred and one may sense changes in tone quality.

As shown in Figure 2.2.18, there is practically no variations in the normal direction up to a phase difference of about 30° but will increase sharply above 90°. Time differences of about 250 μs is permitted.

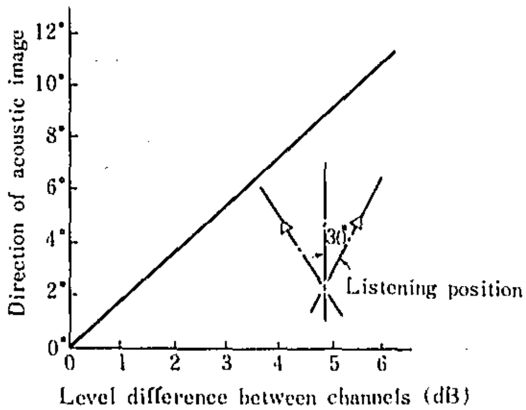


Figure 2.2.16
Relation between level difference between channels and direction of acoustic image (According to Makita's formula)

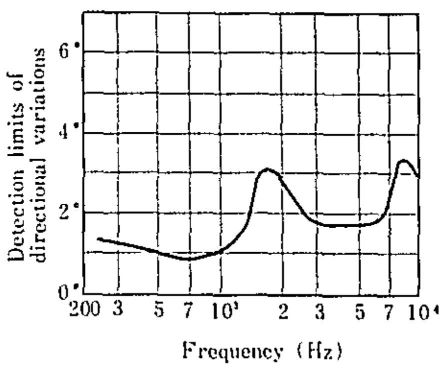


Figure 2.2.17
Detection limits of variation in left and right directions of sound source at front (Mills)

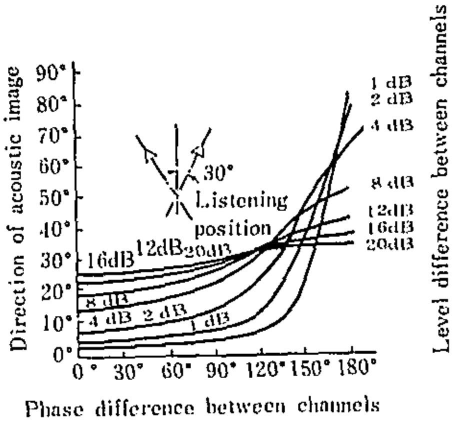


Figure 2.2.18 Phase difference between channels and direction of acoustic image (From Makita's formula)

Table 2.2.1 Standards of phase difference between channels (E.B.U.)

Detection Limit		Permitted Limit	
Frequency(Hz)	Phase Difference	Frequency(Hz)	Phase Difference
50	90°	50	90°
200	30°	200	45°
3,750	30°	2,500	45°
15,000	90°	10,000	90°
		15,000	90°

2.3 Stereo Broadcasting Modes

2.3.1 Conditions for Stereo-Type FM Broadcasts

As stereo sound signals are composed of a left and right signals, as a rule, the transmission of these signals require two separate transmission circuits.

However, since the usage of 2 radio waves is not desirable from the standpoint of utilization of radio waves, it will be necessary to use a single radio wave.

When stereo type FM broadcasts are to be made, a system satisfying the following conditions will be desirable.

(1) To be compatible with monophonic broadcasts
Compatibility means trouble-free monophonic reception of stereophonic broadcast waves, involving problems of sound quality and coverage area.

(2) Satisfactory sound quality to be obtained in both monophonic and stereophonic broadcasts.

(3) The difference in coverage area of monophonic and stereophonic broadcasting should be minimal.

(4) To be a system that can be popularized at prices approximately the same as for monophonic receivers.

(5) To be a system with international applicability.

2.3.2 Various Types of Stereophonic Broadcast Systems

If the types of signals are classified from an acoustical standpoint, they would be as follows.

(1) Sum and Difference Systems

In this system, the sum of the left (L) and right (R) signal M (L + R) and, the difference signal S (L - R) are respectively transmitted.

If these sums and differences are formed at the receiver side, they will be $M + S = 2L$ and $M - S = 2R$, and left and right signals R and L will be obtained.

(2) Left-right changeover system

This is a system in which the left and right stereophonic signals are switched over alternately and transmitted on a single circuit. The receiver side is synchronized with the transmitter signal changeover and thus produces the left and right signals.

If the higher portion of these changeover waves are disregarded, this system would actually be no different from the sum and difference system.

(3) Directional signal system

In addition to signal M, which is the sum of the stereo signals, a signal carrying information relative to the direction of the sound source is also transmitted. Signal M is controlled in the receiver by the directional signal, so outputs equal to the left and right signals are secured.

Although the Percival and Enkel systems are available, they are both based on the stereophonic signal envelope. These systems are ideal transmission systems as their directional signal band width is in the neighborhood of 2 to 300Hz but they have not come into practical use due to a serious drawback in their reproduced stereo effects.

If classified from the standpoint of the system of modulation, they would be as follows.

- Frequency Division System
  - FM - FM System
  - AM - FM System
    - Dual side bands
    - Carrier wave suppression
    - Single side band
- Time Division System
  - PAM - FM System

The stereophonic broadcasting systems suggested so far are shown in Table 2.3.1.

Table 2.3.1 A list of compatible stereophonic broadcasting systems

System	Modulation System	Contents of Main Channel	Contents of Subchannel	Band Width of Main Channel	Band Width of Sub-channel	Contents of First Signal	Contents of L and R Stereo-phonic Signal Output	Remarks
Crosby	FM - FM	(L + R)	(L - R)	15kHz	15kHz	(L + R)	2L, 2R	
Cablest	FM - FM	(L + R)	- R	15kHz	7kHz	(L + R)	L, R	
Halstead	FM - FM	(2L - R)	(2R - L)	15kHz	8kHz	(2L - R) (2R - L)	$\frac{3}{2}L$, $\frac{3}{2}R$	
Parker	FM - FM	(L + R)	- R	15kHz	30Hz ∿3,500Hz	(L + R)	L, R(30Hz∿3,500Hz) L+R (3,500Hz over)	
Zenith Radio Corp	AM - FM	(L + R)	(L - R)	15kHz	15kHz	(L + R)	2L, 2R	
G.E	AM - FM	(L + R)	(L - R)	15kHz	15kHz	(L + R)	2L, 2R	Suppression of Subcarrier
Geluk	FM - FM	(L + R)	(L - R)	30Hz ∿15kHz	30Hz ∿15kHz	(L + R)	2L, 2R	
Telefunken	FM - FM	(L + R)	(L - R)	15kHz	10kHz	(L + R)	2L, 2R	≒ Crosby
R.T.F.	AM - FM	L	R			L	L, R	
Philips	AM - FM	(L + R)	(L - R)			(L + R)	2L, 2R	
Siemens	Left Right Change-over System PAM - FM					(L + R)	L, R	Sampling Frequency 30kHz
Lowe - Opta	AM - FM	(L + R)	(L - R)			(L + R)	2L, 2R	≒ Philips L-R Balanced modulation
Lowe - Opta HMD	Left Right Changeover FM	(L + R)	(L - R)	15kHz	15kHz			Subcarrier Wave AM
Percival	FM - FM	(L + R)	Directional Information $D=\frac{L'}{L'+R'}$	15kHz	100Hz	(L + R)	(L+R)-(L+R)D (L+R)D	Directional Signal System
Enkel	AM - FM	(L + R)	DL DR	15kHz	(100Hz×2)	(L + R)	(L+R)DL (L+R)DR	

2.4 Technical Standards of Stereophonic Broadcasting

2.4.1 Standard System of Transmission

(1) Definition of the terms

A. Main channel signal for stereophonic use

Refers to the sum of the left and right signals.

B. Subchannel signals for stereophonic use

This is the difference between the left and right signals and refers to the side-band generated by carrying out carrier wave suppression amplitude modulation on the stereophonic subcarrier wave.

C. Pilot Signal

Refers to the control signal transmitted to enable receipt of stereophonic broadcasts.

(2) Modulation Signal

Composed of the stereophonic main channel signal, stereophonic subchannel signal and the pilot signal, and is structured according to the following related equations.

$A = M + S + P$

$M = L + R$

$S = (L - R)\sin \omega t$

$P = P \sin (\frac{\omega}{2} t + \theta)$

A: Signal voltage of main-carrier modulation

M: Signal voltage of main-channel

S: Signal voltage of subchannel

P: Pilot signal voltage

L: Left signal voltage

R: Right signal voltage

ω: 2π X subcarrier-frequency

p: Voltage amplitude of pilot signal

θ: Phase of pilot signal

t: Time

Frequency arrangements of modulated signals for stereophonic broadcasts are as shown in Figure 2.4.1.

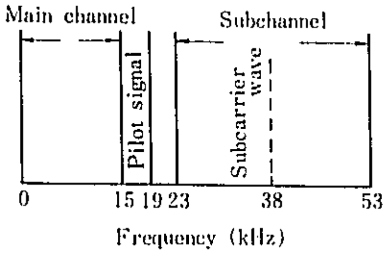


Figure 2.4.1

(3) The frequency of the stereophonic subcarrier wave shall be 38kHz.

(4) The phase of the pilot signal shall be 0°.

(5) Maximum percentage of modulation

The maximum percentage of modulation only by the left (or right) signal for the stereo main channel and subchannel signals shall be 45%. The pilot signal modulation shall be 10% at this time.

(6) Pre-emphasis

Pre-emphasis with a time constant of 50μs shall be applied to the left and right signals or to the sum and difference of these signals.

2.4.2 Stereo Broadcasting Transmission Characteristics

(1) The permissible deviation of the pilot signal frequency shall be ±2Hz.

(2) The permissible deviation of the pilot signal phase shall be ±5°.

(3) The permissible percentage of modulation of the pilot signal shall be over 8% but under 10%.

(4) The permissible percentage of modulation by the residual portion of the stereo subcarrier wave shall be under 1%.

(5) Overall frequency characteristics

The overall frequency characteristics from the left (or right) signal input terminal to the transmitter output terminal should be within the range of the permissible limit curve of the standard pre-emphasis characteristics shown in Figure 2.4.2.

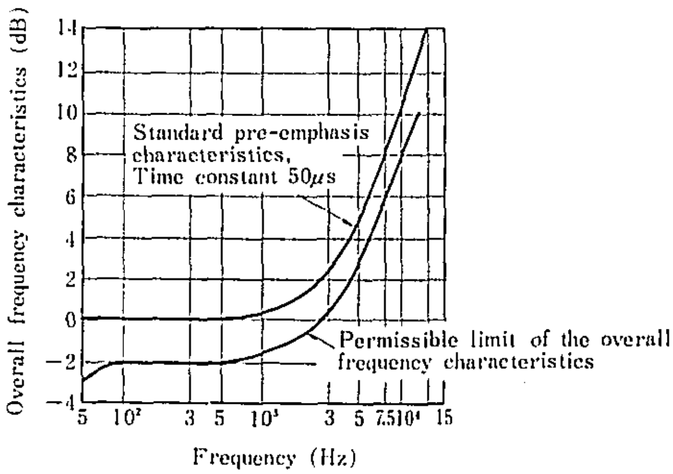


Figure 2.4.2

(6) Overall Distortion Factor

The overall distortion from the left (or right) signal input terminal to the output terminal of the transmitter must be below the values shown in the following table at maximum modulation.

Modulation Frequency		Overall Distortion Factor
Over 50Hz	Under 10,000Hz	2.0%
Over 10,000Hz	Under 15,000Hz	3.0%

(7) Signal to Noise Ratio

The signal-to-noise ratio from the left (or right) signal input terminal to the output terminal of the transmitter must be over 55 dB under the standard conditions of maximum modulation at 1,000Hz.

(8) Left and Right Channel Separation

Left and right channel separation must be over 30 dB at maximum modulation, with a modulation frequency ranging from 100Hz to 10,000Hz. Left and right channel separation indicates the ratio between the left (or right) signal output and the right (or left) signal output at the output terminal of the transmitter when only a left (or right) signal is transmitted.

(9) Left and right signal level difference.

When equal signals are impressed on the left and right signal input terminals, the difference in levels between the left and right signals at the output terminal of the transmitter must be within 1.5 dB, over the frequency range of 100Hz to 10,000Hz.

2.4.3 Relation between the Transmission System and Left-Right Separation

As shown in Figure 2.4.3, the L and R signals are divided into sum signals (L + R) and difference signals (L - R) in the matrix circuit. Each single then passes through the main channel and subchannel transmission systems and returns to the matrix circuit to be converted into L' and R' singles. There will be no fear of crosstalk between L and R', and R and L', if the matrix circuit is normal and the characteristics of the main and sub-transmission systems match. However, if any difference exists in gain or phase in the two transmission systems, left-right separation will deteriorate and crosstalk will occur.

In this case, as the problem will only be the differences in characteristics between the main and sub-transmission system, set main transmission system characteristics as 1 and sub-transmission system characteristics as $xe^{j\theta}$ (Gain at x times and phase advanced θ radians).

Therefore signal M' reaching the main channel will be $\frac{(L + R)}{2}$ and signal S' reaching the sub-channel will be $\frac{(L - R)}{2} \; xe^{j\theta}$.

If both of these signals are applied to the matrix circuit, the following L' R' output will be obtained at the output terminal.

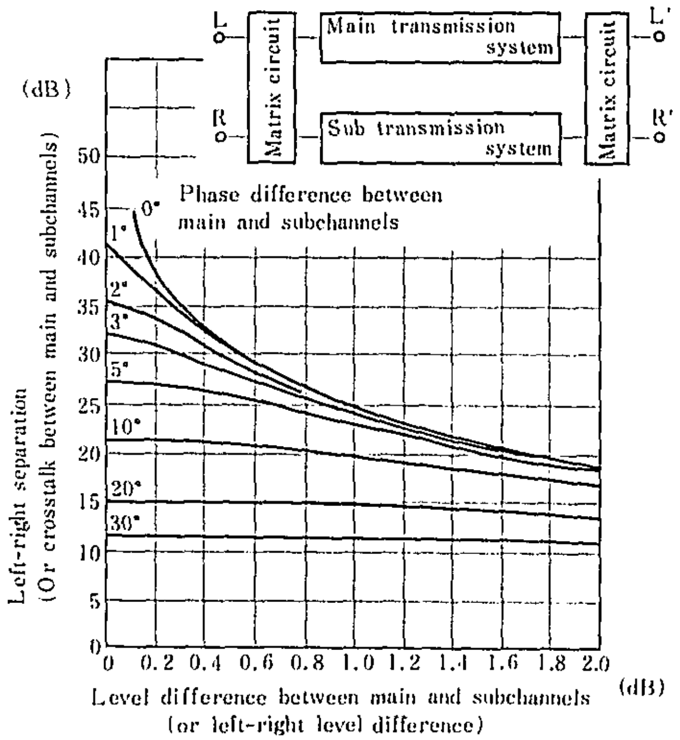


Figure 2.4.3 Left and right separation of characteristic difference in main and subchannel and, left and right level difference due to crosstalk

$$L' = M' + S'$$

$$= \frac{(L+R)}{2} + \frac{(L-R)}{2} \; xe^{j\theta} = \frac{L}{2}(1+ xe^{j\theta}) + \frac{R}{2}(1- xe^{j\theta})$$

$$R' = M' - S'$$

$$= \frac{(L+R)}{2} \quad \frac{(L-R)}{2} \; xe^{j\theta} = \frac{L}{2}(1- xe^{j\theta}) + \frac{R}{2}(1+ xe^{j\theta})$$

If modulation is carried out with the L signal only, the following equation will be obtained by setting R as 0.

$$L' = \frac{L}{2}(1 + xe^{j\theta})$$

$$R' = \frac{L}{2}(1 - xe^{j\theta})$$

Since the left and right separation in this case is $\left|\frac{L'}{R'}\right|$ in accordance with definitions in Paragraph (8), it will be

$$\left|\frac{L'}{R'}\right| = \left|\frac{1 + xe^{j\theta}}{1 - xe^{j\theta}}\right| = \frac{1 + 2x\cos\theta + x^2}{1 - 2x\cos\theta + x^2}$$

Or, if the above equation is expressed by dB's, it will be

$$\text{Left-Right Separation (dB)} = 10 \log \frac{1 + 2x\cos\theta + x^2}{1 - 2x\cos\theta + x^2}$$

The results of the above calculations are shown in Figure 2.4.3. The 30 dB right-left separation in the diagram corresponds to a level difference of 0.3 dB and a phase difference of 3°. A level difference of 0.5 dB will be permitted if the phase difference is zero. Special will be required to hold the difference in characteristics of the main and sub-transmission systems within 0.3 dB and 3 degrees over the entire range. Particular consideration must be taken to the phase characteristics in the high frequency range (near 15kHz) when filters are to be used. As very slight differences in characteristics will worsen the right-left separation when the sum and difference signal is used in stereo signal transmission, it will be advantageous to transmit left and right signals as they are.

2.4.4 Crosstalk and Difference in Level between the Left and Right Signals

Although the permissible limit of level difference between L and R is 1.5 dB in Paragraph (9), let us calculate the permissible amount of crosstalk if this is to be based on crosstalk between the main and sub-transmission system.

Let us consider a case in which equal signals are applied to the L and R terminals in Figure 2.4.3. If there is no crosstalk between the two systems, signal M' of the main transmission system will be $\frac{(L + R)}{2}$ and signal s' of the sub-transmission system will be $\frac{(L - R)}{2}$ and matrix output will be $L' = M' + S'$

$$= \frac{(L+R)}{2} + \frac{(L-R)}{2} = L$$

$$R' = M' - S'$$

$$= \frac{(L+R)}{2} - \frac{(L-R)}{2} = R$$

There will be no level difference between the signals (When L = R)

Next, when $xe^{j\theta}$ crosstalk exists between the two systems, it will be

$$M' = \frac{(L+R)}{2} (1 - xe^{j\theta}) + \frac{(L-R)}{2} xe^{j\theta}$$

$$S' = \frac{(L-R)}{2} (1 - xe^{j\theta}) + \frac{(L+R)}{2} xe^{j\theta}$$

and at matrix output, it will be

$$L' = M' + S' = L$$

$$R' = M' - S' = R (1 - 2 xe^{j\theta})$$

The difference in level between the two signals will therefore be

$$\left|\frac{L'}{R'}\right| = \left|\frac{1}{1 - 2xe^{j\theta}}\right| \qquad (L = R)$$

In the equation in Paragraph 2.4.3 expressing left and right separation when difference in characteristics exists between the transmission systems, the following equation may be set up if 'x' is very small compared to 1.

$$\left|\frac{L'}{R'}\right| = \left|\frac{1 + xe^{j\theta}}{1 - xe^{j\theta}}\right| \fallingdotseq \left|\frac{1}{1 - 2xe^{j\theta}}\right|$$

This matches the equation expressing the level difference between the left and right signals when crosstalk exists between the two transmission systems. Figure 2.4.3 may therefore be used to indicate the relations between crosstalk and level difference. For example, to secure a left-right level difference of 1.5 dB, it will be necessary that the crosstalk be less than 21 dB.

2.5 Stereophonic Modulation Signals

2.5.1 Modulation Wave Forms of Stereophonic Signals

As shown in Figure 2.4.1, the subchannel signal is formed by modulating a 38kHz subcarrier wave with an (L - R) difference signal. However, as unmodulated carrier waves are superfluous in information transmission, a balanced modulator is employed to suppress the subcarrier waves during transmission. As the receiving side will not be able to carry out demodulation without a subcarrier wave, one half of 38kHz, or 19kHz, will be sent as a pilot signal and the receiving side will reproduce a subcarrier wave synchronized to this pilot wave and carry out demodulation. If the 38kHz signal itself was used as the pilot signal, the low frequency component may be cut off, and as it would be very difficult to match the phase of the pilot signal and the subcarrier wave generated in the receiver, the 19kHz pilot signal was selected as it will least affect the main and subchannels.

The following equation shows the subchannel signal when (L - R) is a sine wave. ω_p is the angular frequency of the difference signal (audio frequency) and ω the angular frequency of the subcarrier wave.

$$S = \sin \omega_p t \sin \omega t$$
$$= 1/2 \{ \cos(\omega - \omega_p)t - \cos(\omega + \omega_p)t \}$$

The above equation shows that the signal is only composed of upper and lower side bands and that no subcarrier elements are included. If the above equation is put in diagram form, the envelope indicated by the heavy dotted lines in Figure 2.5.1 (a) is sin ω_p and, would correspond to the heavy dotted lines in Figure 2.5.1 (b) if variations in amplitude are not considered. Also, the phase of the suppressed carrier wave amplitude modulated wave form in Period II will be in reverse relation to the subcarrier wave phase in Period I. This will be used to discriminate the left and right signals explained later. The fine lines in Figure 2.5.2 will be the subcarrier wave form in Period I in Figure 2.5.1 (b).

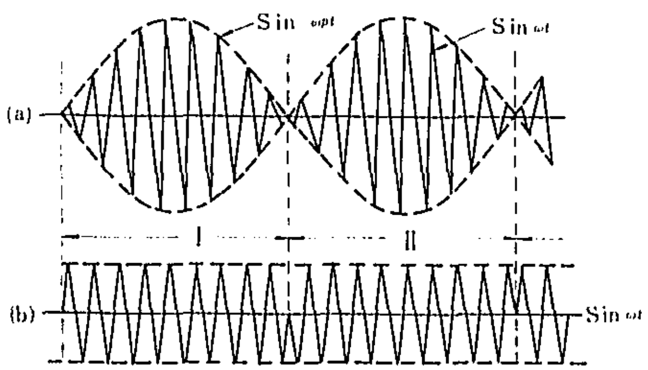


Figure 2.5.1 (a) Suppressed carrier wave amplitude modulated wave form
(b) Carrier wave form

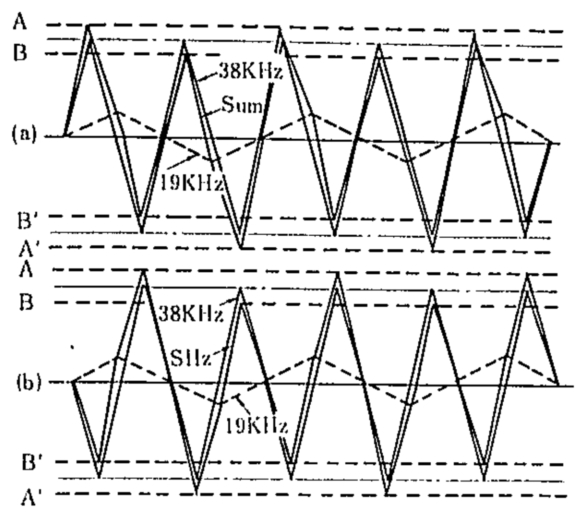


Figure 2.5.2 Superposed wave form of the subcarrier wave and the pilot signal
(a) Conditions in Period I (b) In Period II

Also, as the pilot signal is as per the dotted line in the same diagram, the high and low portion will be formed on every other wave of the superposed wave form as shown by the bold line. If AA' is the external envelope of this wave and BB' the internal envelope, the following may be noted by closely observing the relation between the drop in the wave and these envelopes. That is, the positive portion of the inclination straddles external envelope AA' or internal envelope BB'. In relation to this, the negative portion of the inclination straddles both external and internal envelopes.

Figure 2.5.2 (b) shows the superposed wave form in Period II and, though the 19kHz phase is the same as in (a), but the 38kHz phase is in reverse thus casuing the superposed wave form in this diagram. In contrast to that in (a), the wave top inclination portion all straddles the external and the internal envelope while the negative inclination portion straddles the external envelope AA' or the internal envelope BB'.

Next, if the 38kHz modulated with a sine wave, that is as in Figure 2.5.1 (a), is rewritten as in Figure 2.5.3 (b) and the pilot signal in Figure 2.5.1 (c) were superposed on this, we would obtain Figure 2.5.3 (d). Although Period I and Period II appear alike at first glance, a closer study will reveal that the features of Figure 2.5.2 remain intact. That is, the positive portion of the inclination straddles AA' or BB, in Period I and, AB' or BA' in Period II.

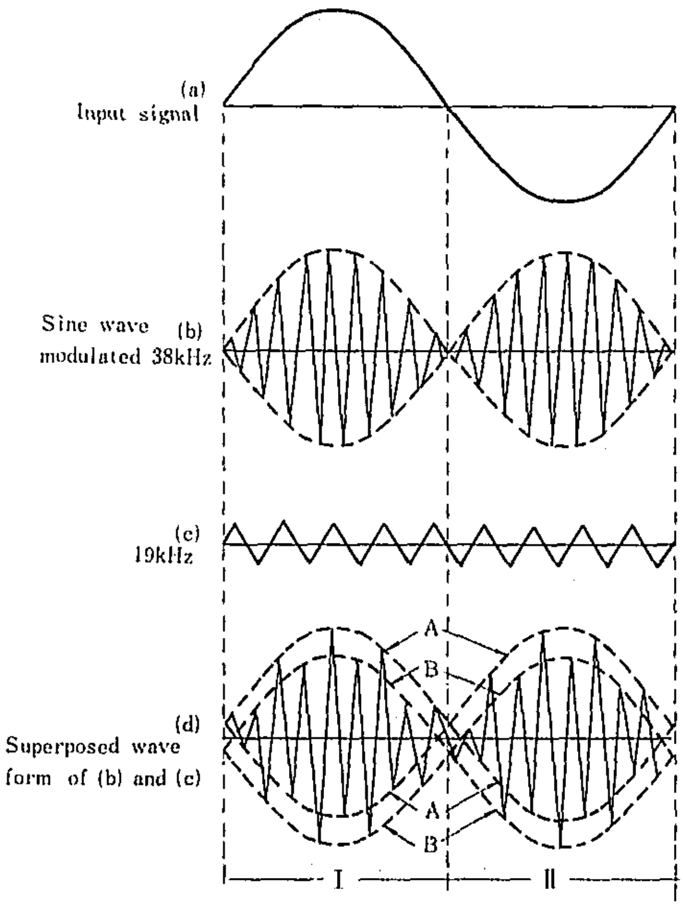


Figure 2.5.3 Superposed wave form (d) of the sine wave modulated 38kHz signal (b) and 19kHz signal (c)

Although the foregoing dealt with modulation by difference signals only, we will now study wave forms modulated by the left signal alone. As this means that R = 0 so both the sum and difference signals follow the wave form of signal L. When signal L is a sine wave, signal M will appear as the sine wave in Figure 2.5.3 (a) and (S + P) will be as in Figure 2.5.3 (d). Although the wave form of (a) plus (d) is as shown in Figure 2.5.4 (a), since this (M + S + P), it will be the composite wave form modulated by the left signal alone.

Since L = 0 when modulated with the right signal only,

Sum Signal = R   Difference Signal = -R

In this instance, the subcarrier wave modulated with (-R) will appear in reversed phase form as in Figure 2.5.3 (b). Its composite signal wave form will therefore appear as in Figure 2.5.4 (b). Although (a) and (b) appear to be the same at first glance, but in (a), the positive portion of the inclination of the area above the time axis, straddles the external envelope or the internal envelope whereas, in (b), the positive portion of the inclination of the area

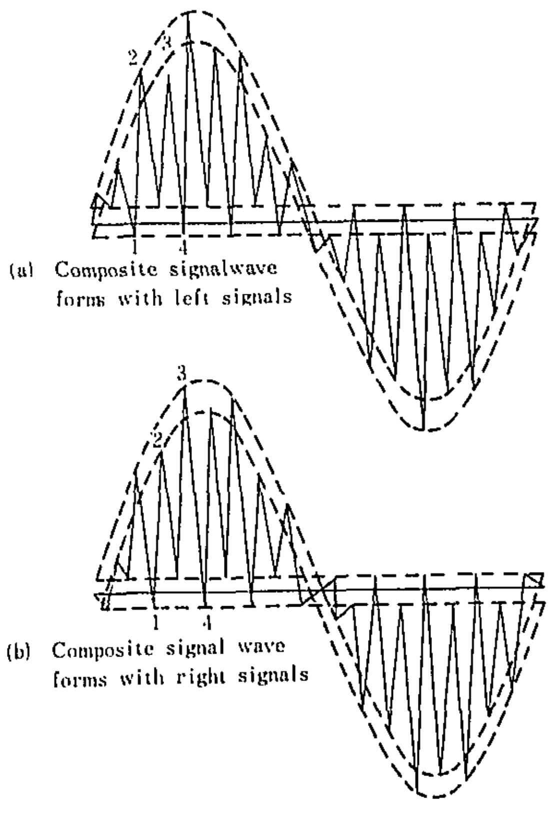


Figure 2.5.4 Composite signal wave form

above the time axis, straddles all the internal and external envelopes. Or, if we look as this from another angle, it may be noted that, between points 1 to 4 on the wave form, the wave traverses from point 2 on the outer envelope to point 3 on the inner envelope with left signals, whereas, with right signals it traverses from point 2 on the inner envelope to point 3 on the outer envelope. This serves as clue to distinguish the composite left and right signals. This feature also remains unchanged even if the polarity is reversed when connected up to an oscilloscope. This may be explained by reversing the top and bottom portion of the diagram in Figure 2.5.4, about its time axis and observing that the same feature is maintained. Observation of the polarity by this method may be simplified by setting the modulating frequency between 1 to 3 kHz or one integer part of 38kHz.

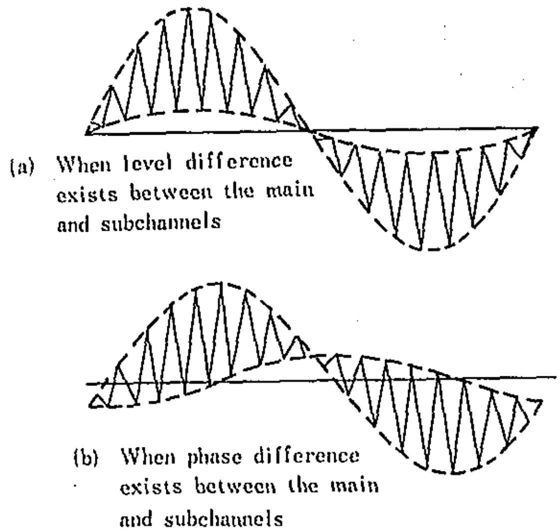


Figure 2.5.6

Composite signal wave form when level or phase differences exists between the main and subchannels

As to maintain the correct left and right polarity is extremely important in stereophonic broadcasting, it will be necessary to normally confirm that the left and right audio input terminals and the left and right modulation wave forms are matched.

The main carrier wave modulation polarity is defined as being positive when the frequency deviates to the high side, and this may not be distinguished by measuring with a sine wave. The general procedure in use is to modulate it with a non-symmetrical audio frequency and observe the direction of frequency deviation on the spectrum of a side band direct-viewer; and the direction of the modulation wave form on an oscilloscope.

The composite signal wave form, when differences in level and phase exist between the main and subchannel, is shown in Figure 2.5.6 and the differences may be easily detected by observing the wave form.

2.5.2 Modulation Level

Let us consider a normal case in which the L and R frequencies and phases are the same but have different amplitudes.

$$\left.\begin{aligned} L &= A \sin\omega t \\ R &= B \sin\omega t \end{aligned}\right\} \qquad (2.6)$$

then

$$\left.\begin{aligned} M &= L + R = (A + B) \sin\omega t \\ S &= L - R = (A - B) \sin\omega t \end{aligned}\right\} \qquad (2.7)$$

and the modulation wave form will therefore be as shown in Figure 2.5.8. Since $(L - R) \sin \omega st$ is at the top and bottom of $L + R$ as a standard, the envelope will be 2L, 2R. As shown in the diagram, when $A > B$, the maximum deviation will be 2A. When $A = B$ or $B = 0$, it will be the same as that shown in Figure 2.5.7. This means that, as long as $A \geqq B$, maximum deviation will be 2A regardless of the size of B.

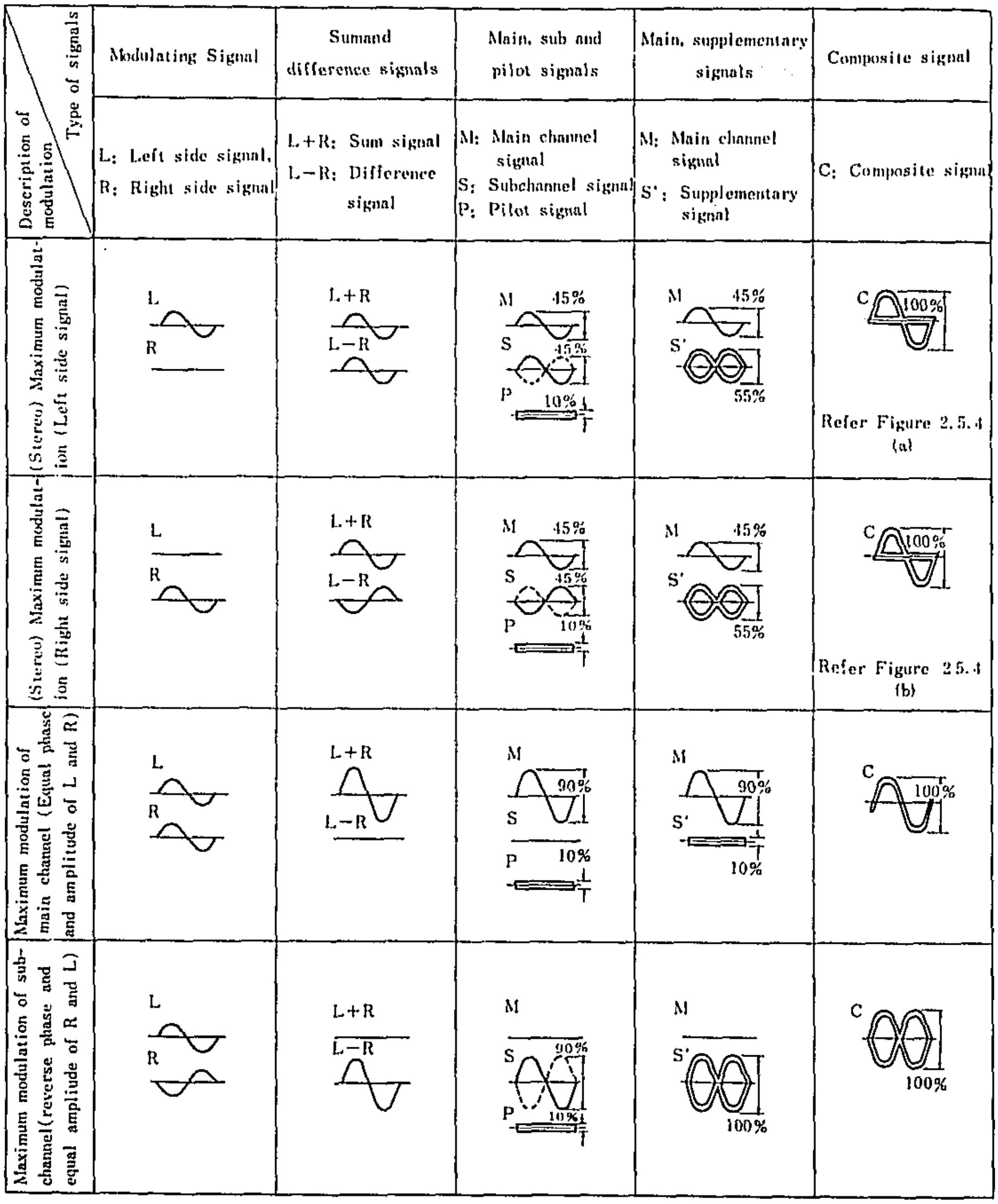


Figure 2.5.7 Stereo signal wave forms

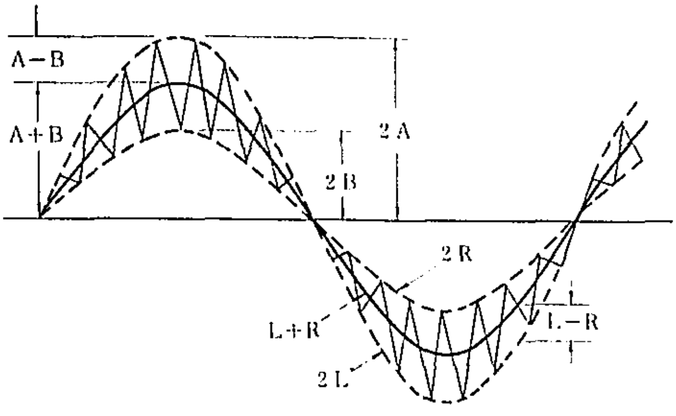


Figure 2.5.8 Modulation wave form with L and R with the same frequency and phase

Next, if we consider the L and R signals are of the same frequency but have a shift in phase, it would appear as in Figure 2.5.9. A sub-carrier wave with an amplitude of $|L - R|$ will exist with L + R as a standards.

This envelope is 2L, 2R and when $|L| > |R|$, its maximum deviation will become $2|L|$. When $|R| > |L|$, $2|R|$ becomes the maximum deviation at that point. The line that connects the outsides of the envelope is the line that connects $|L|$ or $|R|$ whichever is the larger.

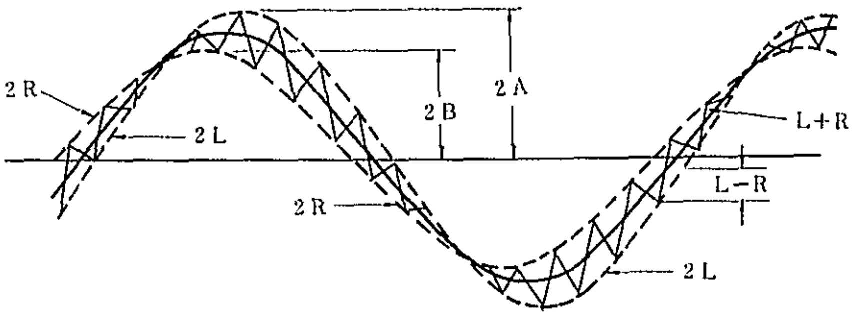


Figure 2.5.9 Modulated wave form with L and R with same frequency but with same frequency but with different amplitude and phase.

The modulated wave form when L and R are modulated with different frequencies will be as shown in Figure 2.5.10. A 38kHz sub-carrier wave is present between 2L and 2R with the center line of this 38kHz naturally being L + R. When A > B, maximum modulation will be determined by 2A regardless of B. If B > A, then maximum modulation will be determined by 2B regardless of A.

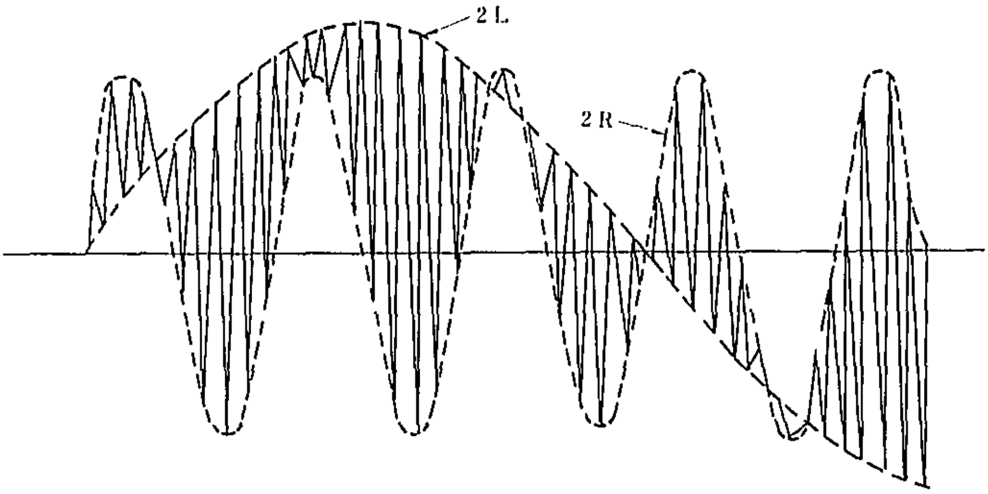


Figure 2.5.10 Modulated wave form when L and R are modulated with different sine wave frequencies

An example of a modulated wave form in which L and R are modulated with an optional wave form is shown in Figure 2.5.11. Same as the previous case, the sub-carrier wave envelope is 2L, 2R.

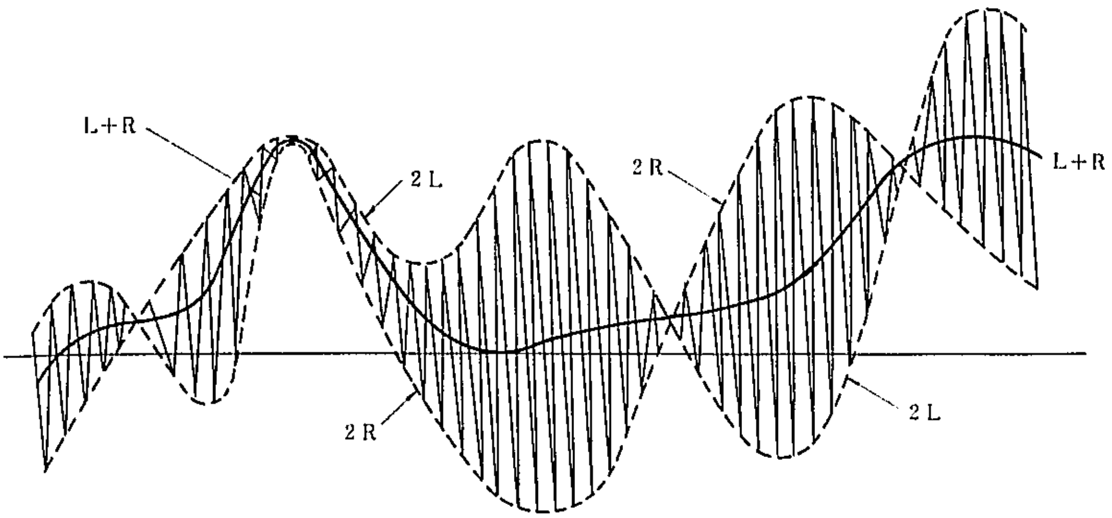


Figure 2.5.11 Modulated wave form when L and R are modulated with an optional wave form

The center line is L + R. It is determined by the larger instantaneous value of L and R. As may be determined from the foregoing explanation, the degree of FM stereo modulation is governed by the larger of the L and R signals and is uninfluenced by the smaller. As a pilot signal of 10% will actually be superposed on this, the remaining 90% is used to transmit the signal. Since frequency deviation becomes 2A, 2B and is twice that in the case of monophonic transmission, stereophonic level is set as follows. First, modulate with the left signal only and, with its audio portion (with the pilot signal and sub-carrier wave removed), modulate 75kHz 45%, that is, cause it to deviate 33.75kHz. If the sub-carrier were added at this time (without the pilot signal), the deviation would be 90% or 67.5kHz. Next, if we remove the left signal and modulate the right signal only until its audio portion is 45% modulated, the L and R levels would correspond to 100% modulation.

Although the monophonic receiver will receive this stereophonic broadcast as L + R, as may be noted from Figure 2.5.11, it will only match stereo deviations at the instant L = R but will normally not be effectively modulated by maximum deviations. Although differing with the program, compared to mono transmission, a 1 dB loss, that is, the 10% pilot signal, will result in stereophonic transmission monophonic reception when L = R. In the worse case, a 7 dB loss may result when L and R sound is only emitted alternately and modulation is only 45%. In normal programs it is said that there will be a drop in modulation equivalent to 2 to 4 dB when the pilot signal is not considered and 3 to 5 dB when the pilot signal is considered.

2.6 FM Stereo Wave Side Bands and Required Band Width

Let us study whether 99% of the side-band energy is within the ±100kHz band in stereophonic broadcasts as it is in monophonic broadcasts.

(1) Frequency deviation and band-width of monophonic broadcasts

Procedure-wise, let us first obtain the band-width for monophonic broadcasts.

If carrier frequency fc is frequency modulated with one audio frequency f_1, theoretically infinite side bands will be generated as in Figure 2.6.3 (a) and their size a_i may be expressed as follows:

$$a_i = J_i\ (m_1) \qquad (2.8)$$

$$m_1 = \frac{\Delta f_1}{f_1} \qquad \Delta f_1 \text{ Frequency Deviation}$$
$$\qquad f_1 \text{ Modulating Frequency}$$

The side band energy is expressed by the square of equation (2.8) and its total energy is the same as when unmodulated and may be expressed as follows:

$$J_0^2\ (m_1) + 2\ \{J_1^2\ (m_1) + J_2^2\ (m_1) + \cdots\} = 1 \qquad (2.9)$$

Since it will be necessary to determine "n" that will satisfy.

$$J_0^2\ (m_1) + 2\ \{J_1^2\ (m_1) + J_2^2\ (m_1) + \cdots + J_n^2\ (m_1) \quad = 0.99 \quad (2.10)$$

to secure a band width with 99% energy, we need only determine "n" that will satisfy

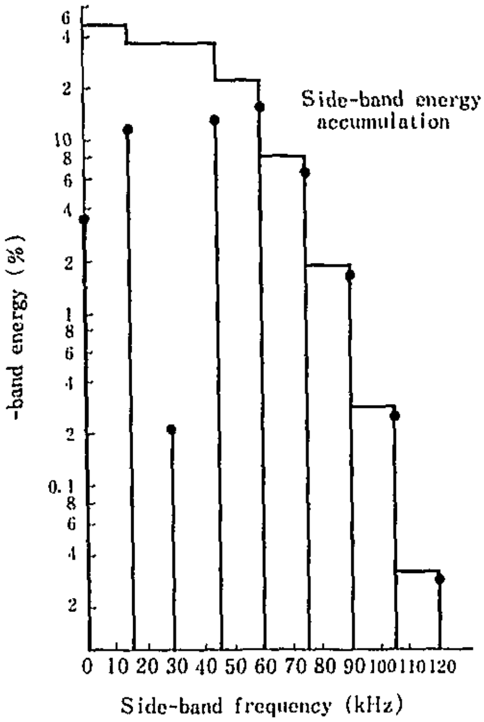
$$J_{n+1}^2\ (m_1) + J_{n+2}^2\ (m_1) + \cdots = 0.005 \qquad (2.11)$$

When this "n" is determined, the modulation frequency multiplied by this "n" will be the 99% frequency band width.

An example of side band energy distribution when Δf_1 = 75kHz and f_1 = 15kHz is shown in Figure 2.6.1. In as much as the upper and lower side bands are symmetrical, only the upper band is shown. Tee accumulated side band energy is the side band energy added successively from the higher side band frequencies and it may be noted from this diagram that the sum of the upper side band energy of 48.4%, the lower side band energy of 48.4% and the carrier wave energy of 3.2% totals 100%. It may be seen from this diagram that a cumulative side band energy of 0.5% indicates the 99% band and that this is 180kHz (±90kHz).

Figure 2.6.2 shows the relation between frequency deviation and frequency band width with a modulation frequency of 15kHz. This shows that the energy in the high side band frequencies cannot be ignored when frequency deviation is increased and also shows that the band width increases in 30kHz (±15kHz) steps.

It is generally said that a 99% width = $(f_1 + \Delta f_1) \times 2$ and this diagram substantiates this fact.



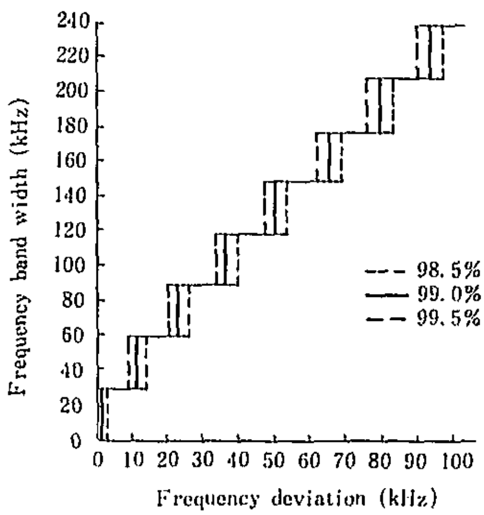


Figure 2.6.1 Side-band distribution of a 15kHz mono modulation wave

Figure 2.6.2 Frequency deviation and frequency band when modulated with a 15kHz mono signal

(2) Side bands when FM is carried out with a number of modulation frequencies.

Let us determine the side band spectrum when FM is carried out simultaneously with 2 modulation frequencies f_1 and f_2. Set frequency deviation due to f_1 as Δf_1, modulation factor $m_1 = \frac{\Delta f_1}{f_1}$,

frequency deviation due to f_2 as as Δf_2, modulation factor $m_2 = \frac{\Delta f_2}{f_2}$ and $f_1 > f_2$.

When modulated with f_1 only, as explained before, the side-bands align with f_1 spacing as shown in Figure 2.6.3 and their size is $J_1(m_1)$.

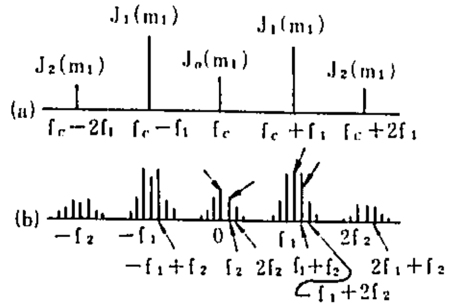


Figure 2.6.3 Side band distribution when modulated with (a) single frequency and (b) 2 frequencies

We will next consider the case when simultaneous modulation is carried out with f_1 and f_2. This, however, may be considered as first modulate with f_1 only to generate the side bands as in Figure 2.6.3 (a) and then modulate this further with f_2. When carrier wave f_c in Figure 2.6.3 (a) is frequency modulated with f_2, upper and lower side-bands are formed at f_2 space intervals. Although their sizes are $J_0(m_2)$, $J_1(m_2)$ and $J_2(m_2)$ for the carrier wave, No. 1 side band and No. 2 side band respectively, since the carrier wave itself decreases to $J_0(m_1)$ due to modulation with f_1, with simultaneous modulation with f_1 and f_2, these side bands (including the carrier wave) become $J_0(m_1)\ J_0(m_2)$, $J_0(m_1)\ J_1(m_2)$, and $J_0(m_1)\ J_2(m_2)$. Although the No. 1 side bands in Figure 2.6.3 (a) are generated by $f_c + f_1$, this is also frequency modulated by f_2 and upper and lower side-bands are formed spaced at f_2 intervals. Since their sizes are also the same as those near f_c at $J_0(m_2)$, $J_1(m_2)$ and $J_2(m_2)$, since the size of $f_c + f_1$ itself is $J_1(m_1)$, the generated side bands will become $J_1(m_1)\ J_0(m_2)$, $J_1(m_1)\ J_1(m_1)$,

$J_1(m_2)$ and $J_1(m_1)\ J_2(m_2)$. These naturally develop on both sides of $f_c + f_1$.

Under the same conditions, side bands are also formed on both sides of the No. 2 side bands $f_c + 2f_1$ or $f_c - 2f_1$ generated by f_1. Their sizes are $J_0(m_2)$, $J_1(m_2)$ and $J_2(m_2)$ multiplied by $J_2(m_1)$ and are shown in Figure 2.6.3 (b) excepting for the fact that their horizontal axis f_c has been deleted. When modulated with f_1 and f_2, side bands are normally generated at frequencies separated $n_1f_1 + n_2f_2$ (n_1 and n_2 are integers and include positive, negative and zero) from the carrier wave and their size may be said to be $Jn_1(m_1)Jn_2(m_2)$. The same conditions will arise when modulated with the 3 modulation frequencies f_1, f_2 and f_3 if their respective frequency deviations are Δf_1, Δf_2 and Δf_3 their modulation factors as m_1 $(=\Delta f_1/f_1)$, $m_2(=\Delta f_2/f_2)$ and $m_3(=\Delta f_3/f_3)$, side bands will be generated at frequencies separated by

$$n_1f_1 + n_2f_2 + n_3f_3 \qquad (2.12)$$

from the carrier wave and it may be easily comprehended that their size is

$$Jn_1(m_1)\ Jn_2(m_2)\ Jn_3(m_3) \qquad (2.13)$$

When calculating side band spectrum with stereo modulation, although simultaneous modulation with a greater number of modulation frequencies are involved and the number of figures in the calculation larger, the form of calculation will be the same as explained.

(3) Stereophonic broadcast band width

When modulating with a single sine wave in mono broadcasting if frequency deviation remains the same, the higher the modulation frequency the wider the side band. Since the same may be applied to stereophonic broadcasts, only the maximum modulation frequency was calculated.

(a) 15 kHz Modulation of the left signal

With the right signal unmodulated, apply a 15kHz sound frequency to the left signal terminal. Let us see what occurs when maximum frequency deviation is applied. The modulating frequency of the main carrier wave will be the sum of the main channel, pilot signal and the subchannel. If f_a is the 15kHz sound signal, f_p the 19kHz pilot signal and f_s the 38kHz subcarrier wave, it will be as if simultaneous modulation of the main carrier frequency is carried out by the 4 frequencies f_a (=15kHz), f_p (19kHz), $f_s - f_a$ (=38-15 = 23kHz) and $f_s + f_a$ (=38+15 = 53kHz). If f_1 to f_4 are these frequencies, Δf_1 to Δf_4 the frequency deviations and m_1 to m_4 the modulation factors, it will be as follows.

f_1 = 15kHz	Δf_1 = 33.75kHz	m_1 = 2.25
f_2 = 19kHz	Δf_2 = 7.5kHz	m_2 = 0.395
f_3 = 23kHz	Δf_3 = 16.88kHz	m_3 = 0.734……(2.14)
f_4 = 53kHz	Δf_4 = 16.88kHz	m_4 = 0.318

If the main carrier wave modulation frequency and the modulating factor are known in this manner, the side band spectrum distribution can be calculated as by the method explained previously.

Figure 2.6.4 shows the side-band energy and cumulative values obtained by this method. As the 99% side band energy corresponds to the 0.5% on the vertical axis, it may be determined from this diagram that the 99% band is ±83kHz and falls within the prescribed ±100kHz.

(b) 15kHz Modulation of the difference signal

In stereophonic broadcasts, the side bands widen most not during sum signals, but when maximum deviation is carried out with the 15kHz difference signal. In this instance it will mean that simultaneous modulation will be carried out by 3 frequencies. If the modulation frequency, frequency deviation and modulation factor were listed, they would be as follows.

$f_1 = 19kHz$ $\Delta f_1 = 7.5kHz$ $m_1 = 0.395$

$f_2 = 23kHz$ $\Delta f_2 = 33.75kHz$ $m_2 = 1.467$

$f_3 = 53kHz$ $\Delta f_3 = 33.75kHz$ $m_3 = 0.637$

Results of calculations carried out by the previous method are shown in Figure 2.6.5. It may be noted that the 99% bandwidth just barely falls within the prescribed ±99kHz.

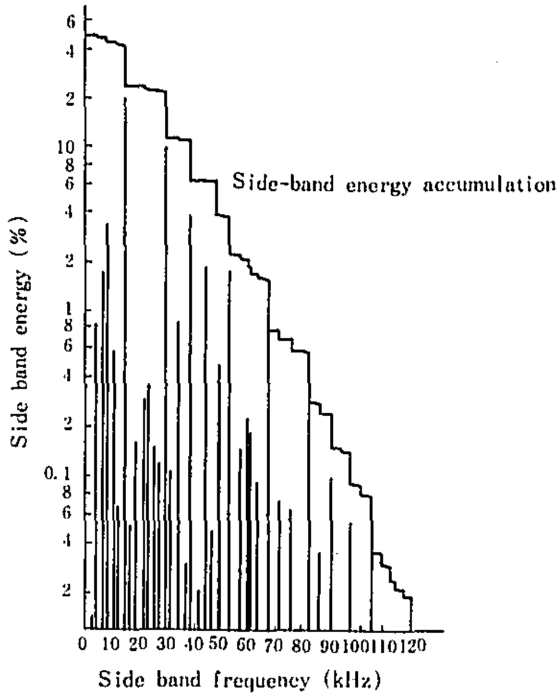


Figure 2.6.4 Side band distribution of 15kHz left side modulation

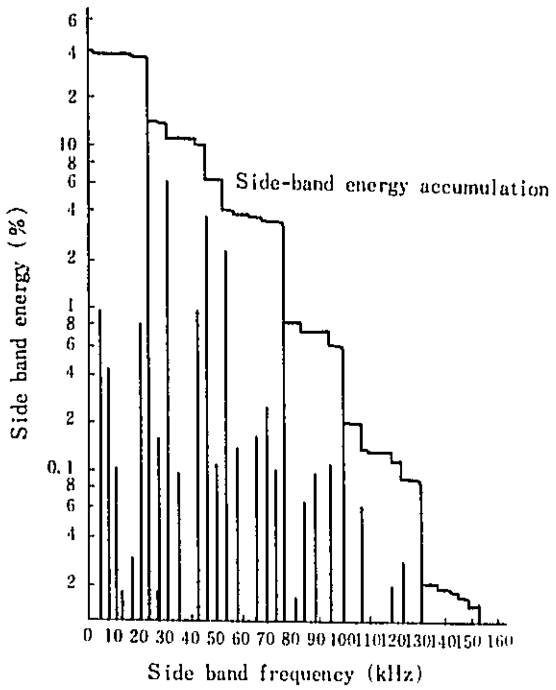


Figure 2.6.5 Side band distribution of 15kHz difference signal modulation

2.7 Signal to Noise Ratio

In article 1.4 we explained that the S/N ratio in FM will be improved by $\sqrt{3}\,mf_0$ over that in AM and in monophonic broadcasts with a maximum frequency deviation of 75kHz and a maximum modulation frequency of 15kHz, it will be 18.8 dB.

Now we shall study about the S/N ratio of stereophonic broadcasts as compared to that in monophonic. Figure 2.7.1 shows the triangular noise distribution of FM detection shown in diagram 5 enlarged to the subchannel. The main channel noise voltage N_M is the same as the N_F derived from equation (15) and, it can be expressed in power, by the following equation.

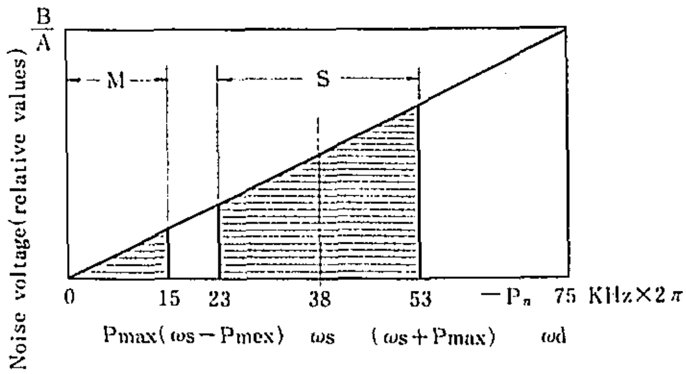


Figure 2.7.1 Noise spectrum of stereophonic broadcasts

$$N_M^2 \equiv N_F^2 = \frac{1}{3} \left(\frac{B}{A}\right)^2 \left(\frac{1}{\omega_d}\right)^2 P_{max}^3$$

Although the noise voltage of the subchannel is distributed from the lower to the upper edge of the side band as shown in Figure 2.7.1, if this is converted to audio frequencies by AM detection, the distribution will be as shown in Figure 2.7.2.

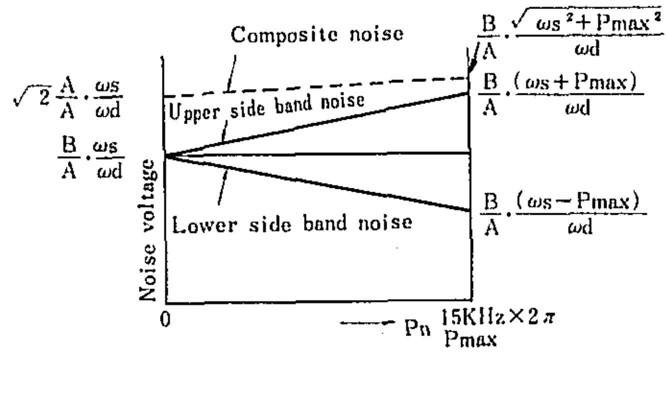


Figure 2.7.2 Subchannel noise spectrum

The noise voltage distribution of the upper side-band $= \frac{B}{A} \cdot \frac{(\omega_s + P_n)}{\omega_d}$ and noise voltage distribution of the lower side band $= \frac{B}{A} \cdot \frac{(\omega_s - P_n)}{\omega_d}$. If we take the square root of the sum of the square of the two noise distribution of composite noise voltage, the subchannel noise voltage distribution will be

$$= \frac{B}{A} \cdot \frac{1}{\omega_d} \sqrt{(\omega_s + P_n)^2 + (\omega_s - P_n)^2}$$

$$\fallingdotseq \sqrt{2} \cdot \frac{B}{A} \cdot \frac{\omega_s}{\omega_d} \quad [P_n^2 \ll \omega_s^2]$$

and will be practically constant within the audio frequency band.

The total noise power N_s^2 (N_s for voltage) of the subchannel may be obtained by integrating the square of the above composite voltage over the audio frequency band.

$$N_s^2 = 2\left(\frac{B}{A}\right)^2 \left(\frac{\omega_s}{\omega_d}\right)^2 \int_0^{P_{max}} dp_n$$

$$= 2\left(\frac{B}{A}\right)^2 \left(\frac{\omega_s}{\omega_d}\right)^2 P_{max}$$

$$= 6 \left(\frac{\omega_s}{P_{max}}\right)^2 N_F^2$$

We will now calculate the noise power ratio between the subchannel and the main channel.

$$\left(\frac{N_s}{N_M}\right)^2 = 6\left(\frac{\omega_s}{P_{max}}\right)^2$$

$$= 38.5 \text{ times} = 15.8\text{dB}$$

The subchannel noise is this much larger. The main channel detection output M' and the subchannel detection output S', also including noise, are expressed by the following equation.

$$M' = m(L + R) + N_M$$
$$S' = m(L - R) + N_S$$

The "m" is the degree of modulation of the main carrier wave, and is 45% as shown in Figure 2.4.1. Then, after entering through a matrix circuit and distributed to left signal terminal and the right signal terminal, the following output will be obtained.

$$L' = M' + S'$$
$$S' = M' - S'$$

Although in the above additions and subtractions, the sound signals may be algebraicly added but as the N_M and N_S noises are completely independent and are moreover irregular, it will be necessary to take the sum of their squares: However, as N_M^2 may be disregarded as compared to N_S^2, if expressed by L', R' outputs they would be as per the following equation.

$$(L')^2 = (2mL)^2 + (N_M^2 + N_S^2)$$
$$\fallingdotseq (2mL)^2 + 6\left(\frac{\omega_s}{P_{max}}\right)^2 N_F^2$$
$$(R')^2 = (2mR)^2 + (N_M^2 + N_S^2)$$
$$\fallingdotseq (2mR)^2 + 6\left(\frac{\omega_s}{P_{max}}\right)^2 N_F^2$$

If L = R here, the S/N ratio of L' (or R') will be

$$\text{Stereo S/N ratio} = \left(\frac{L}{N_F}\right)^2 \frac{(2m)^2}{6\left(\frac{\omega_s}{P_{max}}\right)} \quad \text{(Power ratio)}$$

(Without de-emphasis)

On one hand, the S/N ratio of monophonic broadcasts when modulated with L only will be:

$$\text{Monophonic S/N ratio} = \left(\frac{L}{N_F}\right)^2 \qquad \text{(Power ratio)}$$

(Without de-emphasis)

Therefore

$$\frac{\text{Monophonic S/N ratio}}{\text{Stereophonic S/N ratio}} = \frac{6\left(\frac{\omega_s}{P_{max}}\right)^2}{(2m)^2} \qquad \text{(Power ratio)}$$

(without de-emphasis)

$$= \frac{38.5}{(0.9)^2}$$

$$= (15.8 + 0.9) \text{ dB}$$

$$= 16.7\text{dB}$$

In other words, it is 16.7 dB poorer in stereophonic broadcasts. Of this value, 0.9 dB is caused by using part of the main carrier modulation-degree for transmission of pilot signal, and the rest is caused by the high noise of the sub-channel.

Further, as the received output when stereophonic broadcasts are received in monophonic, receivers output will be only the output of the main channel.

$$(M')^2 = \{m\ (L+R)\}^2 + N_M^2$$

$$= (2mL)^2 + N_F^2 \qquad [\text{When } L = R]$$

Therefore ti becomes

$$\frac{\text{Monophonic S/N ratio}}{\text{Monophonic reception S/N ratio}} = \frac{1}{(2m)^2} \quad \text{(Power ratio)}$$

$$= \frac{1}{(0.9)^2} = 0.9 \text{ dB}$$

and will only be the modulation loss based on transmission of the pilot signal.

The de-emphasis circuit in stereophonic reception normally converts the signal to left and right signals prior to feeding, the results will be same if it was fed into the previous subchannel detector output side. In any event, however, as the noise power contained in the left and right terminal outputs are completely under the influence of the subchannel noise, the effect of de-emphasis on stereophonic output may be considered to be the same as the de-emphasis effect on the subchannel output.

As subchannel output is AM detection, its noise spectrum is practically constant as shown in Figure 2.7.2. The de-emphasis effect in relation to this may be obtained by the following equation.

$$\text{AM de-emphasis effect} = \frac{1}{P_{max}} \int_0^{P_{max}} \frac{1}{1 + (\tau P_n)^2}\, dP_n$$

$$= \frac{\tan^{-1} \tau P_{max}}{\tau P_{max}} = \frac{1}{3.36} \text{ (Power ratio)}$$

$$= -5.3\text{dB}$$

Compared to the de-emphasis effect of 10.2 dB of FM detection obtained in equation (20) this is (10.2 - 5.3) = 4.9 dB poorer. The stereophonic S/N ratio will be as follows when de-emphasis is applied.

$$\frac{\text{Monophonic S/N Ratio}}{\text{Stereophonic S/N Ratio}} = (16.7 + 4.9) \text{ dB}$$

$$\text{(with de-emphasis)} \quad = 21.6 \text{ dB}$$

2.8 Stereophonic Demodulation Circuit

2.8.1 Systems

As the FM detector output, a stereophonic signal (composite signal) identical to the FM modulation signal is obtained. To take out the L and R signals from this signal, the 19kHz pilot signal is extracted and, then, this signal is multiplied or added to a synchronized oscillator, to make a 38kHz sub-carrier wave. This 38kHz sub-carrier is then added to the suppressed AM carrier wave carrying the 'L-R' signal, to convert both side-bands into AM waves. For the detection of this AM wave, there are two methods, one is to detect it to obtain the "L-R" signal and the other is to use the time-division system of demodulation and obtain the L and R signals.

2.8.2 Generation of Subcarrier Wave

The 38kHz sub-carrier wave necessary for the demodulation of L and R signals csn be obtained by extracting the 19kHz pilot signal, out of the composite signal and, multiply it twice. For this procedure, the two-multiplication method and the synchronized oscillation methods are mostly used.

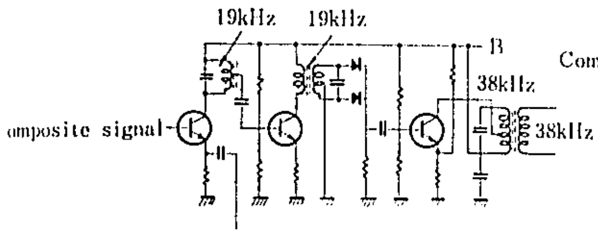


Figure 2.8.1 Subcarrier wave generator using a two-multiplier

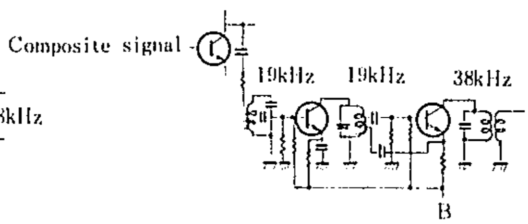


Figure 2.8.2 Subcarrier wave generator using a synchronized oscillator

2.8.3 Method of demodulation using a matrix circuit after detection

An example of the circuit layout is shown in Fig. 2.8.3. Extract the suppressed carrier wave AM signal with a 23 - 53 kHz band-pass filter or a 23 kHz high-pass filter and, add the 38kHz frequency obtained from the subcarrier wave generator to it, to re-convert it to the AM wave.

Next, detect the('L-R') signal and the − ("L − R")signal with a detector multually connected in inverse polarity and, at the matrix circuit, add the main channel("L + R")signal, so it will be separated into audio signals of L and R.

In this method, as the "L - R" signal will have frequency characteristics in regard to amplification and phase, caused in the subcarrier detection circuit, it will be necessary to provide a delay-compensation circuit and a variable amplification circuit to it, before it enters in the matrix circuit, to eliminate the phase and amplification difference of both signals.

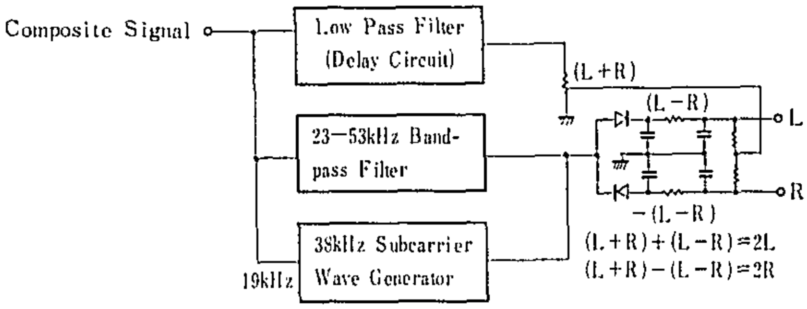


Figure 2.8.3 Method of demodulation using a matrix circuit after detection

2.8.4 Method of Demodulation by Time-Division System (Switching System)

The detection action of the time division demodulation system (Switching System) may be considered as follows. Set the composite signal obtained from the output of the frequency discriminator as

$$A(t) = (L + R) + (L - R) \cos \omega t \qquad (2.15)$$

and by using the pulse signal, which was generated by saturating the 38kHz subcarrier wave as a switching signal, the left side switching signal will be

$$U(t) = \frac{1}{2} + \frac{2}{\pi} \sum_{m=1}^{\infty} \frac{1}{m} \sin \frac{m\pi}{2} \cos m\omega t$$

$$= \frac{1}{2} + \frac{2}{\pi} \cos \omega t - \frac{2}{3\pi} \cos 3\,\omega t + \ldots\ldots \qquad (2.16)$$

The right side switching signal will be

$$U(t+\frac{T}{2})=\frac{1}{2}+\frac{2}{\pi}\sum_{m=1}^{\infty}\frac{(-1)^m}{m}\sin\frac{m\pi}{2}\cos m\,\omega t$$

$$=\frac{1}{2}-\frac{2}{\pi}\cos\omega t-\frac{2}{3\pi}\cos 3\,\omega t+\ldots. \qquad (2.17)$$

Multiply A(t) by left side signal, then L signal will be

$$e_L=\{(L+R)+(L-R)\cos\omega t\}\cdot(\frac{1}{2}+\frac{2}{\pi}\cos\omega t-\frac{2}{3\pi}\cos 3\,\omega t+\ldots)$$

$$=\frac{(L+R)}{2}+\frac{(L-R)}{\pi}+\frac{2}{\pi}(L+R)\cos\omega t+\ldots\ldots \qquad (2.18)$$

The low frequency output will be

$$e_L=\frac{(L+R)}{2}+\frac{(L-R)}{\pi}=\frac{L}{2}(1+\frac{2}{\pi})+\frac{R}{2}(1-\frac{2}{\pi}) \qquad (2.19)$$

and similarly,

$$e_R=\frac{(L+R)}{2}-\frac{(L-R)}{\pi}=\frac{R}{2}(1+\frac{2}{\pi})+\frac{L}{2}(1-\frac{2}{\pi}) \qquad (2.20)$$

As the L and R signals are included in equations, (2.19) and (2.20), the value of separation will be only 13 dB, at the present state. Now, if the equation

$$\frac{L+R}{2}(1-\frac{2}{\pi}) \qquad (2.21)$$

is substracted by using a differential amplifier, it can be separated, to left and right as $2L/\pi$ and $2R/\pi$ respectively.

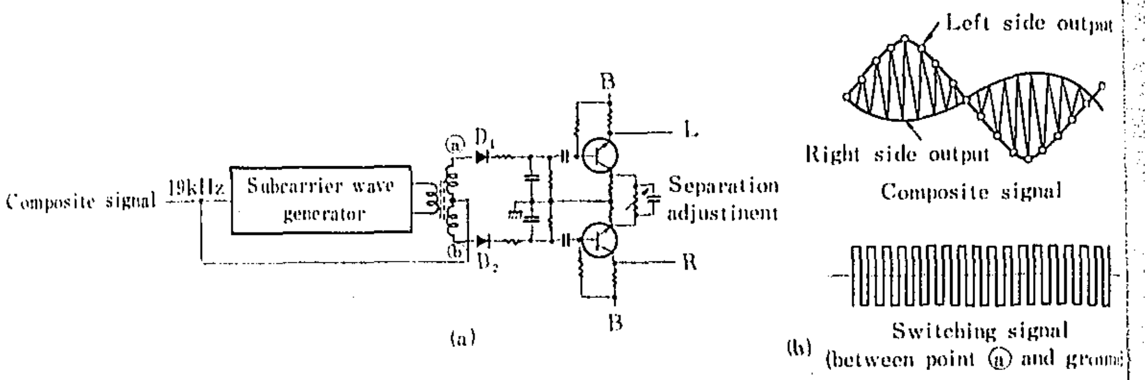


Figure 2.8.4 Principle of demodulation of time-division system

Figure 2.8.4 shows the principle of the time-division system. A diode is used as series-switch in relation to the composite signal and, this switch is driven by the subcarrier wave. The signal is applied to the switching diode D_1 and D_2, together with the 38kHz subcarrier wave. If both are of equal phase, D_1 will be conductive when point ⓐ) of the subcarrier wave is positive and, as shown in the diagram, an L envelope output will be obtained at the L terminal. Next, when point ⓐ is negative, D_2 becomes conductive and an R envelope output will be obtained.

However, as complete separation is not available with the outputs as they are, complete separation is realized by providing a differential amplifier in the detector output circuit to apply cross-talk.

2.8.5 Demodulation Efficiency and Subcarrier Wave Phase-Shift of the Switching System

As shown in equations (2.19) and (2.20), the signals appearing in the L and R output, due to switching becomes 1/2, because the demodulating signal of the (L + R) channel is the average value of the square wave pulses, and becomes $1/\pi$ as the demodulating signal of the (L - R) channel is the average value of the sine waves. This relation is shown in Figure 2.8.5. The separation is poor in this condition, if we substract $\frac{(L+R)}{2}(1-\frac{2}{\pi})$, that is, if we provide separation adjustments, complete separation will be possible.

If we consider this in relation to the flow-angle and separation of the switching frequency (subcarrier wave), and if it is

$$e_L = \frac{1}{k}(L+R) + \frac{1}{k}(L-R)$$

$$e_R = \frac{1}{k}(L+R) - \frac{1}{k}(L-R) \qquad (2.22)$$

from equations (2.19) and (2.20), then, with $e_L = \frac{2L}{k}$ and $e_R = \frac{2R}{k}$, crosstalk would originate because of the difference in demodulation efficiency in relation to (L + R) and (L - R) $\sin\omega t$.

As demodulation efficiency of the switching system will be

$$\eta_M = \frac{\theta}{2} \ (\theta\text{:rad}) \qquad (2.23)$$

in the main channel, when the switching flow angle is θ, and

$$\eta_s = \frac{1}{\pi}\sin\frac{\theta}{2} \qquad (2.24)$$

in the subchannel, crosstalk may be reduced if the two were brought as close together as possible. In other words, this difference may be reduced by reducing the flow angle.

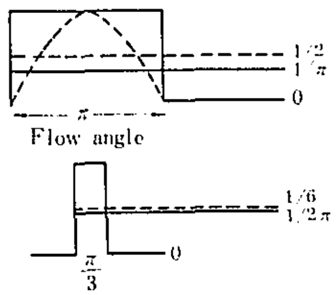


Figure 2.8.5 Flow angle and demodulation efficiency of the switching frequency (38kHz)

Normally, the separation of this system is calculated by the following equation.

$$20\log \frac{1+m}{1-m} \text{ (dB)} \qquad (2.25)$$

$$m = \frac{\theta}{2 \sin \theta/2}$$

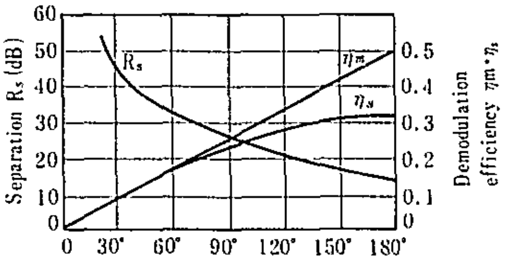


Figure 2.8.6 Relations between the switching flow angle, demodulation efficiency and separation

Although this is in case the phase of the subcarrier wave matches when phase shift occurs, the demodulation efficiency of the subchannel will drop by

$$\eta_s = \frac{1}{\pi} \sin \frac{\theta}{2} \cdot \cos \phi \qquad (2.26)$$

ϕ : Subcarrier wave phase shift

Separation will therefore also drop by

$$20 \log \left| \frac{\cos\phi + m}{\cos\phi - m} \right| \text{ (dB)} \qquad (2.27)$$

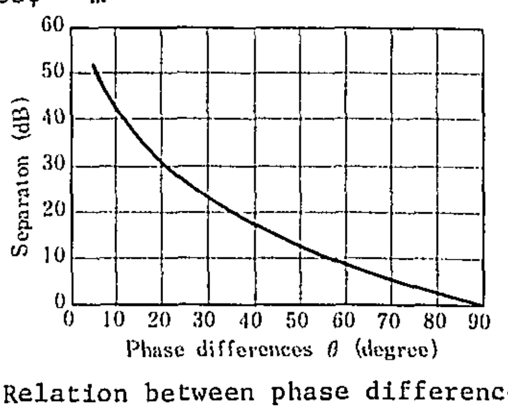


Figure 2.8.7 Relation between phase difference and separation

3. FM TRANSMISSION EQUIPMENT

3.1 Standard Systems of Transmission

(1) Monophonic Broadcasting Transmission Systems

1) Modulation system of main carrier wave ... Frequency Modulation

2) Maximum frequency deviation of main carrier wave .... ±75kHz

3) Maximum frequency of sound signal 15kHz

4) Pre-emphasis characteristics 50μs

(2) Stereophonic Broadcasting Transmission Systems

Refer 2.4.1.

3.2 Technical Standards of Transmission Equipment

(1) Common Standards for Monophonic and Stereo Broadcasts

1) Tolerance of main carrier frequency...... 20/1,000,000

2) Tolerance of band-width occupancy .... 200 kHz

3) Allowable strength of spurious radiation

The average spurious radiation power for each frequency supplied to the feeder, shall be less than 1 mW and shall be 60 dB below the average power of the fundamental frequency.

4) Tolerable antenna power deviation

Upper limit 10%

Lower limit 20%

5) Overall frequency characteristics

It is specified that the modulation frequency between 50Hz and 15,000Hz shall lie between the ideal pre-emphasis characteristics curve and, the allowable limit curve of the pre-emphasis characteristics (or on the curve line) of the 50μs time-constant curve in Figure 3.2.1. Further, in case of inserting a program transmission control signal, this will be separately defined by the Ministry of Posts and Communications.

6) Linearity .... The modulation characteristic to be linear up to 100%

7) Overall distortion factor

When a ±75kHz frequency deviation is applied to the main carrier wave and a de-emphasis of impedance frequency characteristics of the 50μs time-constant is applied, the distortion shall be under 2% for modulation frequencies above 50Hz but below 10,000Hz and, under 3% for modulation frequencies above 10,000Hz but below 15,000Hz.

8) Signal to Noise ratio

When a ±75kHz frequency deviation is applied to the main carrier by means of a modulation frequency of 1,000Hz, the S/N ratio shall be better than 55dB. However, de-emphasis is to be provided.

9) Residual amplitude modulation noise

The noise shall be under (-) 50 dB, in comparison with that of the main carrier transmitter output at 100% amplitude modulation, with de-emphasis provided.

10) Polarization of wave

It is ruled that the polarization of the waves radiated from the transmitting antenna must be horizontal. However, vertical polarization or other polarization may be used when approved by the Minister of Posts and Communications. In case an FM station and a VHF TV station are co-located, a vertical polarization may be approved for the FM station if the TV station is of vertical polarization, considering the common usage of the transmitting and receiving antennas.

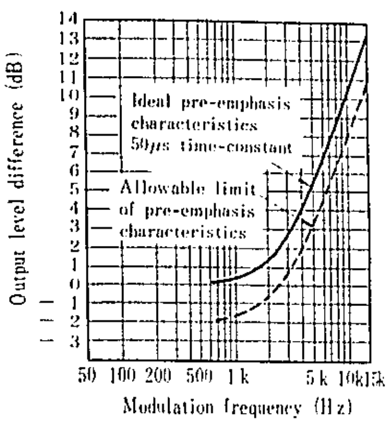


Figure 3.2.1 Pre-emphasis Characteristics

(2) Standards for Stereophonic Broadcasts only.
Refer 2.4.2.

3.3 Composition and Layout of Broadcasting Stations

As for FM transmitter stations employing the modulation system, we have stations with equipment capable of transmitting its own monophonic and stereophonic programmes and stations which are mostly receiving their master station programmes for rebroadcast and, stations which could only transmit its own monophonic programmes for local service. The composition and layout of these stations are shown in Figure 3.3.2.

For the way of connecting the studio site to the transmitter site, land cables and STLs are used. But in case the distance between the studio and transmitter site is short, cables can be used, but when the distance is long, STLs are used because of deterioration in land cable characteristic will occur.

In case monophonic programmes are only to be broadcast, stereo-modulators will be unnecessary but in case stereophonic programmes are to be broadcast, a stereo-modulator will be necessary at the transmission terminal. In case the STLs are to be used, stereo-modulators will be equipped at the studio site in the stage of the STL. When cables are to be used, the stereo-modulators will be inserted in the FM transmitter first stage.

In case a FM station is not required to transmit its own programmes, and is always receiving its master station's programmes off - air for rebroadcast, a transmitter with a modulator will be unnecessary. In this case, the programme signal received from master station or other fixed station will be directly converted to another frequency and re-transmitted. For this purpose, a specially designed translator system is mostly used. A block diagram of these are shown in Fig. 3.3.3.

As for the programme signal for the FM rebroadcasting stations to receive, the VHF broadcast wave and the fixed station wave mentioned above could be used. In general, the VHF waves are used, but in case the field intensity of the VHF wave is weak or it is interfered by other radio services or has distortion due to multipath propagation, insufficient for broadcast quality, a relay unit is located at an appropriate place between the master station and the FM broadcasting station. We call this a relay unit.

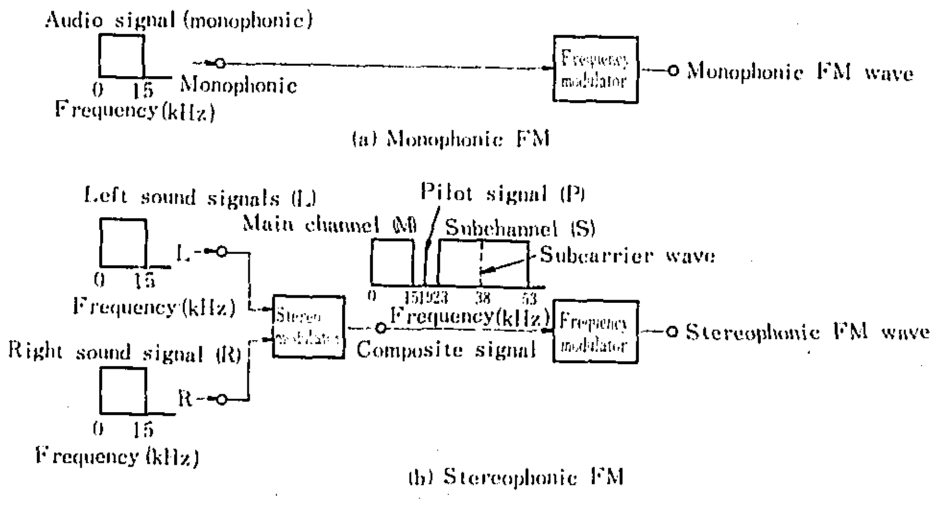
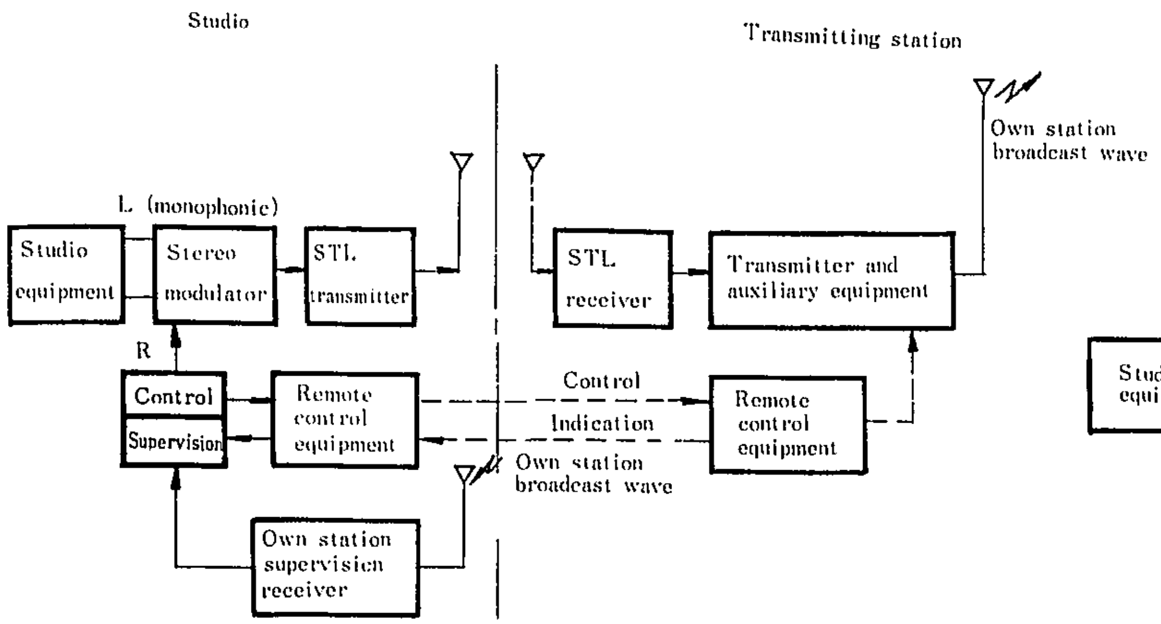
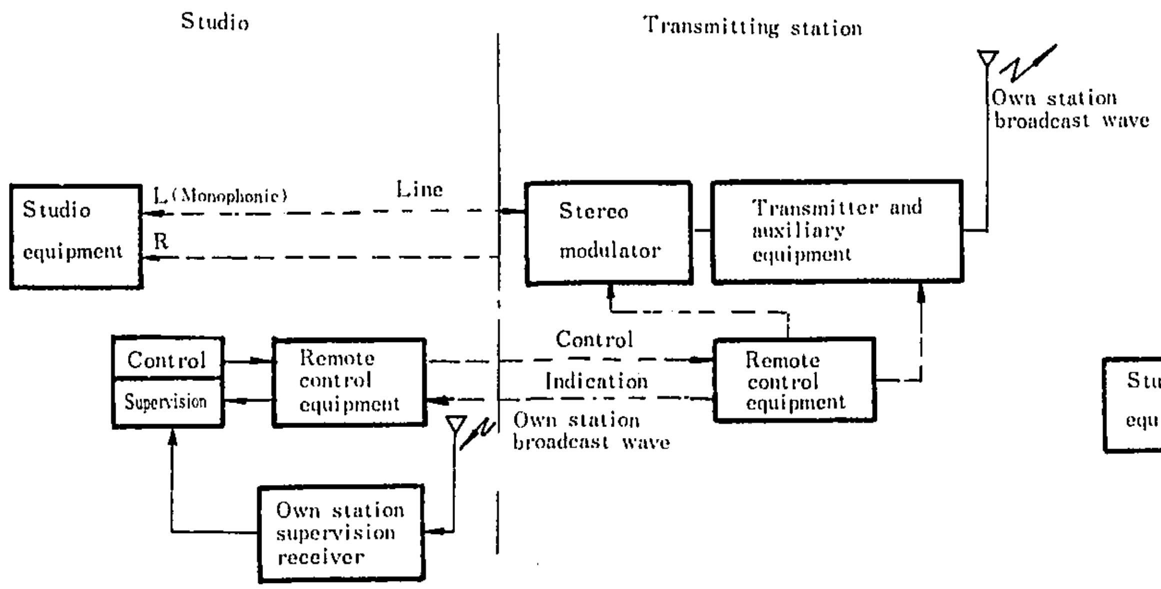


Figure 3.3.1 Monophonic and Stereophonic FM

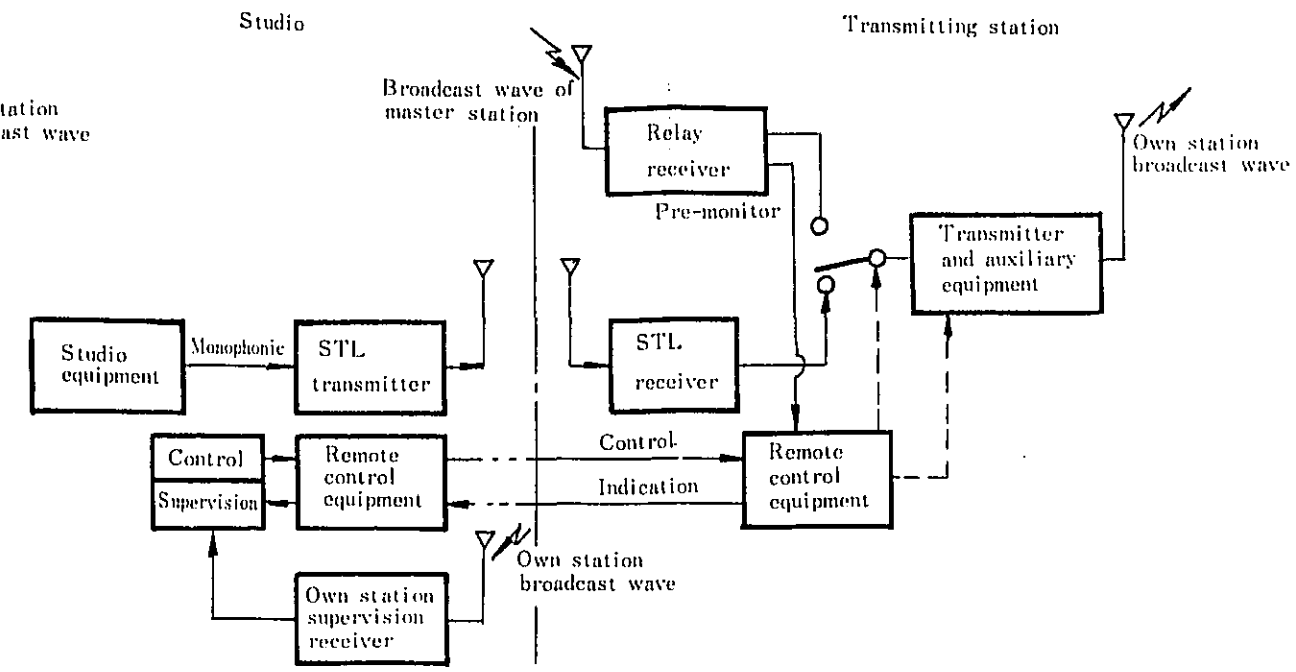


(a) When program transmission between studio and transmitting station is provided by STL

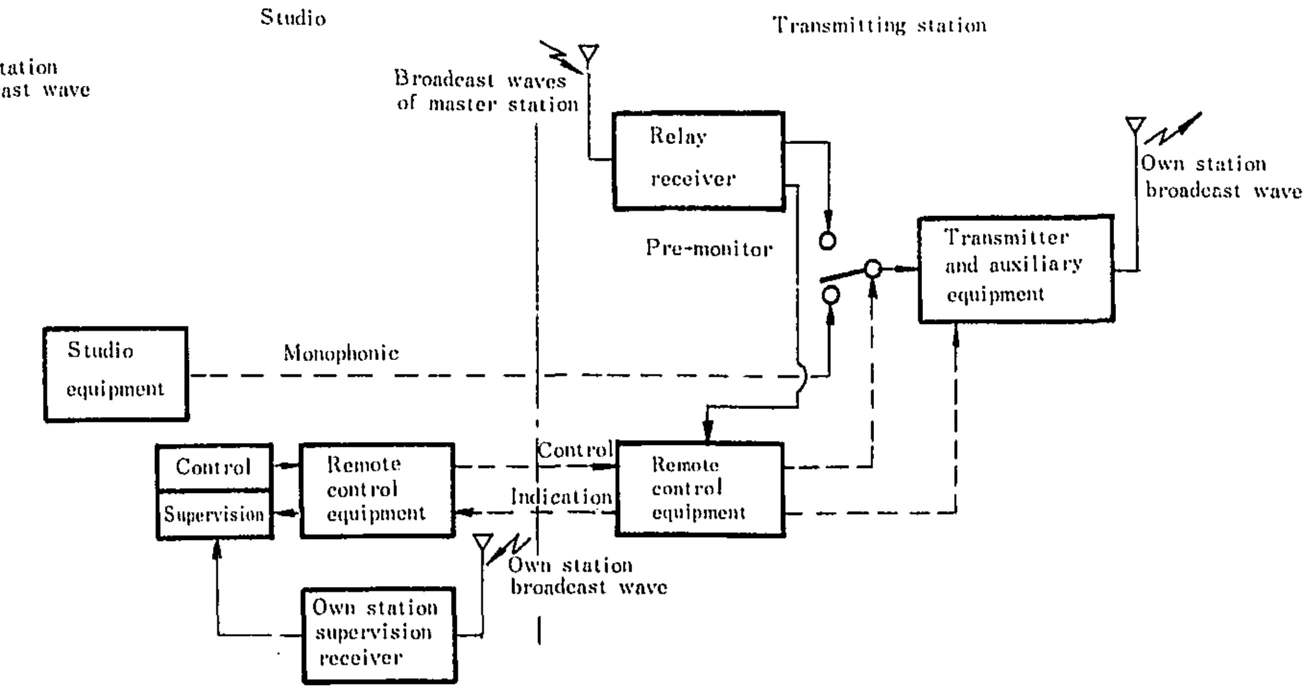


(b) When program transmission between studio and transmitting station is provided by cable

Composition and layout of regional master station



(a) When program transmission between studio and transmitting station is provided by STL



(b) When program transmission between studio and transmitting station is provided by cable

Composition and layout of general broadcast stations

Figure 3.3.2 Composition and Layout of an FM Broadcasting Station

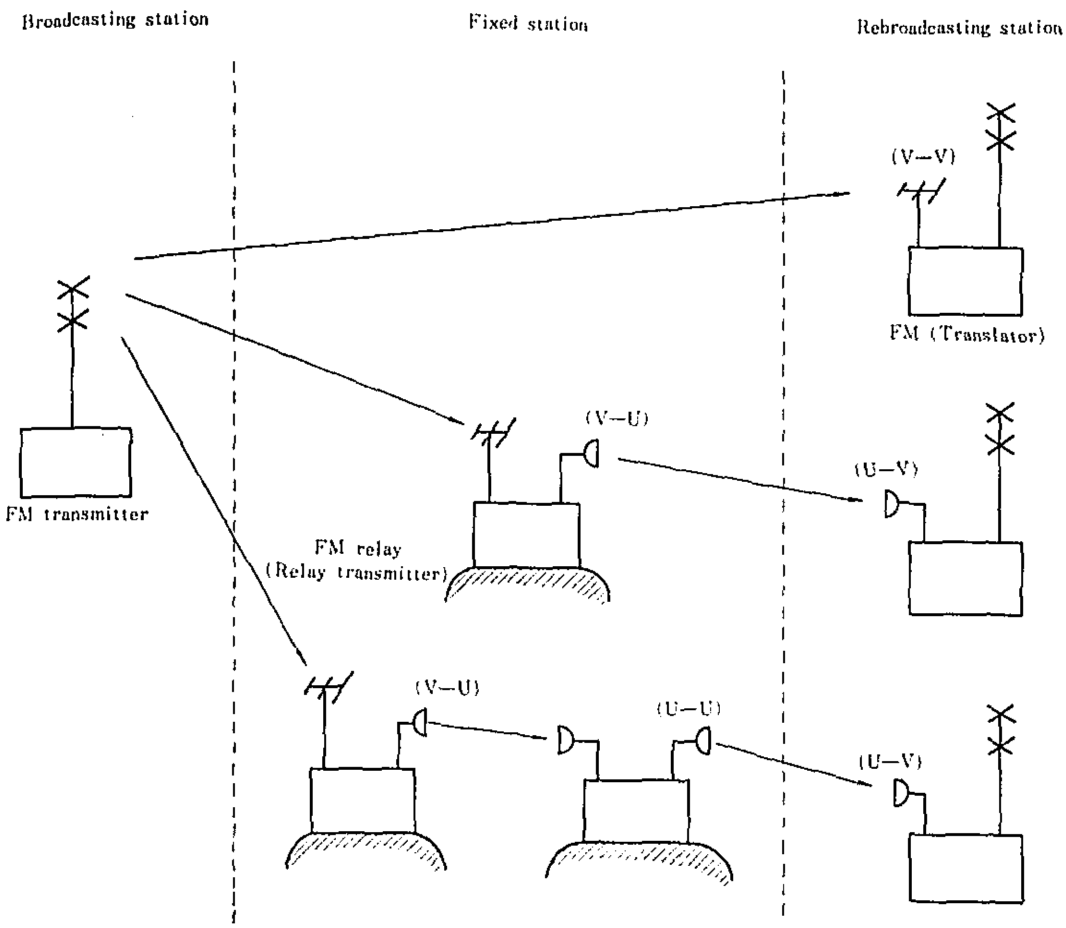


Figure 3.3.3 Off-air relay system

3.4 FM Transmitter

3.4.1 Methods of Modulation

The following are representative methods of modulation.

(1) Direct frequency modulation

(i) System of adding a variable reactance element to oscillator

(ii) Reactance tube system

(2) Indirect frequency modulation

(i) Serrasoid system

(ii) Vector Synthesis system

The advantages of the modulation system (1) are simplicity in circuitry, availability of wider modulation bandwidth and ability of modulating directly with composite signals. As the sensitivity of modulation is good, the number of multiplication could be less, compared to other systems, which results in reducing spurious radiation. Also, in the case of stereophonic broadcasting, where the studio and transmitter site is separated, only one line will be necessary for transmission of the composite signals.

However, the disadvantages are that as it employs a self-excited oscillator of poor stability, a frequency control circuit to suppress the fluctuation will be needed.

The advantage of the modulation system (2) is superior stability of frequency due to the use of crystal oscillators. However, there are disadvantages such as limitation on modulation sensitivity from distortion and other characteristics, and requires a large number of multiplication stages to obtain the desired frequency deviation. Spurious radiations are therefore easily generated and difficulty will be experienced in securing a wide modulation frequency bandwidth. Further, in stereophonic FM broadcasting stations where the studio and transmitting station is separated, two lines for the left and right signals will be required.

A stereo FM broadcasting transmitter composed of a combination of stereo modulators necessary for the transmission of stereophonic programs from own station is shown in Figure 3.4.1.

At present, the system in (i) of (1) is used exclusively for FM broadcasting transmitters.

3.4.2 Stereo Modulator

(1) Composition of a Stereo Modulator

In the block diagram in Figure 3.4.2, the L and R signals each enter the input regulations amplifier where a pre-emphasis of a time-constant of 50μs is applied, and the left and right signal levels are equalized. This output is then divided into (L + R) and (L - R) signals at the matrix circuit. As observed in Figure 3.4.3, the matrix circuit halves the left and right signals by a transformer and derives an (L + R) signal by adding R of equal phase to L and an (L - R) signal by adding R of inverse phase to L.

The (L + R) signal becomes the output of the M channel, after passing through the delay compensation circuit and, the amplitude regulator for adjustment of separation. The reason for this signal to pass through the delay circuit is that because the (L - R) signal will be delayed when passing through the sub-modulator circuit, the (L + R) signal must also be delayed to an equivalent amount.

The (L - R) signal enters the ring modulator, where it modulates the 38kHz subcarrier wave conducted from the crystal oscillator, and becomes an "S" subchannel signal of only both side-bands with carrier suppressed. The operation of the ring modulator, as shown in Fig. 3.4.4 (a), is such as that when the subcarrier wave becomes sufficient large, the rectifier will become open-circuit or short-circuit against the L - R signal and therefore, as shown in the equivalent circuit of (b), it will activate as a switch to switchover the polarization. Fig. (c) to (f) are the voltage wave forms for each portion. If the rectifier is balanced, the carrier will be completely eliminated.

To form the 19kHz pilot signal, which is half of the 38kHz subcarrier wave frequency, a portion of the subcarrier wave oscillator output is fed into the flip-flop circuit. The output of this circuit is fed to the phase and amplitude regulators and then amplified and becomes the pilot signal. Further, an auxiliary 19kHz signal which is used for making phase adjustment of the pilot signal and subcarrier wave, by means of a synchroscope is connected to the check terminal.

In monophonic program transmissions, the program is transmitted through the L side channel and the matrix circuit is bypassed by means of a changeover switch. The subchannel and pilot signal circuits are disconnected.

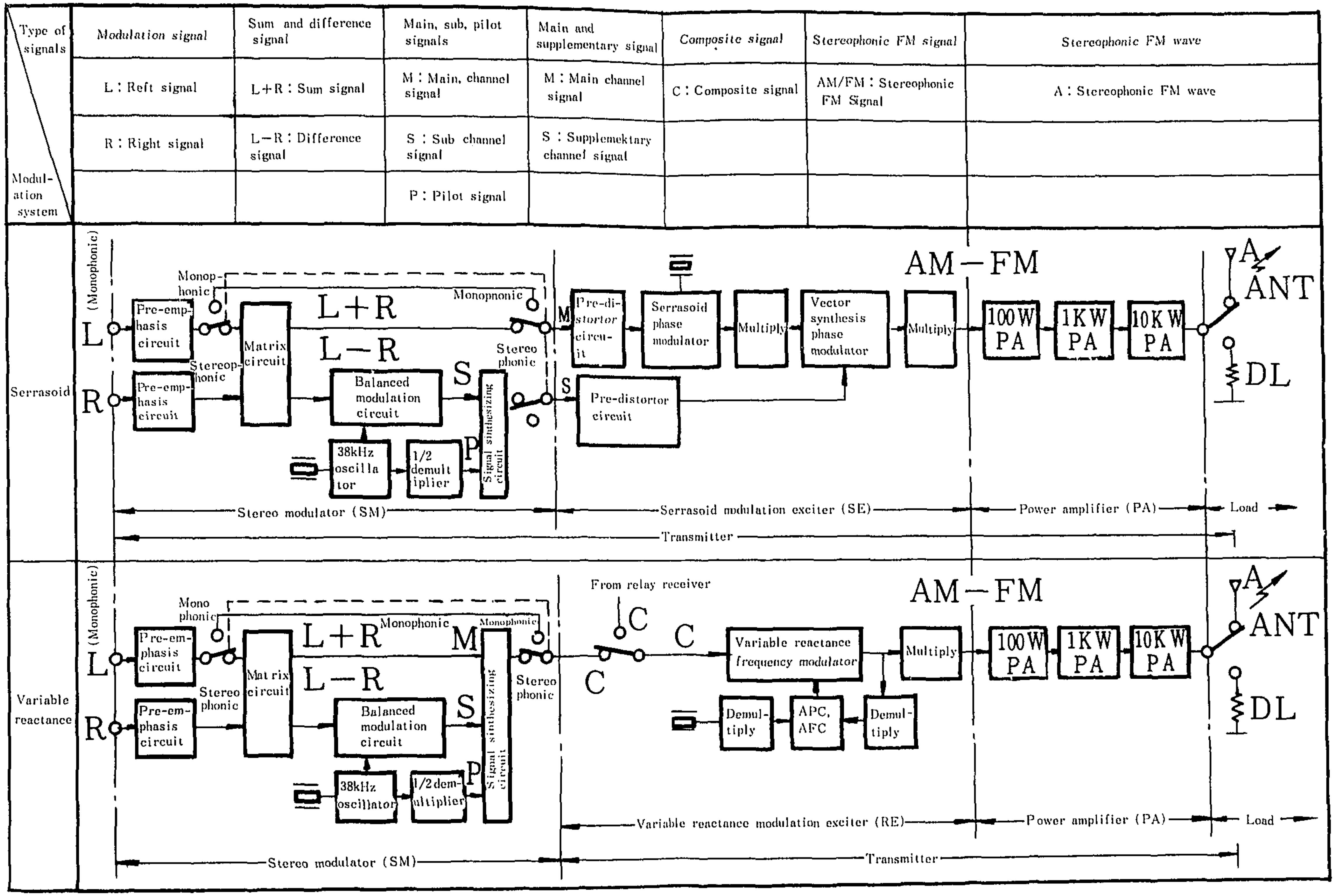


Figure 3.4.1 Block diagram of a stereophonic FM broadcasting transmitter

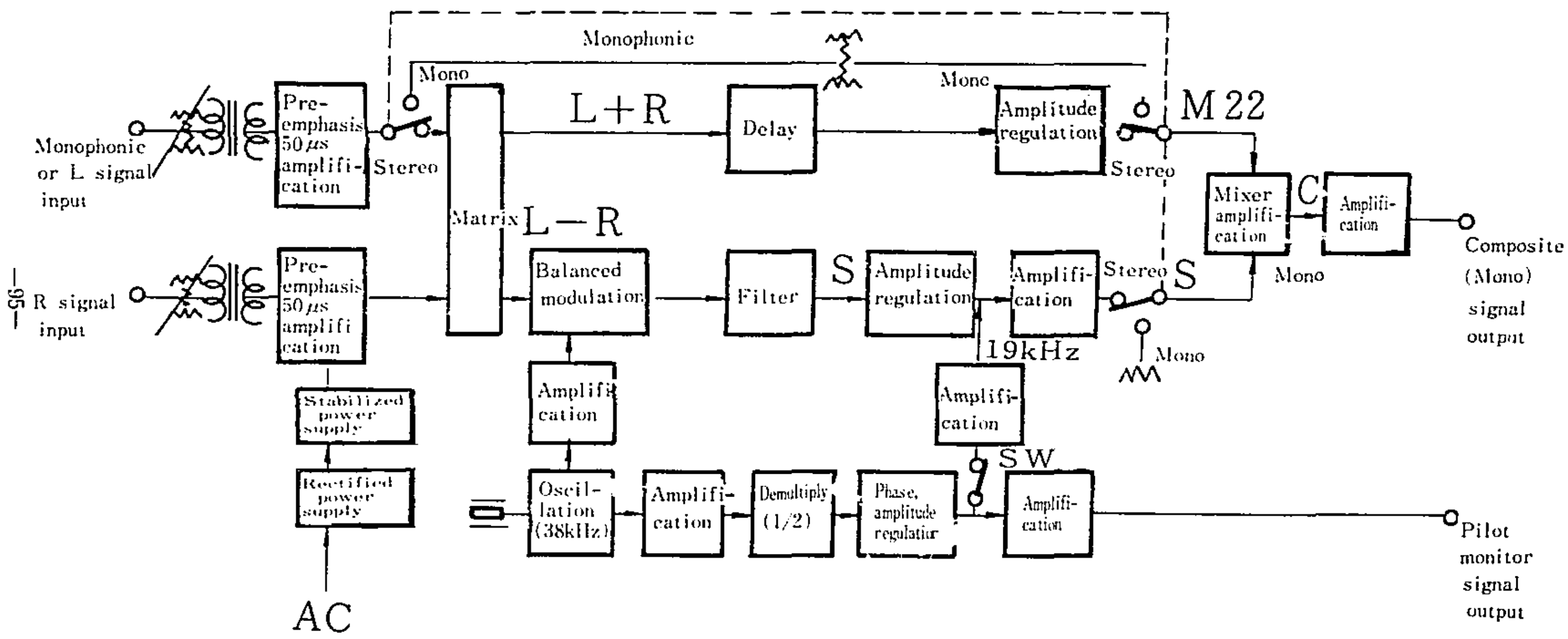


Figure 3.4.2 Block Diagram of a Stereo Modulator

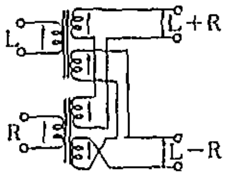


Figure 3.4.3 Matrix Circuit

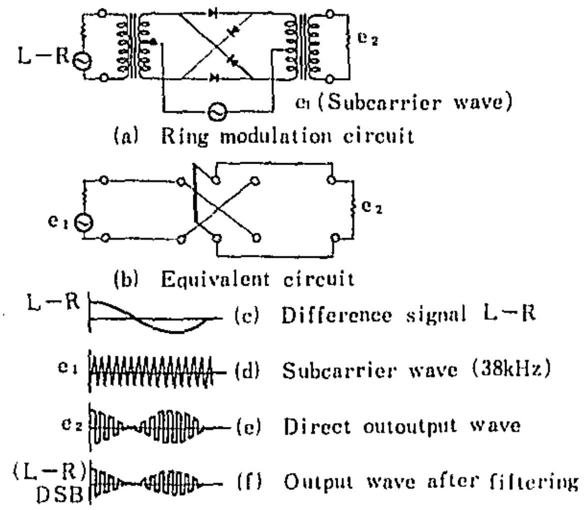


Figure 3.4.4 Balanced Modulation Circuit

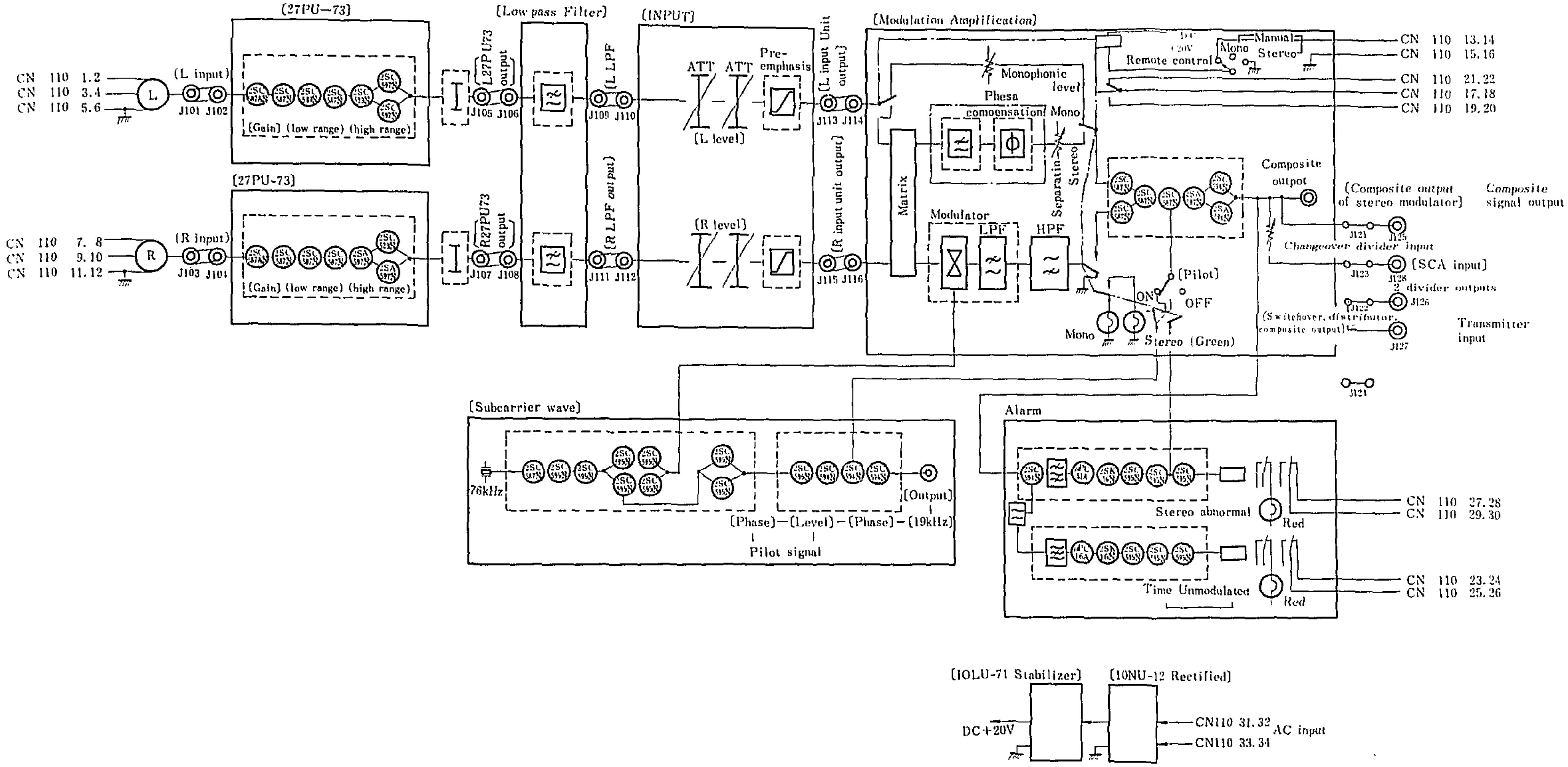


Figure 3.4.5 Block Diagram of a Stereo Modulator

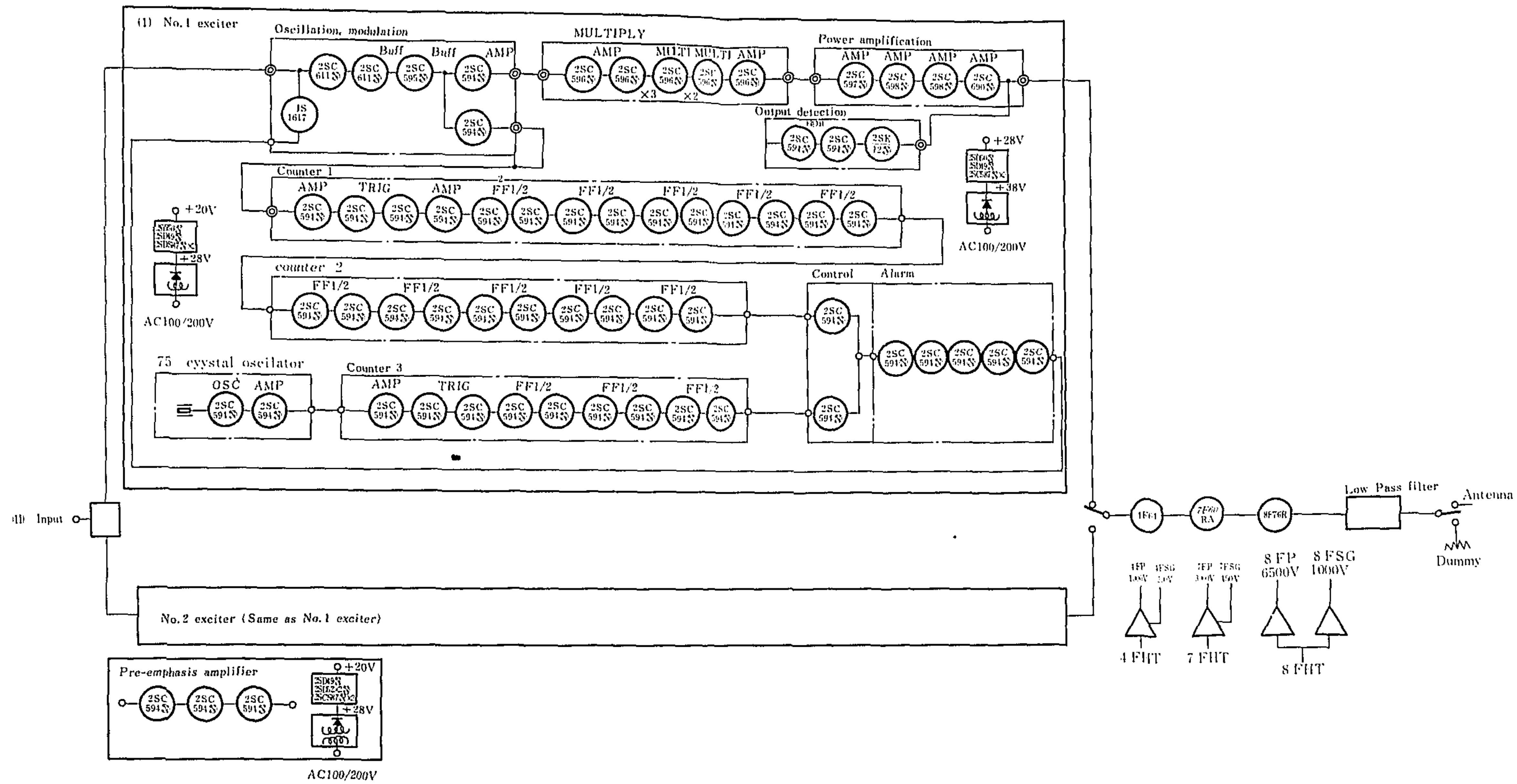


Figure 3.4.6 Block Diagram of a 10kW FM Broadcasting Transmitter

(2) Type ST-71 Stereo Modulator

This unit is completely transistorized and the principal performances are shown in Table 3.1. The modulated stereophonic signal can be taken out from the final output and, provide alarm for the following.

1) 'Stereo Abnormal' when pilot signal is off during stereophonic operation.
2) 'Non-modulation' when output signal is off during operation.

3.4.3 FM Transmitter

(1) Composition

An example of the composition of a 10kW FM broadcasting transmitter is shown in Figure 3.4.6.

(2) Reactance modulation exciter

The composition and system of a reactance modulation excite, using a variable reactance element is shown in Figure 3.4.6. The monophonic or composite signals entering the input is stepped up from an impedance of 75Ω to 300Ω and the level is adjusted by means of a variable resistance attenuator, and then fed to the oscillation modulator unit.

The oscillation modulator unit is self-excited at a frequency 1/6 of the transmission frequency and, as shown in Fig. 3.4.7, the capacitance of the variable capacitor diode is varied by the input signal and, provides frequency modulation with a frequency deviation of ±12.5 kHz.

The relation between the inverse voltage and capacity of the variable capacitance diode is shown in Figure 3.4.8.

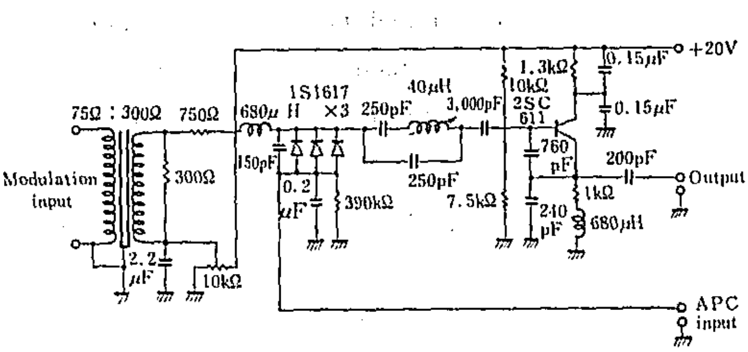


Figure 3.4.7 Modulation Circuit with Variable Capacitance Diode

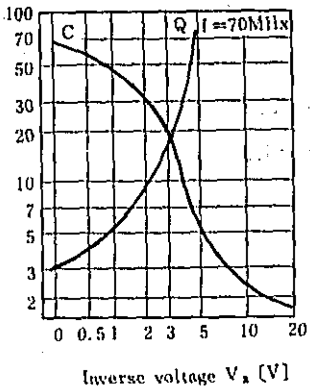


Figure 3.4.8 Characteristics Curves (Ta= 25°C) 1S1617 - 1619 C-V, Q-V Characteristics

The output frequency of the oscillation modulator unit is multiplied 6 times, to meet the transmission frequency, with an output power of about 20 mW.

This output is fed into the power amplifier unit and amplified up to 10W to excite the next stage. The frequency control output from the oscillation modulator unit is stepped down 1/32 by Counter Unit No. 1 and further stepped down 1/32 by Counter Unit No. 2, and totally become 1/1024 of the oscillation frequency or approximately 14kHz.

An approximate 110kHz oscillation is derived from the standard crystal oscillator and this is stepped down 1/8 in Counter Unit No. 3 to approximately 14kHz.

As shown in Figure 3.4.9, the outputs of Counters 2 and 3 are compared in the control unit and a DC component proportional to the phase difference is extracted. After thorough smoothing of the wave form, it is fed back to the variable capacitance diode of the oscillation modulator unit, to compensate the frequency drift.

When the voltage vectors of the standard and comparison signal outputs V and V have a phase difference of 90°, as shown in Figure 3.4.9, the detected DC component will be zero as vectors $(V_1 + V_2)$ and $(V_2 - V_1)$ are equal as shown in Figure 3.4.10.

Next, when no phase difference exists between vectors V_1 and V_2, the sizes of vectors $(V_1 + V_2)$ and $(V_2 - V_1)$ will differ as in (c) and a DC output in proportion to this difference will appear.

Although (d) will be the same as in (c), since the direction of vector is inversed to (c), a DC output with an opposite polarity to (c) will be obtained.

Generally, a DC output as (b), corresponding to the phase difference between V_1 and V_2 will be generated. From the foregoing relations, the phase detector has a phase and a "s" voltage curve as shown in Figure 3.4.11.

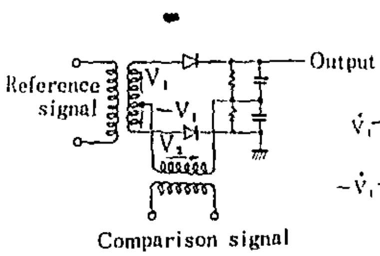


Figure 3.4.9
Principle of a
Phase Detector

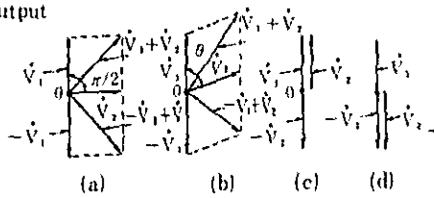


Figure 3.4.10
Vectors of the
Various Voltage

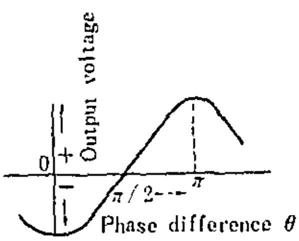


Figure 3.4.11
Characteristics
of a Phase
Detector

(3) Power Amplifier

The output power of FM broadcasting transmitters are in general, over 250 Watts. The power of transmitters are classified into 10 dB steps, such as 1 kW, 10 kW, etc. Stations assigned with output power of 250 or 500 W can use a 1 kW transmitter, by reducing the output power. The 10 kW transmitters normally use a 1 kW transmitter with an additional 10 kW power stage. In case a power of 3 kW or 5 kW is required, the 10 kW transmitter can be operated by reducing the output power, as mentioned above.

There are many advantages such as in means of manufacture, maintenance and increase of power by employing the foregoing method. Figure 3.4.12 shows an example of a high-pass filter to insert in the output of a power amplifier.

Working attenuation standards

2nd Harmonics	over 50 dB
3rd Harmonics	over 40 dB
4th Harmonics	over 30 dB
5th Harmonics	over 20 dB

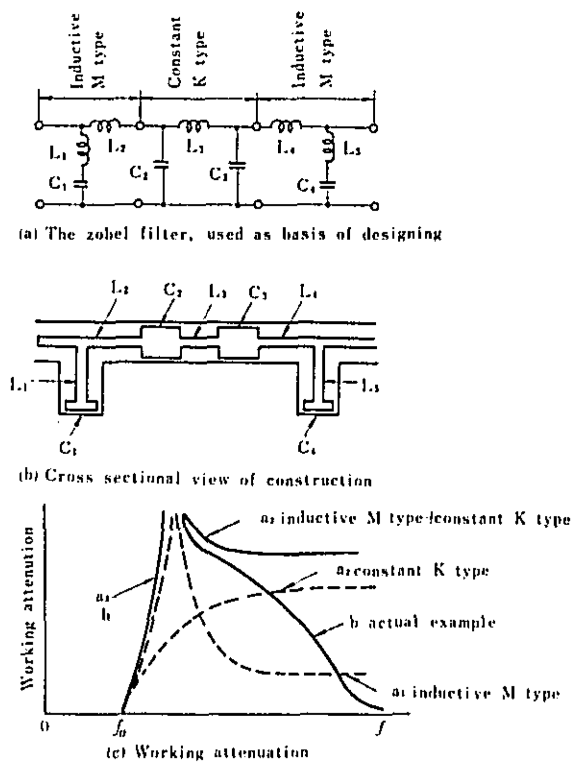


Figure 3.4.12 High Frequency Filter

3.4.4 Principal Standards

Table 3.1 List of principal standards for FM broadcasting transmitters

Items \ Classification	Standard system - equipment regulations etc.	BSS: Broadcasting transmitter	BSS: Stereo modulator	BSS: Radio relay receiver	BSS: Monitor receiver
Transmission frequency range	76 - 90MHz	Class 1 within 76 - 84 MHz or 82 - 90MHz	-	-	-
Tolerance of transmitting frequency	Within $\pm 20 \times 10^{-6}$	Within 1KHz	-	-	-
Input frequency range	- - -	Monophonic (50∿15,000Hz) (50∿53,000Hz)	(50∿15,000Hz)	Class I within 76-90MHz	-
Standard input level	-	(A)∿(D): $1V_{pp}$	(A)∿(D): +10dBm	55-80dBμ (Open)	70-80dBμ (Open)
Input impedance	-	75Ω	600Ω (balanced)	50Ω	*
Output impedance	-	50Ω	75Ω	*	600Ω (balanced)
Maximum frequency deviation	±75KHz	*	-	-	-
Subcarrier wave frequency	38KHz	-	38KHz	-	-
Allowable variation in the pilot signal frequency	±2Hz	-	±1.5Hz	-	-
Pilot signal phase	0°	-	0°	-	-
Allowable variation of the pilot signal phase	±5°	-	±3°	-	-
Modulation of the pilot signal	10%	-	10%	-	-
Allowable degree of modulation of the pilot signal	8∿10%	-	8∿10%	-	-
Allowable value of modulation by the residual subcarrier wave	1%	-	0.7%	-	-
Allowable variation in antenna power	+10% ∿ -20%	*	-	-	-
Allowable strength of spurious radiation	To be under 1 mW and moreover under -60dB. To be less than 1mW when transmission power is under 1W.	To be under 0.5mW and moreover under -63dB in relation to the standard and working output. To be under -43dB when standard and working output is under 1W.	-	-	-
Permissible value of residual amplitude modulation	(Unmodulated) -50dB	Unmodulated: -53dB, 1,000Hz(D): -40dB	-	-	-
Occupied frequency bandwidth	200KHz	15KHz(A),(D): 200kHz	-	-	-
Emphasis	Pre-emphasis 50μs	(*)	*	De-emphasis 50μs	*
Permissible limits of the overall frequency characteristics	0 -2dB -3dB 100Hz 10kHz 50Hz 15kHz	With 1,000Hz as the standard, 50∿15,000Hz: ±1dB	With 1,000Hz as the standard, 50∿15,000Hz: ±0.5dB	*	*
Permissible value of L and R separation	100∿10,000Hz: 30dB	100∿10,000Hz(A): 36dB	*	100∿10,000Hz(A) 33dB	100∿10,000Hz(A) 30dB
Permissible value of L and R levels	100∿10,000Hz: 1.5dB		1,000Hz: 0.2dB	-	1,000Hz: 0.2dB
Permissible value of the overall distortion factor	From 50∿10,000Hz:2% 10-15kHz:3%	50∿10,000Hz (A)-(D): 1%.2% with a 2.5dB larger input Over 10kHz but under 15kHz (A)-(D):1.5% 3% if 2.5dB larger input	50∿15,000Hz(A)-(D): 1%. 2% if input 2.5dB larger	50∿15,000Hz(A)-(D): 1% 50-10,000Hz (A)-(D):1% 2% if input 2.5dB larger Over 10kHz but under 15,000kHz(A)-(D): 1.5%. 3% if input 2.5dB larger	
Permissible value of S/N ratio	1,000Hz (A),(D): 55dB	1,000Hz(A),(D):60dB However, hum element 65dB	1,000Hz(A),(D): 65dB	1,000Hz(A),(D): 60dB	*
Drawing range of the detuning frequency	-	Over ±240kHz	-	-	-
Detuning frequency retention range	-	Over ±360kHz	-	-	-
Effective selectivity				Local station: ±600kHz: Under -40dB, ±700kHz: Under -50dB ±80kHz: Under -60dB Over ±800kHz: Under -60dB with deteriorating characteristics. Other stations: ±500kHz: Under -50dB, over ±500kHz: under -50dB with deteriorating characteristics.	

Note 1.

(A), (B), (C) and (D) are as per the Table at right

	Description	Modulation signal	Modulation signal	Sub channel modulation	Pilot signal modulation
(A)	Maximum percentage of modulation	With L or R signal only	45%	45%	10%
(B)	Maximum percentage of modulation of the main channel	L and R signals of equal phase and equal amplitude	90%	0%	10%
(C)	Maximum percentage of modulation of the sub-channel	L and R signals with reversed phase and equal amplitude	0%	90%	10%
(D)	Maximum percentage of monophonic modulation	Monophonic signal	100%	0%	0%

Note 2.

*Symbols: Same as at left for the same item

Note 3.

Standard system: Standard transmission systems for VHF Broadcasts (Ministry of Posts and Communications Ordinance 26, July 1, 1968),
Equipment Regulations: Radio Equipment Regulations (Ministry of Posts and Communication Ordinance 32, August 20, 1968).

3.4.5 Auxiliary Facilities

(1) Radio Relay Receiver

Receives the broadcast wave of the master station and extracts the composite stereophonic signals, to modulate its own station transmitter for rebroadcasting.

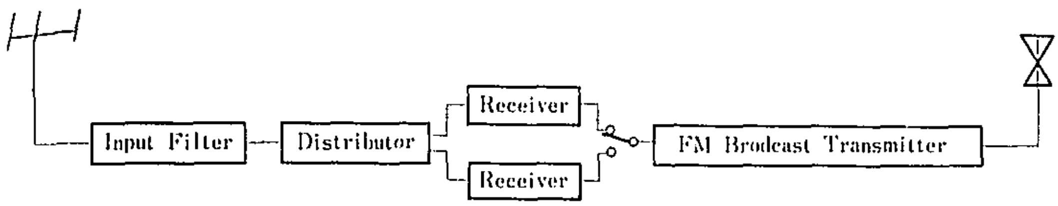


Figure 3.4.13

It should be considered so that a sufficient DU ratio could be obtained against its own transmitting wave. If the DU ratio is small, abnormal oscillation may occur between the transmission and reception waves. Therefore, it will be necessary to take consideration of the DU ratio, in case the difference in frequencies is small, especially when there is fading. Measures for these are to separate the transmitting and receiving point, or to adopt diversity reception, to avoid the interference from its own transmitting wave. It is also necessary to make studies against distortion caused from multi-propagation paths. Figure 3.4.14 shows a block diagram of a typical radio relay receiver.

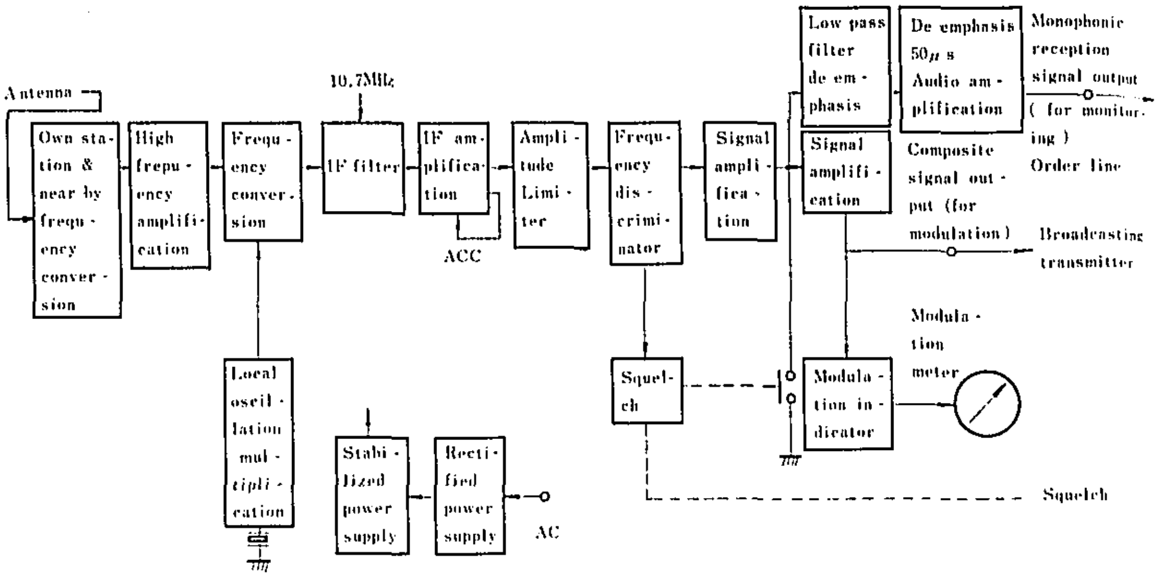
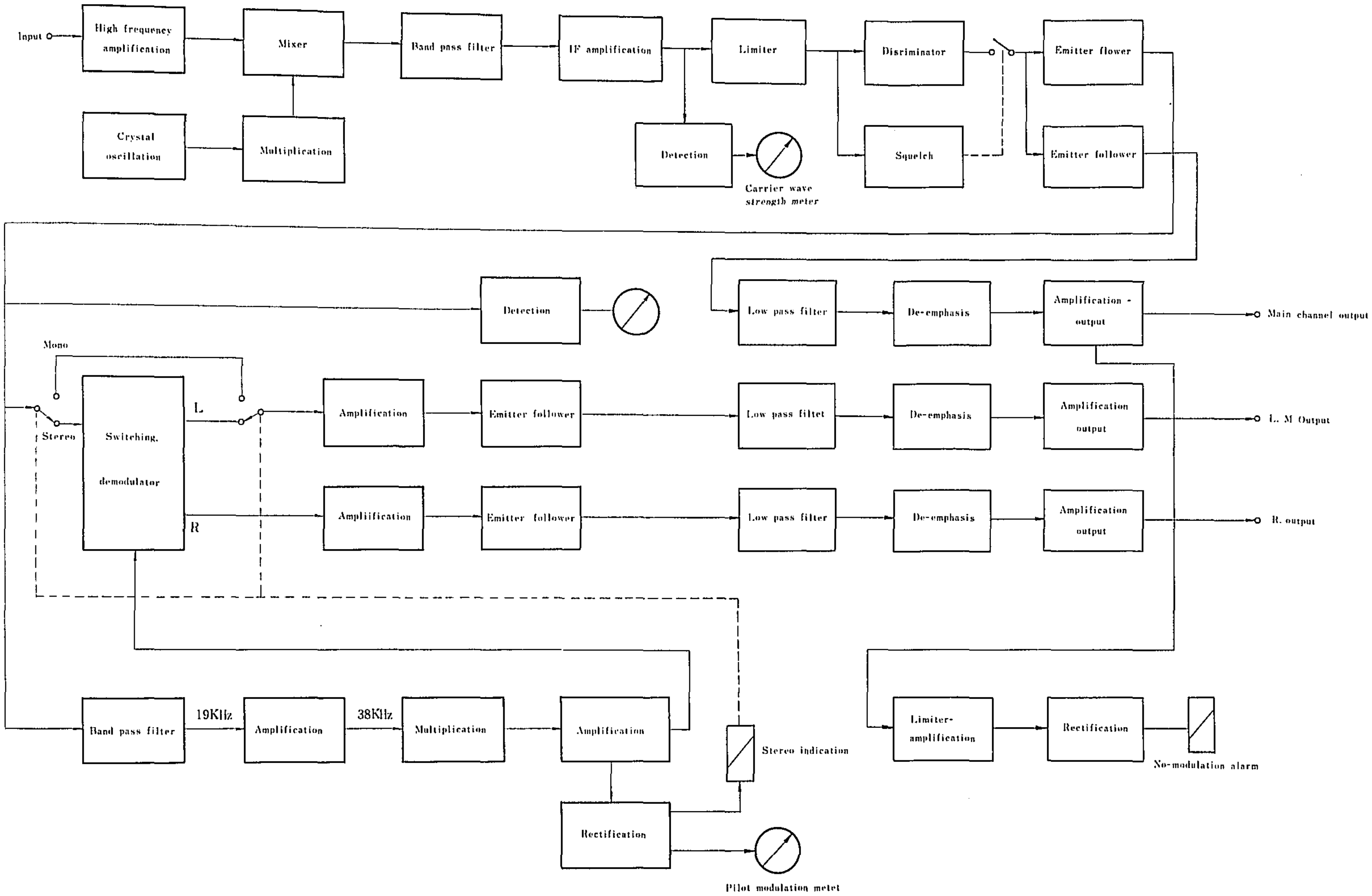


Figure 3.4.14 Block Diagram of a Radio Relay Receiver

(2) Stereophonic FM Supervision Receiver

A monitoring receiver is provided at the studio to supervise the conditions of its transmitter. Figure 3.4.15 shows the block diagram of this receiver.

Figure 3.4.15


Input
High frequency amplification
Mixer
Band pass filter
IF amplification
Limiter
Discriminator
Emitter flower
Crystal oscillation
Multiplication
Detection
Carrier wave strength meter
Squelch
Emitter follower
Detection
Low pass filter
De-emphasis
Amplification - output
Main channel output
Mono
Stereo
Switching. demodulator
L
R
Amplification
Emitter follower
Low pass filtet
De-emphasis
Amplification output
L. M Output
Ampliification
Emitter follower
Low pass filter
De-emphasis
Amplification output
R. output
Band pass filter
19KHz
Amplification
38KHz
Multiplication
Amplification
Stereo indication
Rectification
Pilot modulation metet
Limiter-amplification
Rectification
No-modulation alarm

3.5 FM Rebroadcast Transmitter and FM Relay Unit

3.5.1 Ratings and Performance

Table 3.2 Ratings and Performance of FM Re-Broadcasting Transmitters and FM Relay Units

Classification / Items	Type VV	Type VU and UU	Type UV
Receiving frequency	Designated frequency within 76∿90MHz	Designated frequency within 76∿90MHz or 920∿960MHz	Designated frequency within 920∿960MHz
Transmitting frequency	Designated frequency within 76∿90MHz	Designated frequency within 920∿960MHz	Designated frequency within 76∿90MHz
Maximum frequency deviation (100% modulation)	Input 75kHz, Output 75kHz	Input VHF 75kHz, UHF 150kHz, Output 150kHz	Input 150kHz, Output 75kHz
Standard input level	55∿80dB/ Vμ(open-circuit voltage)	VHF 55∿80dB/μV UHF 60 80dB/μV	60∿80dB/μV
Output	Type 30VV-71 1W, Type 40VV-71 10W, Type 50VV-43 100W	Type 30VU-71 1W, Type 30UU-71 1W, Type 43VU-41 20W	Type 40UV-71 10W, Type 50UV-41 100W
System	Heterodyne frequency conversion system (Shift local)	Heterodyne frequency conversion system	Heterodyne frequency conversion AFC system
Power supply	100/200V single phase AC 50/60Hz		
Ambient conditions, Temperature	Stable operation within the following ranges, -20 to +60°C (However, must be mechanically sound and stable for temperatures between -20 to -10°C and 40 to +60°C)	30VU: -20 to +60°C (Same conditions as in parenthesis at left), 30UU, 43VU: -20 to +40°C (However, must be mechanically sound and stable between the temperatures of -20 to -10°C)	(Same as Type VV)
Humidity, power supply voltage	45 ~ 90%, rated power supply voltage ±5%		

Table 3.3 Performance of FM Relay Re-Broadcasting Transmitters and FM Relay Units

Classification / Items	Type VV	Types VU, UU	Type UV
(General Performance)			
Permissible variation in transmitting frequency	±150Hz (When receiving input frequency variation is zero)	±2kHz	±1KHz
Input impedance	VSWR Under 1.3	VHF VSWR Under 1.3 UHF VSWR Under 1.2	VSWR Under 1.2
Output impedance	VSWR: to match load under 1.3	VSWR: to match load under 1.2	(Same as Type VV)

Classification / Items	Type VV	Types VU, UU	Type UV
Spurious radiation strength	Under -63dB	Under -63dB at center frequency ±500kHz and under -43dB at other frequencies	(Same as Type VV)
Effective Selectivity (1) D/U against its own radiated waves D/U	±600kHz Under -40dB ±700kHz Under -50dB ±800kHz Under -60dB with declining characteristics		
(2) D/U against other station radiated waves	±500kHz Under -50dB with declining characteristics	VHF input over ±500kHz Under -40dB with declining characteristics UHF input over ±2MHz Under -40dB with declining characteristics	Over ±2MHz Under -50dB with declining characteristics
Squelch operation	Adjustable to operate below a desired input-signal level of 30dB/μV (Open-circuit voltage). Under conditions however, where an interference wave (Receiving frequency over ±1,000kHz of own station frequency) of 10dB is added to the input.	Adjustable to operate below the desired input signal level of 30 dB/μV	(Same as Type VU)
Output fluctuation	+0.5 and -1.0dB of rated value at input levels of 50 - 85dB/μV(open-circuit voltage)	Within ± 1dB of rated value at input levels of 50 - 85dB/μV	(Same as Type VV)
AFC operation	-	-	Capable of controlling the overall receiving input and transmitter output frequency variations to under 1/10.

(Monophonic characteristics)

Amplitude frequency characteristics	Within ± 1dB through 50 - 15,000Hz with 1,000Hz as standard		
Distortion factor	Modulation frequency / Degree of modulation	50 - 10,000Hz	10,000 - 15,000Hz
	100%	Under 1%	Under 1.5%
	133% (100% level ±25dB)	Under 2%	Under 3%
S/N ratio	Over 60dB at 1,000Hz 100% modulation		
Residual amplitude modulation noise	Under -53dB when unmodulated Under -40dB at 1,000Hz and 100% modulation		
Occupied frequency bandwidth	Within 200kHz at 15,000Hz 100% modulation	-	(Same as Type VV)

Items \ Classification	Type VV	Types VU, UU	Type UV
(Stereophonic performance)			
Amplitude frequency characteristics	Within ±1dB between 50∿15,000Hz with 1,000Hz as the standard		
Distortion factor			

Distortion factor: Degree of modulation \ Modulation frequency	50 - 10,000Hz	10,000 - 15,000Hz	
Maximum modulation (45-45-10)	Under 1%	Under 1.5%	Degree of pilot signal modulation to be a constant value of 10%
Maximum modulation +2.5dB	2%	Under 3%	

Items	Type VV	Types VU, UU	Type UV
S/N ratio	Over 60dB at 1,000Hz maximum modulation		
L and R separation	Over 33dB between 100 and 10,000Hz at maximum modulation with respective to L and R signals		
Occupied frequency band-width	Within 200kHz at maximum modulation of 15,000Hz	-	(Same as Type VV)

3.5.2 Composition of an FM Re-broadcast Transmitter

(1) Type VV

(a) Block diagram

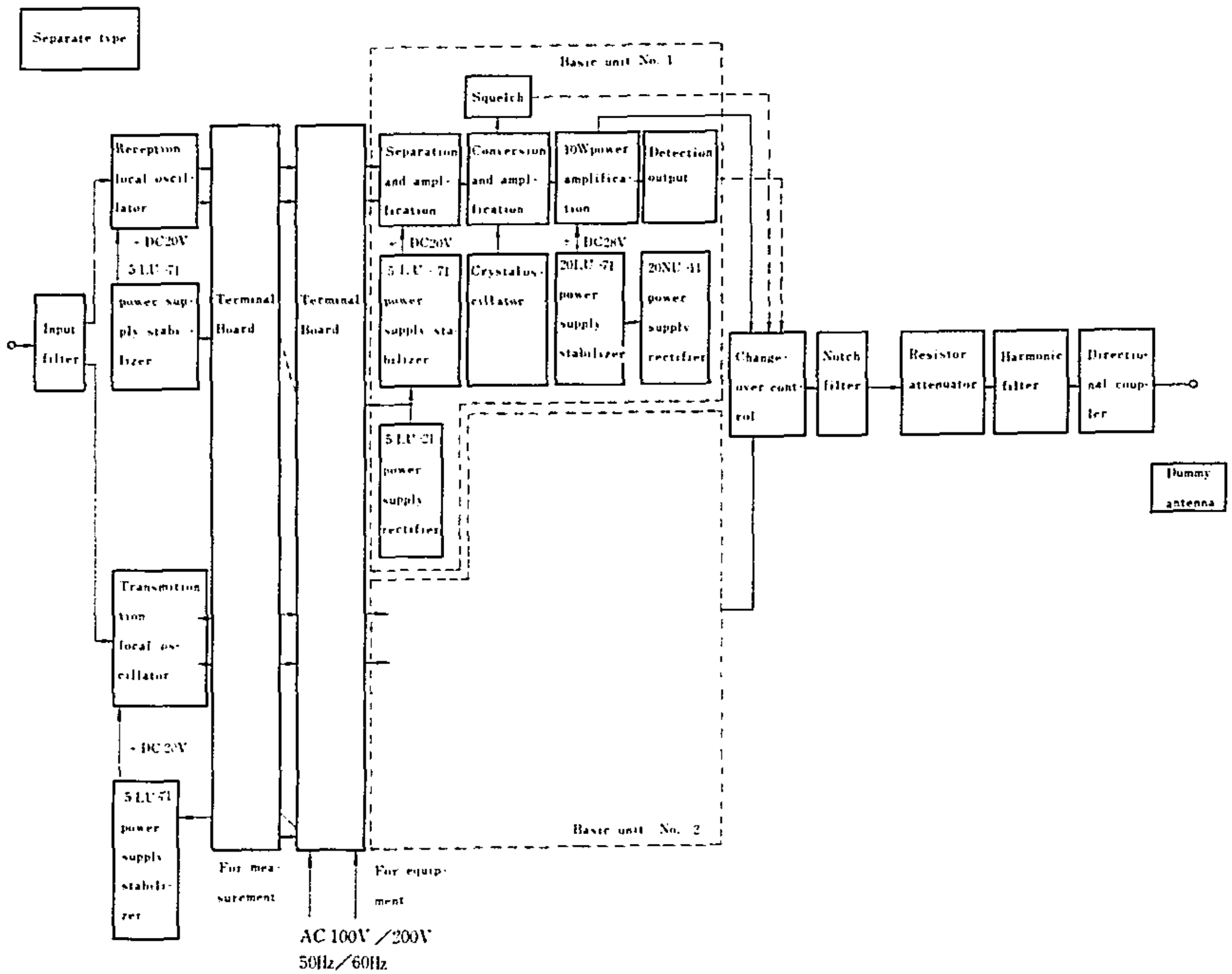
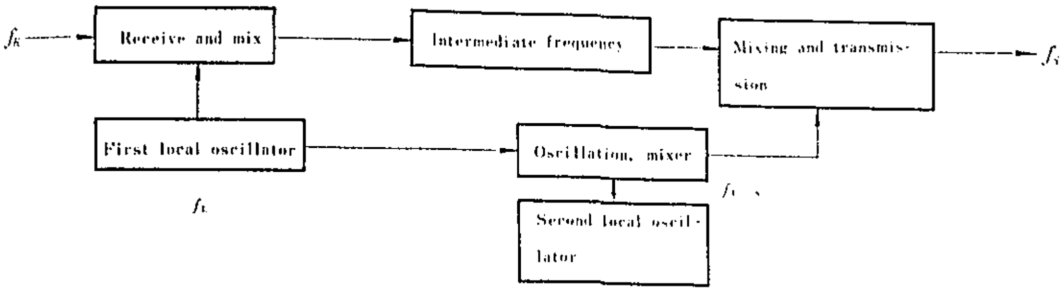


Figure 3.5.1 Type 40VV-71 FM Re-broadcasting Transmitter

(b) Shift-local system

As the tolerance of the transmitting frequency cannot be maintained within the prescribed range by merely adding another crystal oscillators to the local oscillator of receiver and transmitter, the following shift-local system is normally used.



Receiving frequency	f_R
Transmitting frequency	f_S
First local oscillator frequency	f_L
Local oscillator mixer output frequency	$f_{L.S}$

if $\quad f_S = f_R \pm \Delta f$

$$f_{L.S} = f_L \pm \Delta F$$

$$\Delta F = \Delta f$$

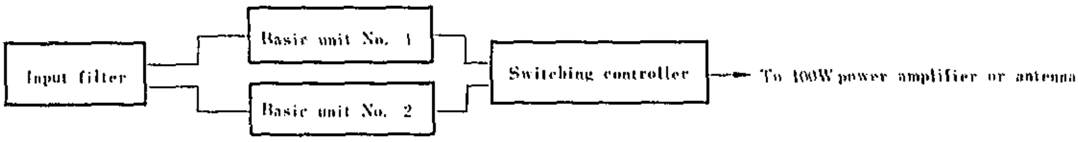
then

$$f_R - f_L + f_{L.S} = f_R - f_L + f_L \pm \Delta F = f_R \pm \Delta F = f_R \pm \Delta f$$

and therefore fluctuations of the first local oscillator will not appear as transmitting frequency fluctuations but only the second local oscillator will appear as frequency fluctuations. Since the second local oscillator frequency is selected as the receiving and transmitting frequency, and the frequency is rather low, it is quite easy to maintain the frequency within the prescribed limit.

(c) Control of basic unit

The units composed of solid-state units of the FM rebroadcasting transmitter are called the basic unit.



The power supplies for Basic Unit No. 1 and No. 2 and always turned on. The basic unit in standby condition is switched to the dummy load by the switching controller and will be switched into the circuit in case trouble develops in the other basic unit.

Control of the basic unit

The basic unit will be switched over to the following 3 conditions by the switching controller.

(1) Basic unit No. 1 in operation (there will be no switch over to No. 2 unit in case of breakdown)

(2) Automatic (changeover will be provided automatically to either unit, in case of breakdown)

(3) Basic unit No. 2 in operation. (There will be no switch-over to No. 1 unit in case of breakdown).

In any of the above three conditions, when there is no input signal entering, the squelch will be operating and the basic units will be all connected to the dummy load and the 100 W power amplification stages will not be excited. This means that there will be no noise broadcast.

When input signal enters, the pre-selected basic unit will be connected to the antenna or the 100 W power amplifier. The other unit will be connected to the dummy load.

In case of automatic operation, if the basic unit in operation breaks down, the squelch or output detector will activate and switchover to other unit will be provided in about 5 seconds.

In case of automatic operation, if the basic unit is switched over to the standby unit, and the trouble of the unit restores by itself for some reason, the restore button should be pushed in order to maintain the normal standby operation condition.

In case there is no input entering the basic unit, the squelch will operate, and in case of the conversion amplification unit and 1W or 10W power amplification unit, the output detector will operate and, switchover will be executed.

(2) Type UV

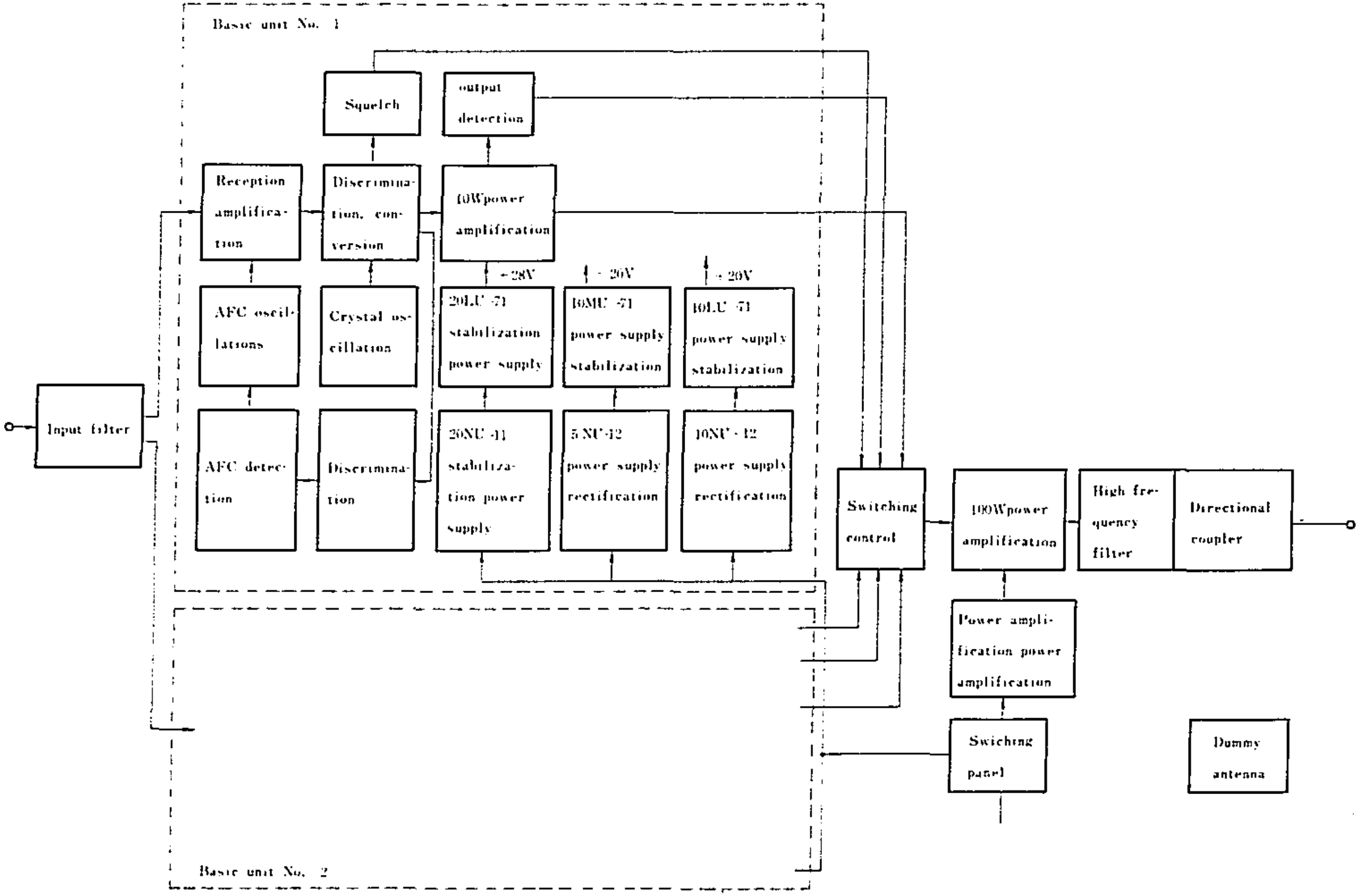


Figure 3.5.2 Type 50UV-41 FM Rebroadcasting Transmitter

3.5.3 Composition of FM Relay Unit

(1) Type VU

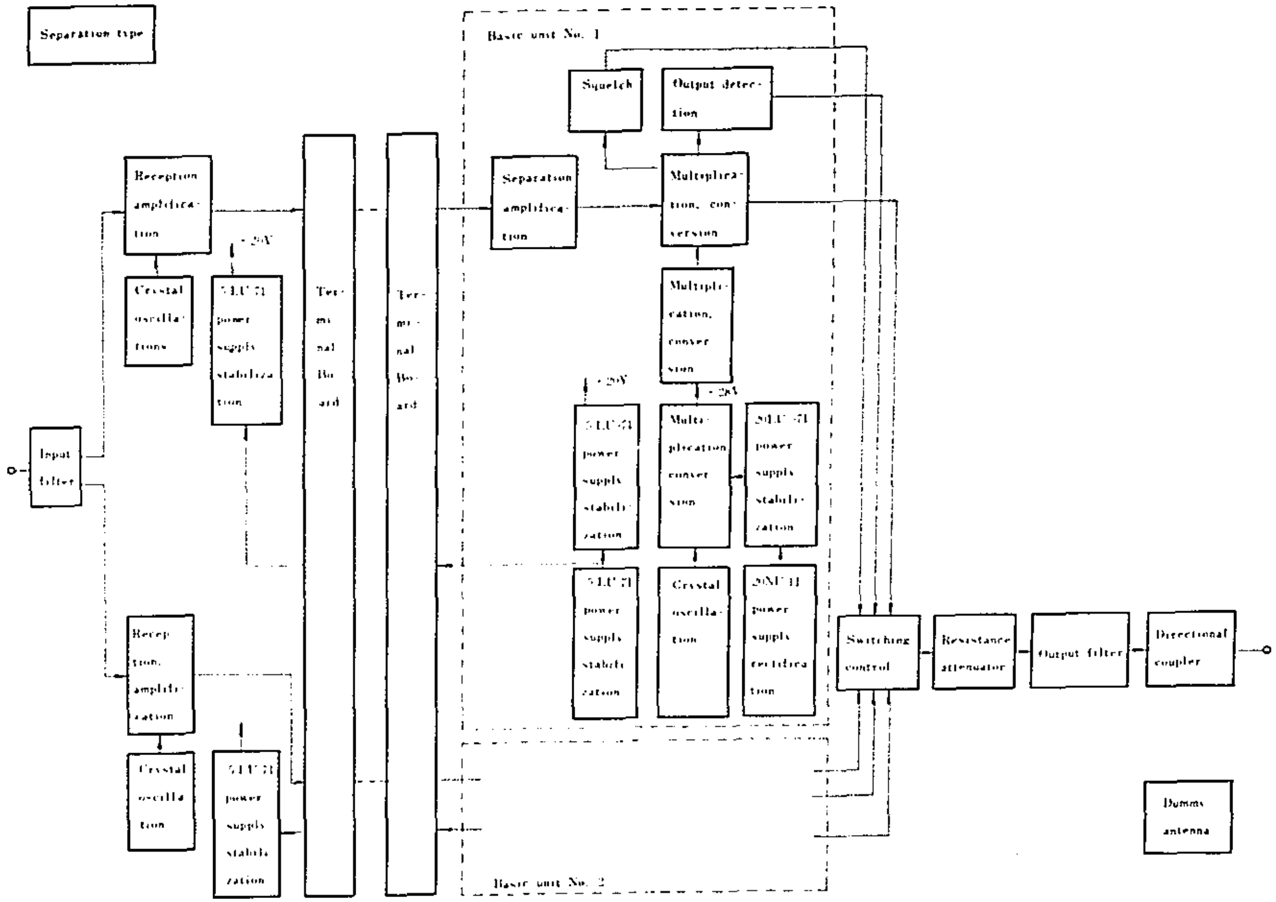


Figure 3.5.3 Type 30VU-71 FM Relay Unit

(2) Type UU

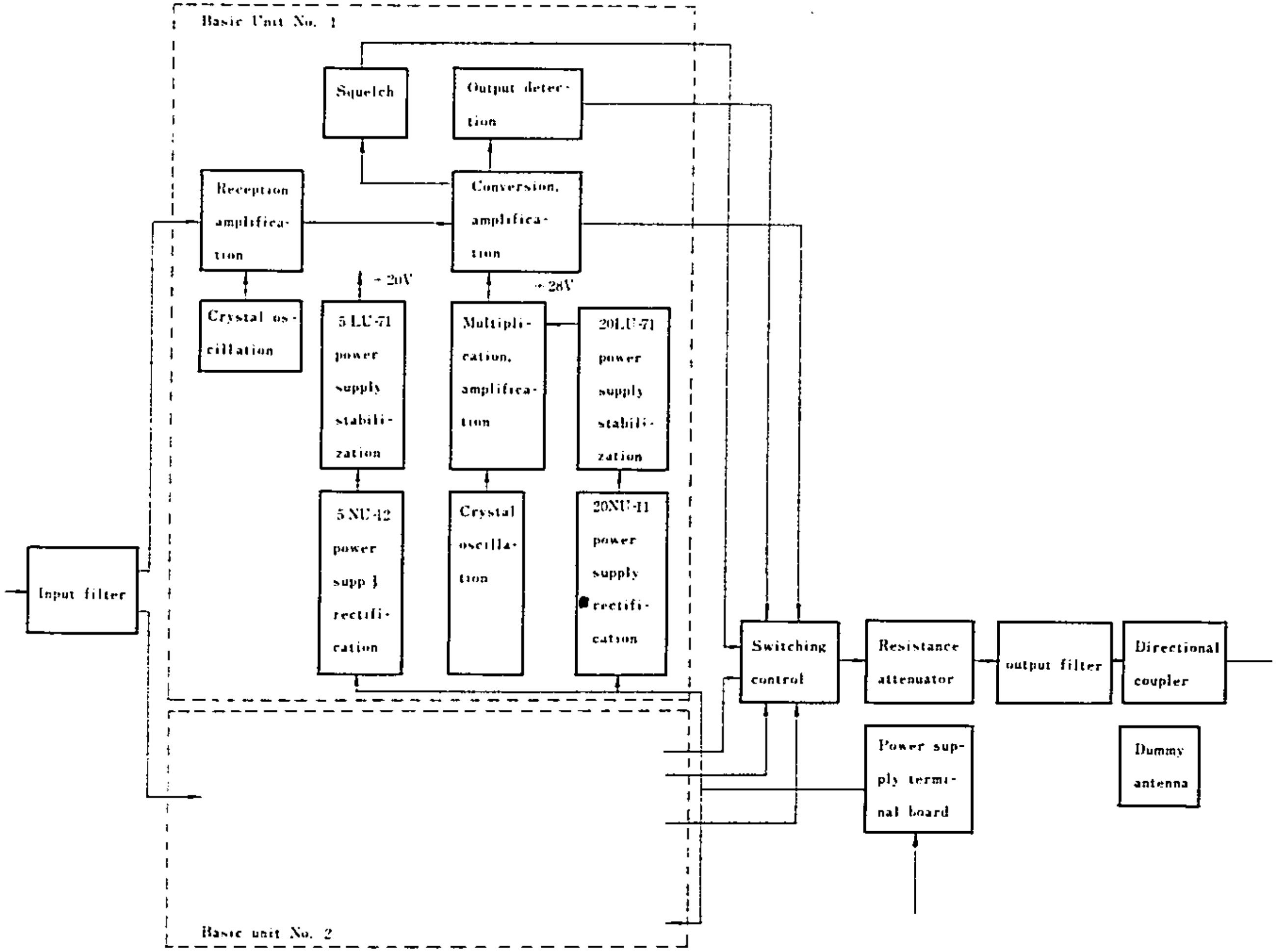


Figure 3.5.4 Type 30UU-71 FM Relay Unit

3.6 Circuit-Transmitter and FM Rebroadcast Transmitter

In the transmission of programmes from studio site to the transmitter site, a method of modulating the FM transmitter by the output of the STL receiver was widely used, but, recently, a method of combining the function of STL transmitter and FM rebroadcasting transmitter was adopted. By this method, the SN ratio, degree of signal separation has been greatly improved.

A block diagram of this combination is shown in Figure 3.6.1.

3.7 Measuring Instruments for FM Stereophonic Broadcasts

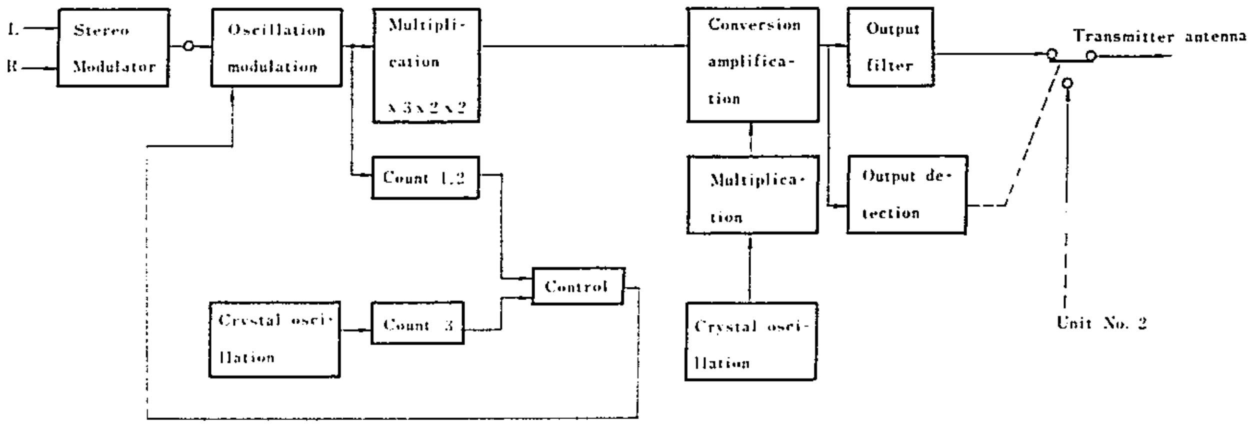
The necessary measuring instruments will be a stereophonic signal generator, FM wide band detector and a stereophonic signal demodulator.

3.7.1 Stereo signal generator (Type HSG-501B)

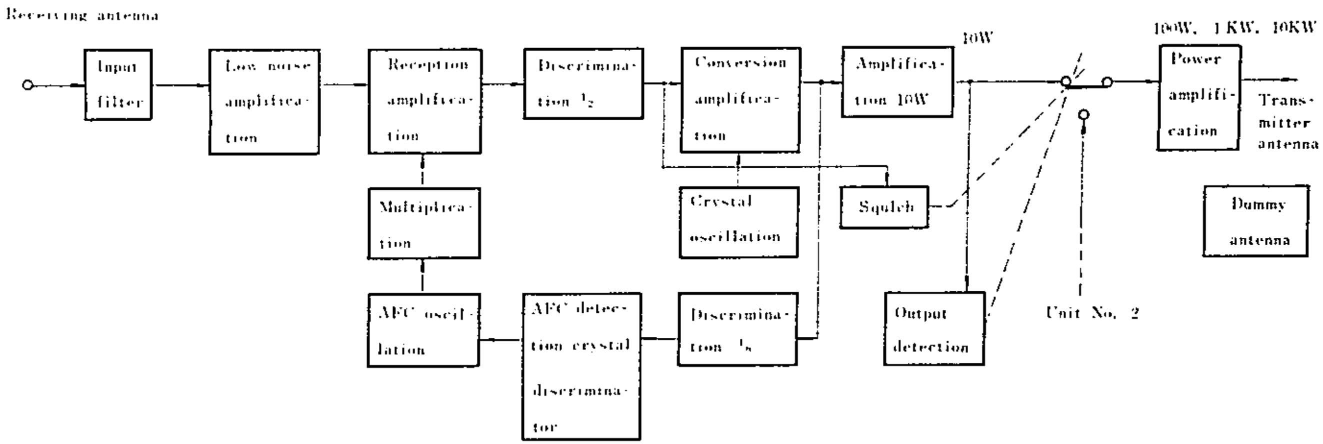
1. Principal performance data

Table 3.4

	Item	Performance
Input	Signals	L and R input signals and oscillator input signal
	Frequency range	50 - 15,000Hz
	Impedance	600Ω balanced
	Level	Approximately +4dBm
	Pre-emphasis	Time constant 50 μs
Main channel output	Signals	Main channel (L and R sum signal)
	Frequency characteristics	Within ±0.5dB between 50 - 15,000Hz (Standard 400Hz)
	Impedance	600Ω unbalanced
	Level	Over +10dBm (From Table 3.5B degree of modulation)
	Distortion factor	Under 0.3% between 50 - 15,000Hz at an output of +10dBm
	Signal to noise ratio	Over 70dB at an output of +10dBm
Composite signal output	Signals	Composite signal (Main channel, subchannel and pilot signals independently or in combinations)
	Frequency characteristics	Within ±0.5dB from 50 - 15,000Hz, within ±0.3dB from 100 - 10,000 Hz



30ST - 71 circuit transmitter



70UV - 41FM Rebroadcasting transmitter

	Item	Performance
Composite signal output (cont'd)	Impedance	75Ω unbalanced
	Level	Over $1V_{p-p}$ (75Ω termination)
	Distortion factor	From table 3.5(B), (C) degree of modulation Under 0.5% from 50 - 100Hz, Under 0.3% from 100 - 10,000Hz, Under 0.4% from 10,000 to 18,000Hz
	Signal to noise ratio	Over 70dB for $1V_{p-p}$ output
	Suppression of sub-carrier wave	Under -40dB for an output of $1V_{p-p}$
	L and R separation	Over 42dB from 50 - 10,000Hz Over 36dB from 10,000 - 15,000Hz
	Pilot signal frequency	Within 19kHz ±0.01%
	Pilot signal output level	Variable over approximately $0.08 - 0.1V_{p-p}$
	Phase difference between pilot signal and the subcarrier wave	Within ±3°
19kHz output	Impedance	High impedance unbalanced
	Level	Over $1V_{p-p}$
	Frequency	Same as pilot signal within 19kHz 19kHz ±0.01%
Power supply voltage		100V, 50/60Hz, single phase
Power consumption		Approximately 10VA

A list of degree of modulation are shown in Table 3.5.

Table 3.5

Name	Modulation signal	Degree of modulation main channel	Degree of modulation, sub-channel	Degree of modulation, pilot signal
A Maximum degree of modulation	Left(or right) stereophonic signals only	45%	45%	10%
B Maximum degree of modulation of main channel	Equal phase and equal amplitude of L and R stereophonic signals	90%	0%	10%
C Maximum degree of modulation of sub-channel	Reverse phase and equal amplitude of L and R stereophonic signals	0%	90%	10%
D Maximum degree of modulation of monophonic signals	Monophonic signals	100%	0%	0%

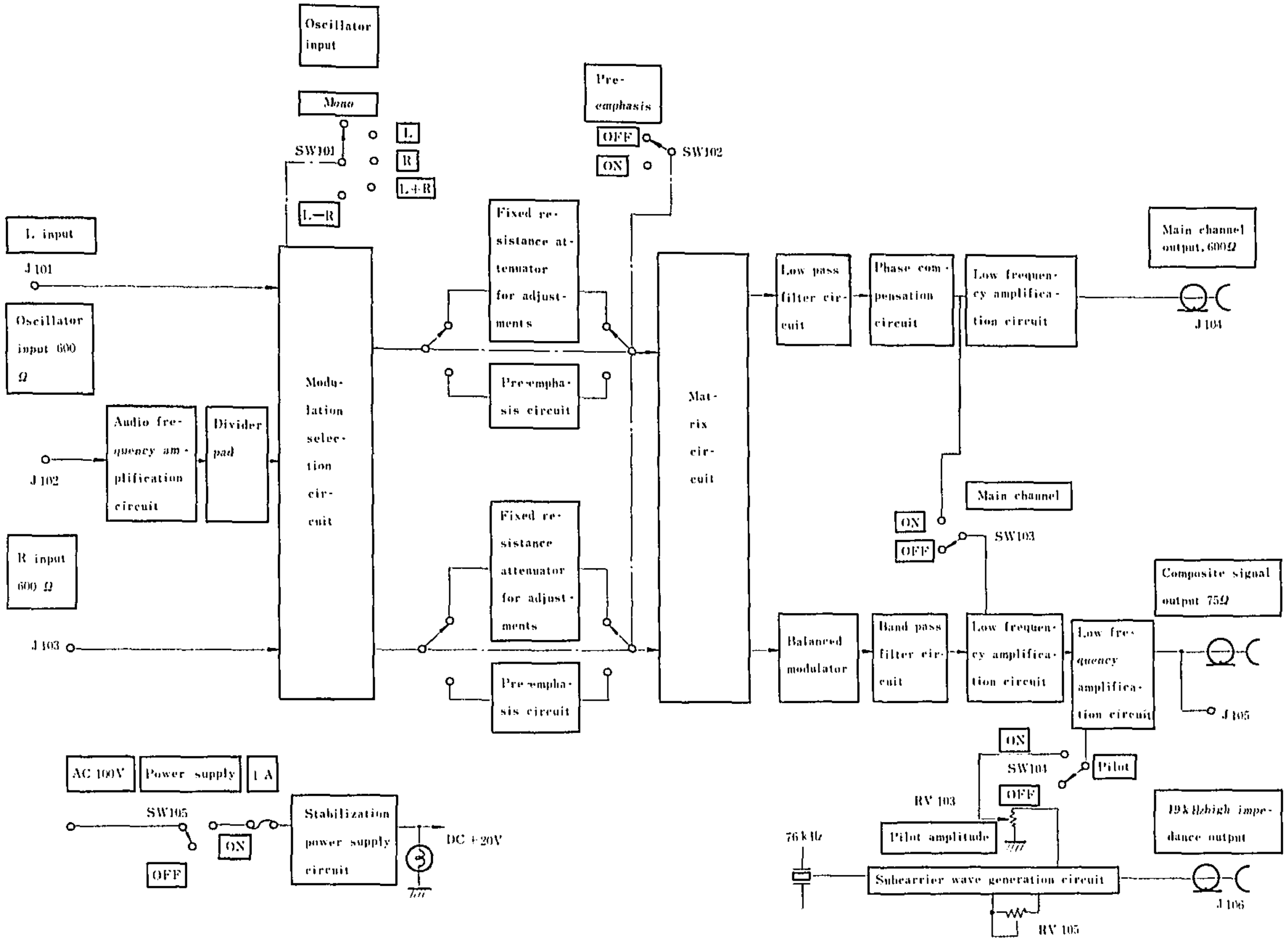


Figure 3.7.1 Block Diagram of the Type HSG-501B Stereophonic Signal Generator

3.7.2 FM Wide Band Linear Detector (Type RDA-501A)

1. Performance

1.1 Principal Performance

Table 3.6

Item			Ratings and electrical performances
Carrier waves	Frequency	Range	6-100MHz (Note: Receiving frequencies that interfere with the intermediate frequencies cannot be used)
		Error	±2%
	Input	Impedance	50Ω (unbalanced), VSWR: Under 1.1 (76 - 90 MHz), under 1.3 (6-100MHz)
		Level	FM input: Over approximately 95dB, AM input: Over approximately 114dB (50Ω termination) 1μV = 0dB
	FM	Indication range	0∿ ±10/50/100/200kHz
		Indication error	±5%(of maximum indication value) from 50Hz to 100kHz
	AM	Indication range	-70dB∿-50/ -40/ -30dB -40dB = 1%
		Indication error	±5%(of maximum indicated value) from 50Hz∿53kHz
Inter-mediate frequency	Center frequency		46MHz
	Band width		Approximately 1.2MHz/-3dB
	Frequency discriminator linearity		Differential characteristics of ±0.2dB within the range of +400kHz of the center frequency
	Detector sensitivity		Over 3mV/kHz

Item			Rating and electrical performances
Low frequency	Output	Frequency range	50Hz ∿ 100kHz
		Impedance	75Ω (unbalanced)
		Level	Over $1V_{p-p}$ (with 75Ω load)
		De-emphasis	50μs and 75μs and OFF
Overall character-istics	FM	Overall characteristics	Within ±0.2dB (from 50Hz ∿ 100kHz)
		Distortion factor charac-teristics	Under 0.25% (with ±100kHz deviation)
		S/N	Over 70dB (with +100kHz deviation)
		L and R separation	Over 40dB (100Hz ∿ 10kHz), Over 36dB (50Hz ∿ 15kHz)
	AM	S/N when measuring resi-dual amplitude modulation noise	Over 50dB (at -30dB modulation)
Frequency deviation calibration			With built in oscillator, ±75kHz, deviation within ±2% of indication error.
Power supply			100V AC ±10%, 50/60Hz, approximately 10VA
Weight and measurement			9kg (without rubber feet and cord) W430 × H150 × D280mm

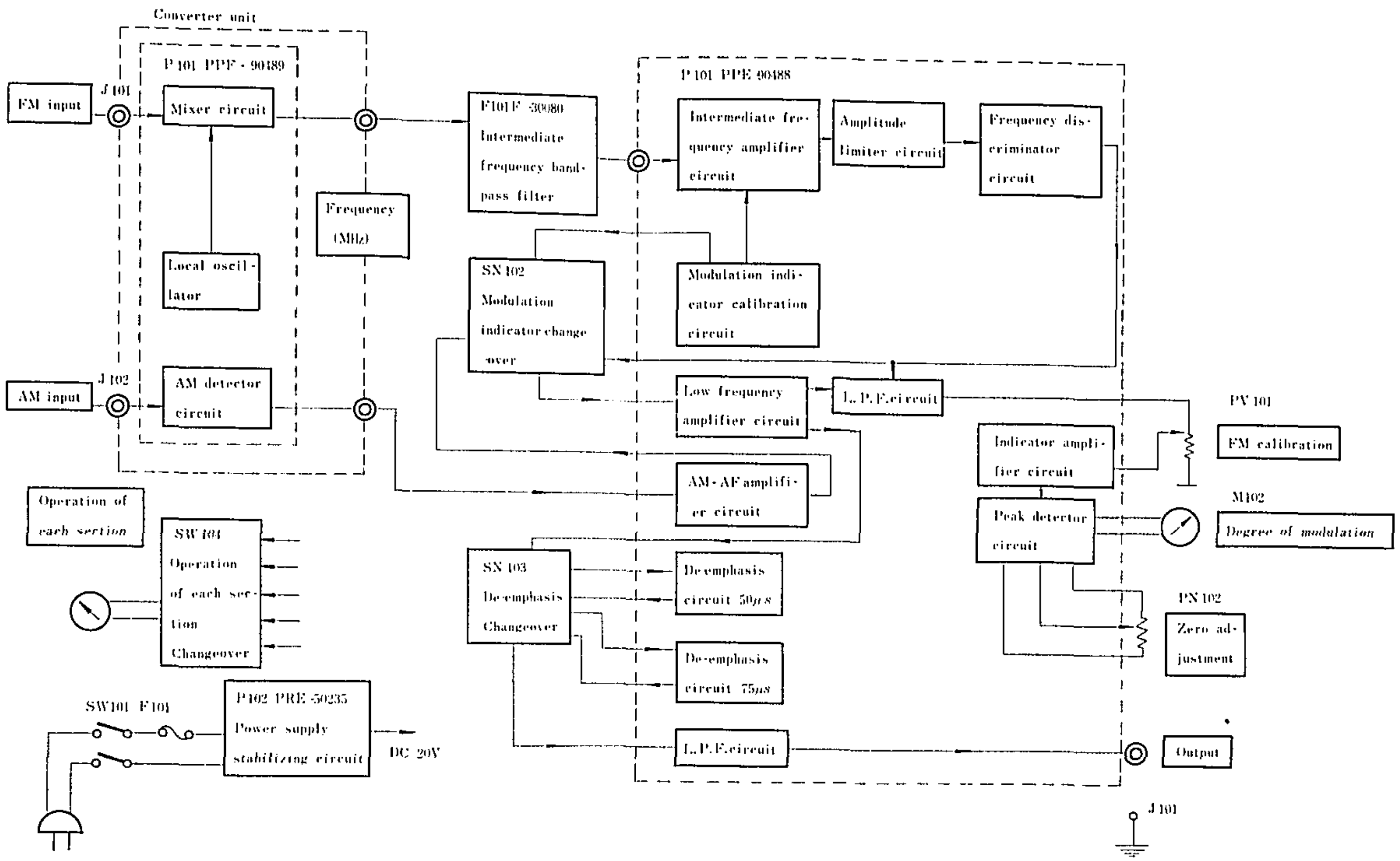


Figure 3.7.2 Block Diagram of Type RDA501A FM Wide Band Linear Detector

3.7.3 Stereophonic Signal Demodulator (Type HSD-501B)

1. Principal specifications

Table 3.7

Item		Performance
Input	Signal	Monophonic and composite signals
	Frequency range	Monophonic signal 50Hz ∿ 15kHz
		Composite signal 50Hz ∿ 53kHz
		Pilot signal 19kHz
	Impedance	75Ω unbalanced
	Level	$1V_{p-p}$
Output	Signals	Monophonic and composite signal detector output
	Frequency range	Monophonic signal 50Hz ∿ 15kHz
		Composite signal 50Hz ∿ 15kHz
	Impedance	600Ω balanced
	Level	+10dBm from Table 3.5 at various degrees of modulation
	Frequency characteristics	Within ±0.5dB from 50Hz to 15kHz
		Within ±0.3dB from 100Hz to 10kHz
	Distortion factor	To correspond with Table 3.5 at various degrees of modulation up to an output of +10dBm
		Under 0.4% from 50Hz to 100Hz
		Under 0.3% from 100Hz to 10kHz
		Under 0.5% from 10kHz to 15kHz
	Signal-to-noise ratio	Over 65dB at +10dBm output
	L and R separation	Over 36dB from 50Hz to 15kHz Over 42dB from 100Hz to 10kHz
Power supply voltage		100V AC, 50/60Hz, single phase
Power consumption		Approximately 10VA

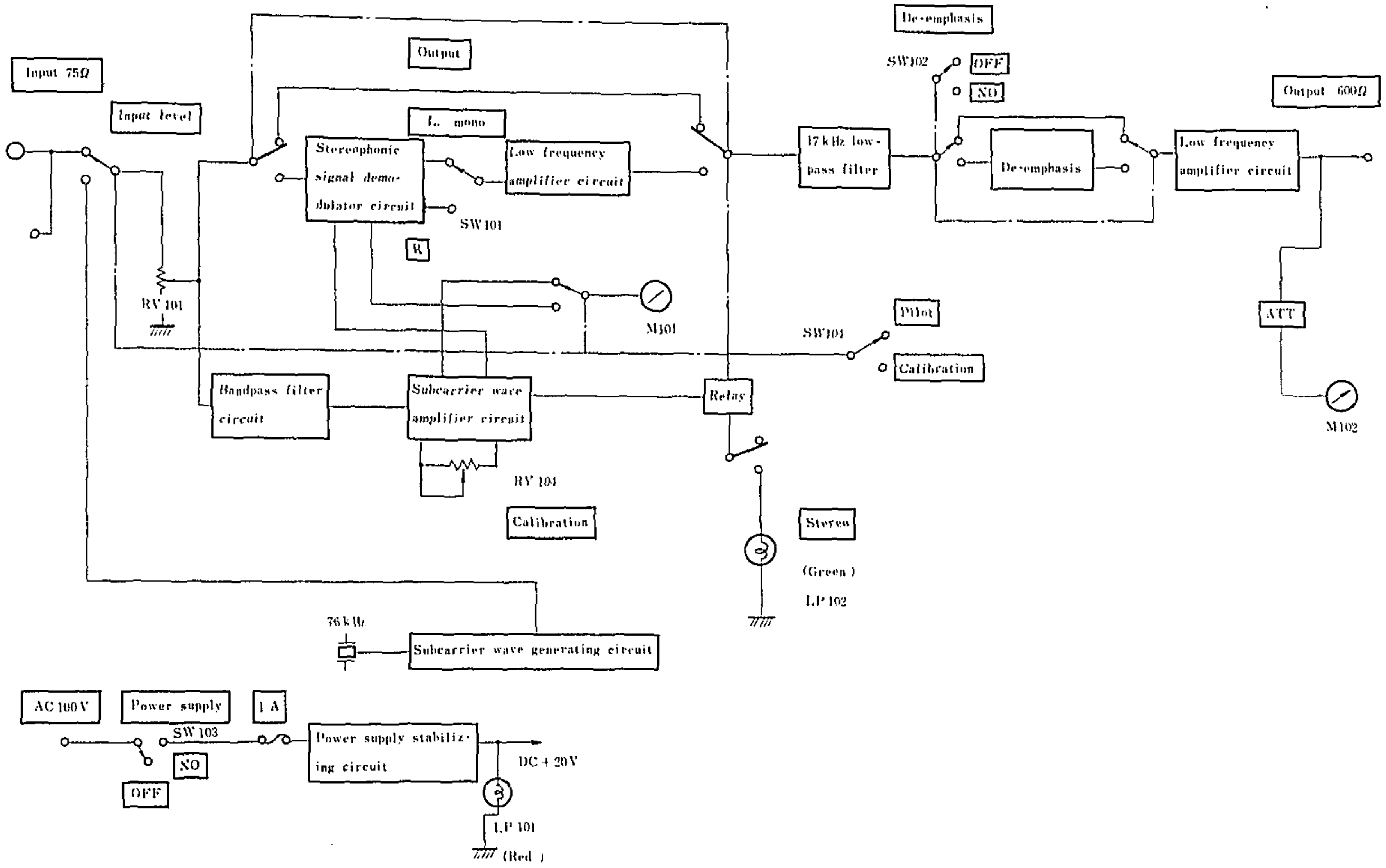


Figure 3.7.3 Block diagram of Type HSD-501B Stereophonic Signal Demodulator

3.7.4 Measurement of Left and Right Signal Separation

(1) Method using oscilloscope wave form

Create a composite signal wave form output of a linear detector on an oscilloscope. Then the degree of separation can be obtained by the following equations.

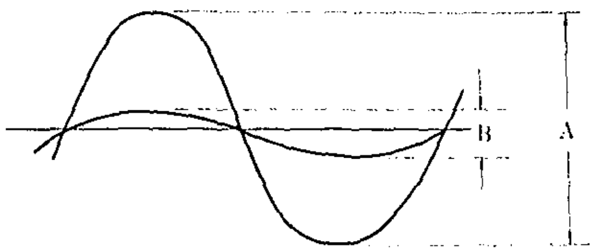


Figure 3.7.4

$$\text{Separation} = 20\log\frac{A}{B}\ (\text{dB})$$

(i) When only level difference exists

$$\frac{L}{R} = \frac{M+S}{M-S} = \frac{A}{B}$$

$$\therefore \text{Separation} = 20\log\frac{A}{B}\ (\text{dB})$$

(ii) When only phase difference exists

$$M + S = P\sin\ (P_t + d) + P\sin P+$$

$$= 2P\sin\ (P_t + \frac{d}{2})\ \cos\frac{d}{2}$$

$$M - S = P\sin\ (P_t + d) - P\sin P+$$

$$= 2P\cos(P_t + \frac{d}{2})\cdot\sin\frac{d}{2}$$

$$\text{Separation} = 20\log\frac{A}{B}$$

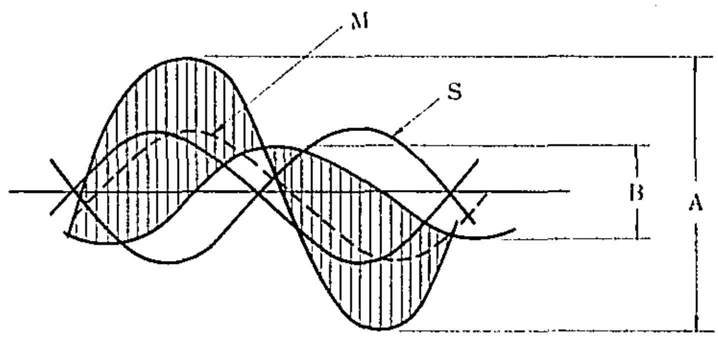


Figure 3.7.5

(2) Method Using Instruments

Insert a left (or right) signal only and determine the degree of separation from the difference in the left (or right) and right (or left) output levels of the stereo demodulator. (output levels expressed in dB).

3.7.5 Measurements of Phase of Pilot Signal and AM Subcarrier Wave Phase

The stereophonic modulation signal can be expressed by

$$(L+R) + (L-R)\cos(\omega_s t + \phi) + P\cos\frac{\omega_s t}{2}$$

$L+R$ = Main Channel Signal

$L-R$ = Subchannel Signal

ω_s = Subcarrier Wave Angular Frequency

$\omega_s/2$ = Pilot Angular Frequency

ϕ = Phase difference between the subcarrier wave reproduced by pilot signal and the subcarrier wave transmitting through the subchannel

P = Amplitude of the pilot signal

To measure θ, create a Lissajous' figure of the subcarrier wave and pilot signal on the socilloscope.

First, project only the pilot signal, not the modulation signal onto the vertical axis of the oscilloscope, and then applying the pilot oscillator output to the horizontal axis, put the phase of the 19kHz component of the vertical and horizontal axis in phase with $\phi 1$. This is to eliminate any phase differences of the 19kHz frequency itself in the oscilloscope and other circuits.

Next, add the modulation signal, and add the AM subchannel or main and subchannel composite signal to the vertical axis, to form a Lissajous' figure of 2:1.

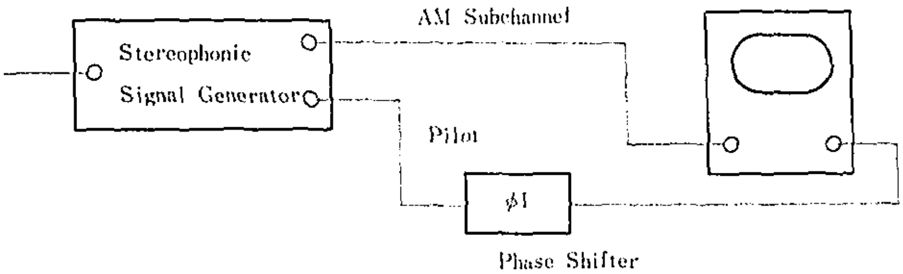


Figure 3.7.6

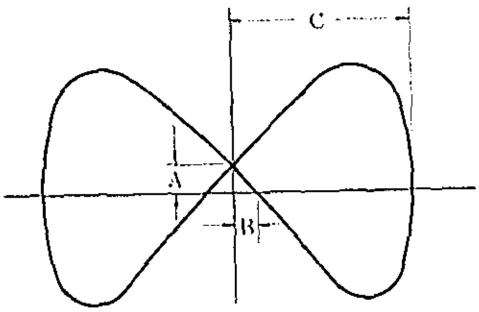


Figure 3.7.7

If $y = \sin(\omega_s t + \phi)$ denotes the subchannel, and $x = \sin \frac{\omega_s}{2} t$ denotes the pilot signal and the vertical axis is also set as 1, then $A = \sin \phi$. Next, x will be obtained from the time $y = 0$ up to this time will be $x = \sin \frac{\phi}{2}$ and the point at which the Lissajous' figure intersects the horizontal axis will be, $B = \sin \frac{\phi}{2}$ if the horizontal axis is also set 1.

It will therefore be possible to determine ϕ from the equation $\frac{\phi}{2} = \sin^{-1} \frac{B}{C}$.

As the polarity of the subcarrier wave reverses between a - b and b - c as shown in Figure 3.7.8 when a subchannel signal is applied to the vertical axis, the upper and lower part of the Lissajous' figure in Figure 3.7.7 will be interchanged and will become as shown in Figure 3.7.9.

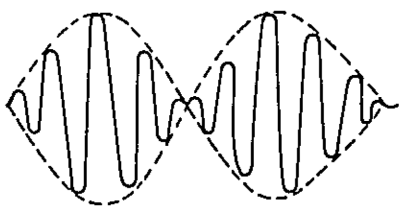


Figure 3.7.8

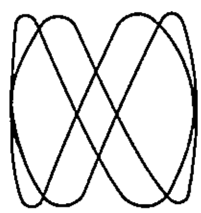


Figure 3.7.9

Supplement 1.

CCIR PROBLEMS IN RELATION TO STEREOPHONIC BROADCASTS

QUESTION No. 199 (X)

STEREOPHONIC BROADCASTING

(Los Angeles. 1959)

The C.C.I.R.

CONSIDERING

(a) that stereophonic recording of sound on both disc and magnetic tape is already becoming well established in the industry and such discs and tapes are already on sale to the public in some countries;

(b) that experimental transmissions of stereophonic sound programmes have already been made by broadcasting stations in a number of countries;

(c) that, if such transmissions become general without international coordination, serious problems of interference to existing broadcasting services could arise;

(d) that by the adoption of suitable techniques on an international scale such interference problems could be avoided and spectrum occupancy reduced;

(e) that it is desirable to achieve international standardization of transmission parameters so as to make possible the standardization of some parts of receivers for stereophonic broadcasting;

:UNANIMOUSLY DECIDES that the following question should be studied:

1. by what methods can satisfactory stereophonic sound be broadcast to ensure maximum economy in frequency usage;
2. what systems can ensure compatibility together with no sigunificant loss of coverage or increase in mutual interference with existing services;
3. what parameters should be standardized?

STUDY PROGRAMME No. 163 (X)

STEREOPHONIC BROADCASTING STANDARDS FOR COMPATIBLE SYSTEMS

IN SOUND AND TELEVISION BROADCASTING

(Los Angeles, 1959)

The C.C.I.R.,

UNANIMOUSLY DECIDES that the following studies should be carried out;

1. investigate the systems for compatible stereophonic broadcasting indicating;
   1.1 the general principles of each system;
   1.2 the detailed specification of each system;
   1.3 the overall theoretical evaluation of the performance of each system;
2. study the systems with particular regard to their feasibility and applicability to existing broadcast transmitters;
3. study the systems with regard to;
   3.1 performance of existing non-stereophonic receivers when tuned to the stereophonic transmission;
   3.2 performance of stereophonic receivers when tuned to the stereophonic signal;
   3.3 performance of stereophonic receivers when tuned to non-stereophonic signals;
   3.4 possibility of adapting existing non-stereophonic receivers for stereophonic reception;
4. investigate the systems with particular regard to;
   4.1 coverage;
   4.2 interference effects;
   4.3 bandwidth involved and other matters concerned with channel utilization;
5. carry out field tests of those systems that appear most satisfactory;

6. study and report on the required technical characteristics of studio-transmitter links and related stereophonic transmission facilities;
7. study the subjective aspects of stereophonic sound;

Supplement 2. PROGRESS IN DETERMINING THE STANDARD STEREOPHONIC SYSTEM IN AMERICA

1. Progress in Survey

The 14 systems suggested were studied by the National Stereophonic Radio Committee (NSRC) and reduced to a total of 7 systems by consolidating similar systems. Studies were further advanced by classifying these into 5 similar types as shown in Table 1, and a report on the survey was submitted to the FCC in May of 1960.

The FCC instructed studies be made in relation to these 7 systems plus a system directly advanced by the Philco Corp. for a total of 8 systems.

Subsequently the 5th system (GEC) and the Philco system were withdrawn by the proponents and, field tests were conducted by the NSRC on the remaining 6 systems plus the 4 - 4A system, which is a combination of the No. 4 and 4-A systems.

A 50kW transmitter of FM station KDKA in Pittsburgh, Pennsylvania was used in these tests. Reception tests were conducted at the 3 field strengths of 1mV/m, 300μV/m and 150μV/m for frequency response, stereophonic separation, distortion, S/N ratio and cross-talk together with broadcast reception evaluation tests from test tapes of standard stereophonic programs.

The results of these tests were submitted to the FCC in October of 1960 and were used as basic data for determining the system.

The FCC published Memorandum 13506 on April 20, 1961 in relation to revisions of the current regulations necessary for FM Broadcasting Stations, to broadcast stereophonic programs. Past developments and reasons for deciding the system were reported and also that these revisions would become effective on June 1 of the same year.

2. Outline of the Various Systems

The technical standards of the various systems surveyed and the transmitter and receiver systems used are shown in Table 1 and Figure 1 to 6.

3. Comparison Studies of the Various Systems

(a) First System (Crosby)

Since this is an FM-FM system and is moreover of a wide-band characteristics (the required stereophonic bandwidth alone is designated as 112.5kHz), this is not compatible with SCA operation. Also, since the drop in S/N ratio during monophonic reception is 5dB higher than in the 4-4A system, this system was rejected for the reason that it will have undesirable effects on the monophonic listeners.

(b) System No. 2A (Calbest)
System No. 2B (Halstead)

As the upper limits of the stereophonic subcarrier wave are 7,000Hz and 8,000Hz, stereophonic separation will be lacking over these cut-off frequencies. These systems were rejected for the reason that they would be deficient in good stereophonic sound quality.

(c) System No. 3 (Percival)

This system is theoretically superior to other systems, however, in practical usage, it lacks the ability to fully display its stereophonic effects. For example, with regard to multiple simultaneous stationary sound, the sound image concentrates in a spot, and furthermore, it has a drawback of rapid variation in gains between the L and R output signals. For the foregoing reasons, this system was also rejected.

System	Particulars / Proponent	Modulation system	Channel bandwidth: Main channel	Channel bandwidth: Subchannel for stereo	Channel bandwidth: Subchannel for SCA	Subcarrier wave frequency: For pilot	Subcarrier wave frequency: For stereo	Subcarrier wave frequency: For SCA	Content of modulated signals: Main channel	Content of modulated signals: Subchannel for stereo	Degree of modulation: Main channel main carrier wave	Degree of modulation: Stereo subchannel Stereo subcarrier	Degree of modulation: Stereo subchannel Main carrier wave	Degree of modulation: SCA subchannel SCA subcarrier wave	Degree of modulation: SCA subcarrier wave Main carrier wave	Contents of output signals: Mono phonic	Contents of output signals: Stereo phonic
1	Crosby Telect Corp	FM -FM	15kHz	15kHz	-	-	50kHz	-	L+R	L-R	±37.5kHz	±25 (FM)	±37.5kHz	-	-	L+R	2L 2R
2A	Calbest Electronics	FM -FM	15kHz	7kHz	-	-	29.5kHz	(67kHz)	L+R	-R	±60kHz	±9.5 (FM)	±15kHz	-	-	L+R	L R
2B	Multiplex Development Corp (Halstead)	FM -FM	15kHz	8kHz	-	-	41kHz	(67kHz)	2R-L	2R-L	±52.5kHz	±9.5 (FM)	±22.5kHz	-	-	2L-R or 2R-L	$\frac{3}{2}L$ $\frac{3}{2}R$
3	EMI Ltd (percival)	FM -FM	15kHz	100Hz	-	-	22kHz	-	L+R	Directional Information L' L'+R'	±67.5kHz	±500 (FM)	±7.5kHz	-	-	L+R	L" R"
4	Zenith Radis Corp	AM -FM	15kHz	15kHz	-	19.5kHz	39kHz	(67kHz)	L+R	L-R	90% (± 67.5kHz)	(AM)	±67.5kHz	-	-	L+R	2L 2R
4A	G E C	AM -FM	15kHz	15kHz	-	19kHz	38kHz	(67kHz)	L+R	L-R	90% (± 67.5kHz)	(AM)	±67.5kHz	-	-	L+R	2L 2R
5	G E C	AM -FM	15kHz	15kHz	-	-	236kHz	(67kHz)	L+R	L-R	62.5% (±58.875kHz)	(AM)	±16.125kHz	-	-	L+R	2L 2R
4-4A	(Composite of 4 and 4A)	AM A -FM	15kHz (50~15000HZ)	15kHz (23~53kHz)	55kHz (20~75kHz) 22kHz (53~75kHz) Common with stereo	19kHz	38kHz (Suppres-sed)	(67kHz)	L+R	L-R	Under 45% (L or R only) (±33.75kHz)	Under 45% (L or R only) (±33.75kHz)		Under 30% Under 10% (common with stereo)		L+R	2L 2R

(d) System No. 4 (Zenis) and No. 4A (GEC)

These are both AM-FM systems and are recognized as being the same theoretically. Comparisons and studies were made in relation to the 4-4A system which is a combination of these two systems.

The results of these studies were that

(1) frequency response and stereophonic separation were equivalent to the Crosby system (No. 1) theoretically and in practice.

(2) Stereophonic broadcasts are practically possible with no effect on monophonic audience.

(Calculation example of reduction in S/N ratio)	System No. 1	System No.4-4A
Monophonic receiver output	6dB	Under 1 dB
Subcarrier wave output	15	23
Left (Right) signal output	13	20

(3) Can be used jointly with SCA multiplication.

(4) The cost of the receiver (broadcasting transmitter) will not be expensive compared with other systems.

With the foregoing reasons, and from the viewpoint of the "principle of carrying out the task of offering high quality artistic-like quality", System No. 4-4A was selected as the standard system. The system was considered most gainful to the public and compatible with economic and related elements without severely obstructing present operations being conducted under current regulations.

Supplement 3. SCA (SUBSIDIARY COMMUNICATIONS AUTHORIZATION)

1. SCA Service is a special supplementary communication service of a non-broadcasting nature under special approval of the FCC (Federal Communications Commission) in relation to applications submitted by broadcasting stations.

SCA service is designated as being a nature, that must be carried out separately from the FM broadcasting hours and, considered to be merely a supplementary service, and restricted from independent operations.

FCC's approval will be required for SCA transmission, but no approval is required for reception facilities. Program contents are background music, news commentaries, time-signals and detailed weather forecasts with no commercials, and are broadcast to special contract listeners only. The listeners' receivers must be those specially designated by the SCA, authority.

The contractors are mainly hotels, restaurants, stores, department stores, hospitals, business offices and factories and the monthly rate is approximately $25.00 per contract.

2. Technical Standards of SCA

(1) SCA subcarrier waves must be FM
(2) The instantaneous frequency of SCA's subcarrier wave must be within the range of 20 - 75kHz and, for stereophonic broadcasting within the range of 53 - 75kHz.
(3) The arithmetic total of SCA's subcarrier wave group modulation of the main carrier wave must not exceed 30% and, for stereophonic broadcasts, must not exceed 10%.
(4) The frequency modulation of the main carrier wave by SCA's subcarrier wave must be 60dB below 100% modulation, within the range of 50 - 15,000Hz.

(Reference)

a. Monophonic Broadcasts + SCA

b. Stereophonic Broadcasts + SCA

Supplement 4. THE EFFECTS OF NUMBER OF MULTI-RELAYS ON CHARACTERISTICS

(i) White noise will be the sum of the power of number of relays

(ii) Increase of distortion and decrease in degree of separation occuring by the re-broadcasting transmitter are indicated by arithmetic totals for each frequency component

(iii) Distortion arising from multi-propagation increases in the form of average squares.

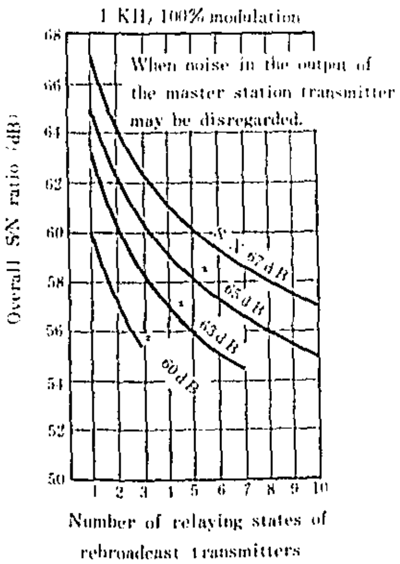


Figure 1. S/N ratio of multi-relaying stage of multi-relaying stages

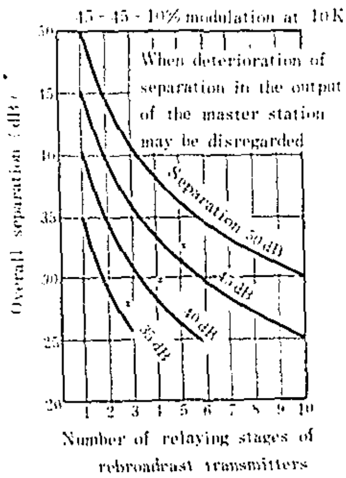


Figure 2. Degree of separation of multi-stage relays

Supplement 5. DISTORTION IN RECEPTION OF MULTI-PATH PROPAGATION FM WAVES

1. Introductory Note

In reception of FM waves, if there is reflection waves from mountains or large structures, besides the direct waves for reception, distortion may occur.

The actual phenomena are extremely complicated, but to grasp the outline of these phenomena, we will explain the clarification results as follows of the most simple case when two FM waves arrive with certain time difference.

2. Indication of Composite Waves

In case that two FM waves propagated along two different paths, and the second FM wave delayed a time of t_0 than the first wave, the instantaneous voltage of the two waves will be expressed by the following equation.

$$e_1 = E_1 \sin(\omega_t + \frac{D}{\mu} \sin 2\pi\mu t) \qquad (1)$$

$$e_2 = E_2 \sin\{\omega(t - t_0) + \frac{D}{\mu} \sin 2\pi\mu(t - t_0) \qquad (2)$$

In the above equations,

- ω = Angular frequency of the carrier wave when unmodulated
- D = Maximum frequency deviation
- μ = Audio signal frequency
- t_0 = Delay time of wave No. 2 in relation to wave No. 1
- E_1 = Amplitude of wave No. 1
- E_2 = Amplitude of wave No. 2

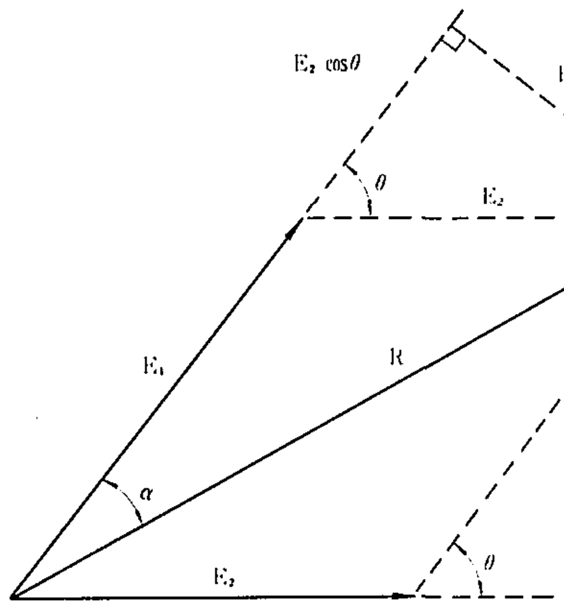


Figure 1. Vector Composition

Figure 1. Vector Composition

Now if we compose these two waves according to Figure 1, the amplitude of composite wave R will be as follows.

$$R = \sqrt{E_1^2 + E_2^2 + 2E_1 E_2 \cos\theta} \tag{3}$$

Here

$$\theta = \frac{D}{\mu}\sin 2\pi\mu t - \frac{D}{\mu}\sin 2\pi\mu(t - t_0) + \omega t_0$$

$$= 2\frac{D}{\mu}\sin \pi\mu t_0 \cdot \cos(2\pi\mu t_0 - \pi\mu t_0) + \omega t_0$$

$$= Z\cdot\cos(2\pi\mu t - \pi\mu t_0) + \omega t_0 \tag{4}$$

However,

$$Z = 2\frac{D}{\mu}\sin \pi\mu t_0 \tag{5}$$

Angle α formed by the composite wave R and E will be

$$\tan\alpha = \frac{E_2 \sin\theta}{E_1 + E_2 \cos\theta} = \frac{x \sin\theta}{1 + x\cos\theta} \tag{6}$$

Here,

$$x = E_2/E_1 \tag{7}$$

Therefore, the composite signal will be

$$e_1 + e_2 = \sqrt{E_1^2 + E_2^2 + 2E\ E\ \cos\theta} \times$$

$$\sin(\omega t + \frac{D}{\mu}\sin 2\pi\mu t - \tan^{-1}\frac{x\sin\theta}{1+x\cos\theta}) \tag{8}$$

If we substitute equations (4) and (7) in the above equation, we will have

$$e_1 + e_2 = E_1\sqrt{1 + x^2 + 2x\cos\{2\cos(2\pi\mu t - \pi\mu t_0) + \omega\ _0\}} \times$$

$$\sin\left[\omega t + \frac{D}{\mu}\sin 2\pi\mu t - \tan^{-1}\frac{x\sin\{Z\cos(2\pi\mu t - \pi\mu t_0) + \omega\ _0\}}{1 + x\cos\{Z\cos(2\pi\mu t - \pi\mu t_0) + \omega t_0\}}\right]$$

Here,

$$Z = 2\frac{D}{\mu}\sin\pi\mu t_0 \tag{9}$$

In equation (1), variations in composite wave amplitude, declinations in the sine functions indicates phase variations of the signal. The FM wave is therefore subjected to amplitude modulation together with distortion of its FM components.

3. Effects of Interference on Envelopes of FM Waves

The amplitude envelope will become the function of the signal voltage ratio x, maximum frequency deviation D, delay time t_0 and audio frequency μ of the 2 FM waves. Examples of this are shown in Figures 2 - 6.

1.

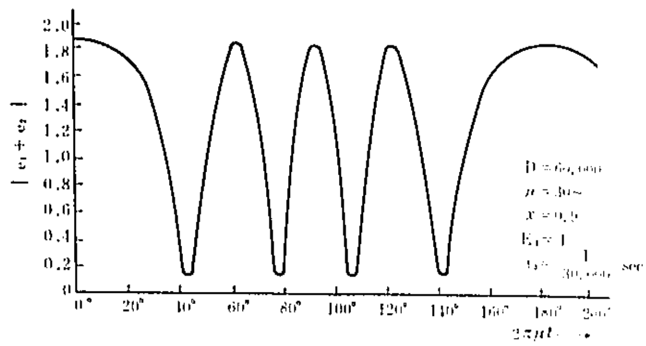


Figure 2

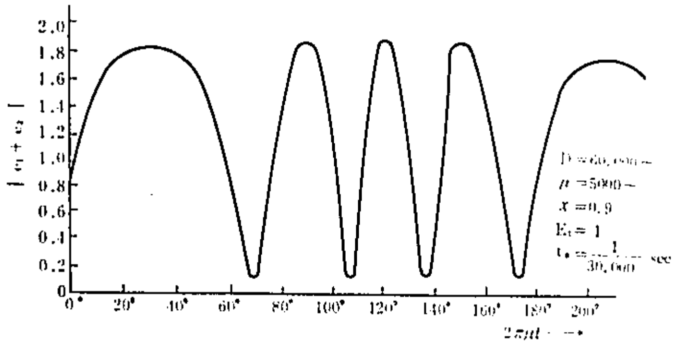


Figure 3

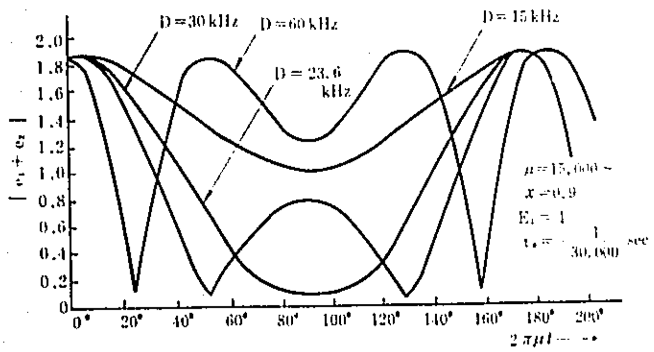


Figure 4. Variations in relation to frequency deviation

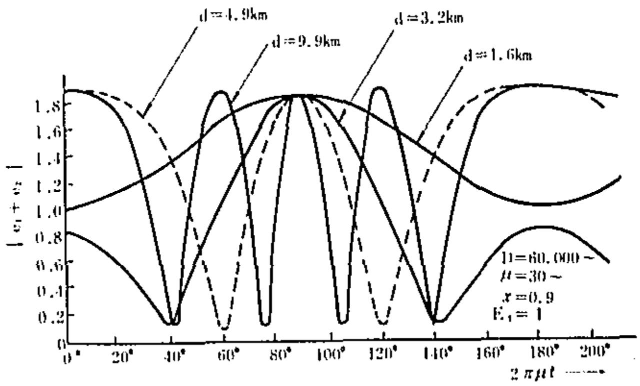


Figure 5. Variations in relation to delay time

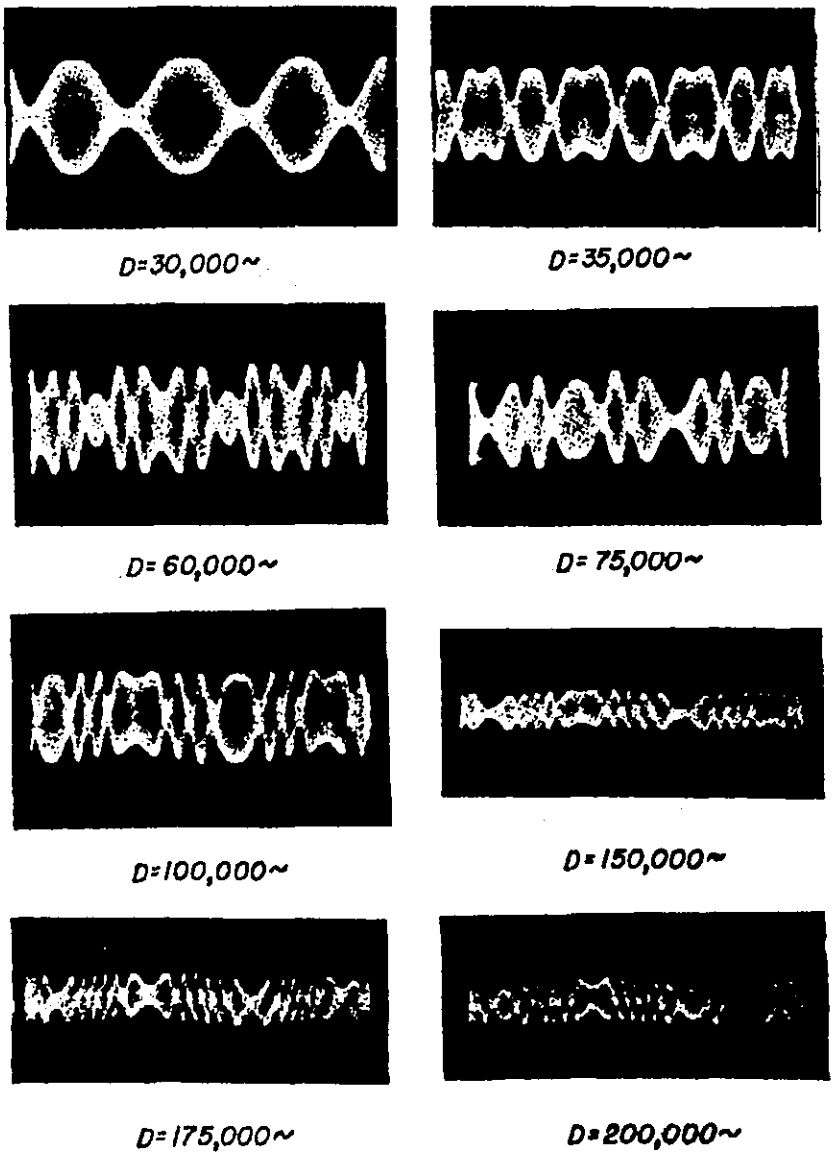


Figure 6. Oscilloscope Wave Form

4. Effects of Interference on Detector Output

Since the detector output is in direct ratio with the instantaneous frequency, it may be expressed as follows.

$$f = \frac{1}{2\pi} \frac{d}{d_t}\left[\omega_t + \frac{D}{\mu}\sin 2\pi\mu t - \tan^{-1}\frac{x\sin\{Z\cos(2\pi\mu t - \pi\mu t_0) + \omega t_0\}}{1 + x\cos\{Z\cos(2\pi\mu t - \pi\mu t_0) + \omega t_0\}}\right]$$

$$= \frac{\omega}{2\pi} + D\cos 2\pi\mu t + \frac{2D\sin\pi\mu t_0 \sin(2\pi\mu t - \pi\mu t_0)}{\dfrac{1/x + \cos\{Z\cos(2\pi\mu t - \pi\mu t_0) + \omega t_0\}}{x + \cos\{Z\cos(2\pi\mu t - \pi\mu t_0) + \omega t_0\}} + 1} \qquad (10)$$

The second term of the above equation expresses the fundamental wave and the third term, the amount of distortion. If we expand the third term to Fourier figures, it will be as follows.

$$\text{Third Term} = -2\mu \sum_{n=0}^{\infty} (2n+1)(-1)^n G(2_n + 1, Z, x, \omega t_0)\sin(2_n + 1)\gamma$$

$$-2\mu \sum_{n=1}^{\infty} (2n)(-1)^n S(2n, Z, x, \omega t_0)\sin 2n\gamma \qquad (11)$$

In this equation $\gamma = 2\pi\mu t - \pi\mu t_0$

$$G(m, n, x, \theta) = \sum_{s=1}^{\infty} \frac{(-x)^s}{s} Jm(sn)\cos s\theta$$

$$S(m, n, x, \theta) = \sum_{s=1}^{\infty} \frac{(-x)^s}{s} Jm(sn)\sin s\theta$$

and the third term contains both the fundamental wave and the higher harmonic components.

The fundamental wave of the detector output is therefore a composite of the original fundamental wave in the second term and, the fundamental wave of the third term and, at times, may be far greater than the original value.

Examples of distorted detector output wave forms are shown in Figures 7 - 9. Although the spikes indicate that higher-order harmonics are present, as the bandwidth of actual receivers are limited, these peaks are actually flattened.

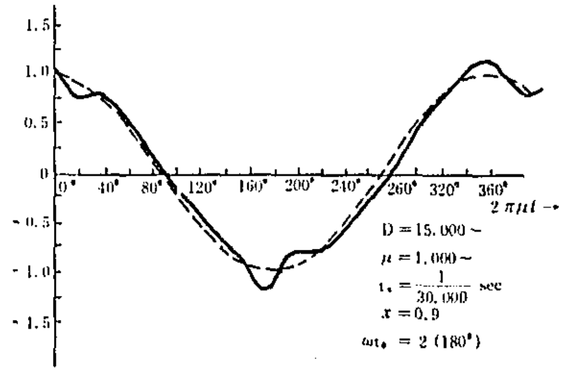


Figure 7

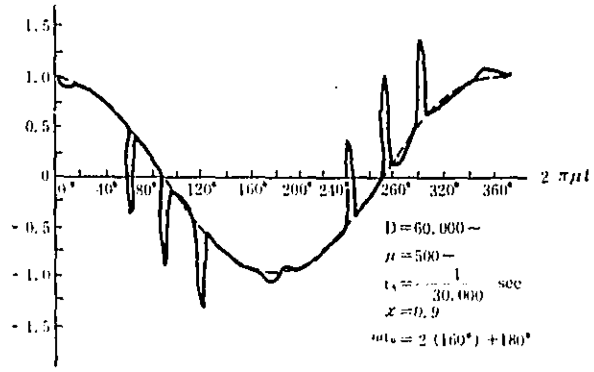


Figure 8

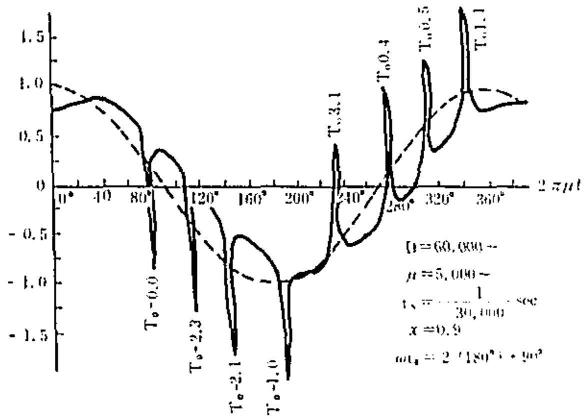


Figure 9

The detector output of a two-wave composite wave applied a delay time of 16.2μs is shown in the oscilloscope photos in Figure 10.

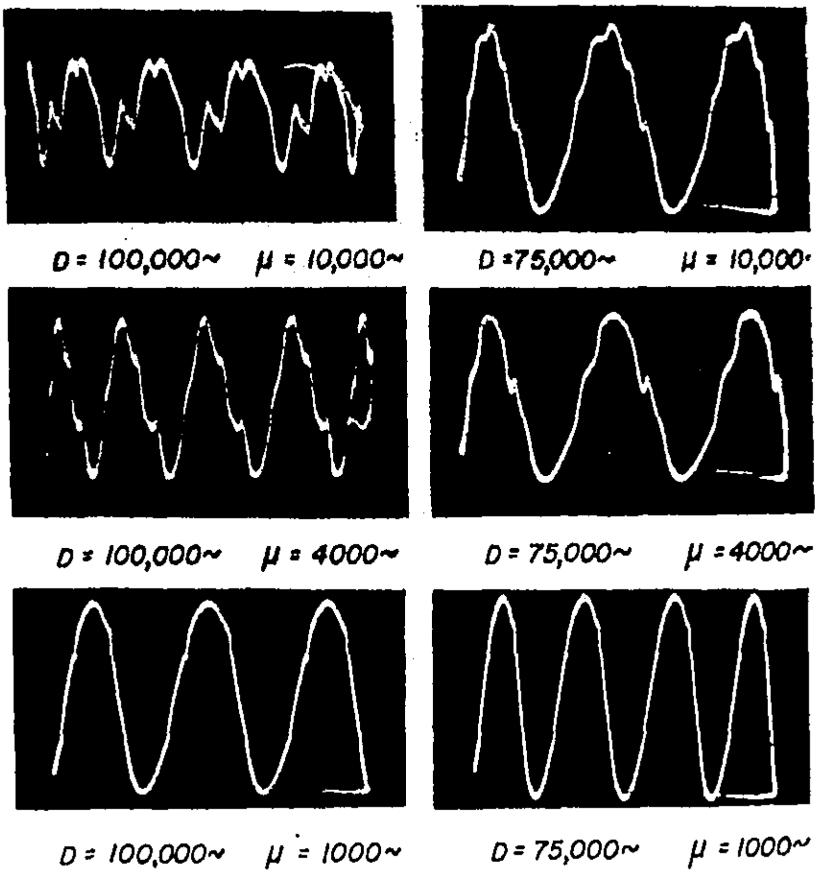


Figure 10. Oscilloscope Wave Forms

5. Conclusion

(1) As explained in the foregoing, it has been proved theoretically and experimentally that reception-distortion exists in FM waves due to the multi-path propagation. The cause of this distortion is due to the fact that the FM waves are subjected to amplitude modulation, the audio signal of the two FM waves will be in same phase and opposite phase, within a cycle and, as a result, it will be amplification moduated and peaks and valleys will occur. They will be a sharp instantaneous irregular fluctuation in frequencies, and also become the distortion component of FM.

(2) The variation of amplitude are generally eliminated by a limiter in the FM receiver, but noise will be added if the limiting action is insufficient or when the valley of the amplitude is deep.

(3) Distortion caused by multi-path propagation of FM waves contains a large amount of high-order harmonics. This is a peculiarity of FM and differs from AM distortion. Distortion generally tend to increase when the frequency of the modulating signal becomes high or when the frequency deviation increases.

(4) When two FM waves are arriving from different directions, distortion may be reduced by employing a sharp directional receiving antenna. However, when the FM waves arrive from the same direction, there will be a limitation. In selecting receiving points for rebroadcasting stations, special care should be taken into consideration with regard to multi-path propagation.

Reference Material

(1) Murray G. Crosby "Frequency Modulation Propagation Characteristics" P.I.R.E. Vol. 24 June 1936

(2) Murray G. Crosby "Observations of Freq. Modulation Propagation on 26 Mc. P.I.R.E. Vol. 29 July 1941

(3) Murlan S. Corrington "Frequency-modulation Distortion caused by Multipath Transmission" P.I.R.E. Dec. 1945

(4) S.T. Meyers "Nonlinearity in FM Radio Systems due to Multipath Propagation" P.I.R.E. May 1946

(5) Takahashi, Kurakake "Distortion of Reception of Multi-propagation Path VHF-FM Broadcast Waves" Japan Broadcasting Corporation Technical Research November, 1960

Supplement 6. ABNORMAL PROPAGATION OF VHF WAVES

In the VHF band, particularly in the FM broadcast frequencies, long distance propagation caused by ionoshere will become evident, from spring to the end of summer, centered around the summer solstice. This kind of interference caused by abnormal propagation affects not only rebroadcasting of broadcast waves, but also to the ordinary FM audiences. Therefore, in rebroadcasting broadcast waves, it will be necessary not only to consider the selection of frequencies to avoid interference at receiving site of its own transmitter but also to avoid interference from abnormal propagation.

The level of the interference wave arising from abnormal propagation becomes considerably high during particular periods or times. A general outline of their characteristics is given below.

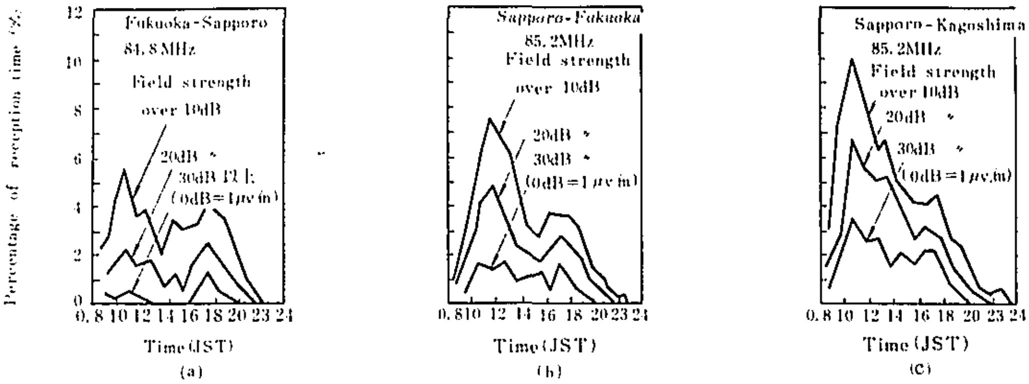


Figure 1. Examples of daily variation characteristics of percentage of reception time

(i) The first maximum in the daily variation characteristics of field strength reception is between 10 and 12 a.m. and the second maximum is between 5 and 7 p.m. An example of these characteristics is shown in Figure 1.

(ii) Continuation time
The average reception time (field strength exceeding 20 dB) for the propagation path between Sapporo and Kagoshima or Fukuoka, is about 50 minutes per day during July. But in case abnormal propagation occurs, it will continue for about two hours.

(iii) Distance characteristics of abnormal propagation
Abnormal propagation tends to develop over distance of about 1,600 km. The geographic location of Hokkaido and Kyushu is therefore most conductive to development of abnormal propagation.

Reference Literature

(1) Technical Research Monthly Japan Broadcasting Corporation, October 6, 1963.

(2) FM Stereophonic Broadcasting Technique, 1965, Takahashi, Fujita

(3) Broadcasting Technique (July 1961), Hirano

(4) Technical Research Monthly - P. 23, March 10, 1967, Japan Broadcasting Corporation - Kurakake

(5) Technical Research - P. 27, January 19, 1967, Kurakake, Ikeda

(6) Japan Broadcasting Corporation's Technical Handbook Supplement Volume

(7) Technical Research Monthly - P. 21, December 7, 1965, Japan Broadcasting Corporation - Soejima