FM Direct Stereo Decoder*

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In the conventional FM stereo receiver an antibirdie noise filter is required to eliminate beat noise due to the harmonics of the switching signal and the adjacent broadcast signal. This filter causes deterioration in distortion and separation characteristics. It has been difficult to obtain a receiver having good anti-interference capability and high-fidelity sound at the same time. In the new system these problems are solved by switching the sinusoidal subcarrier directly with a pulse train which is derived from the FM signal.

0 INTRODUCTION

Today the sound quality in FM stereo broadcasting is excellent because of the development of circuitry and devices in both transmitters and receivers. In receivers the improvement in signal-to-noise ratio, distortion, and left–right channel separation has approached 20 dB over the past 10 years. However, some of these improvements have been largely concealed by noise due to beat notes generated by adjacent-channel interference, radio-frequency intermodulation, and multipath signal interaction.

1 CONVENTIONAL STEREO RECEIVER

In conventional stereo receivers the FM broadcasting carrier is detected and converted to a composite signal. Most stereo decoders reproduce the 38-kHz subcarrier by phase-locked loop, and the stereo signal is decoded by switching the composite signal synchronously. The stereo composite signal involving the spurious signal $C(t)$ is given by

$$C(t) = (L + R) + (L - R)\sin\omega_s t + P\sin\frac{\omega s}{2}t + X_n \sin\omega_n t \qquad (1)$$

where

L = left-channel audio signal
R = right-channel audio signal
P = amplitude of pilot carrier
X_n = amplitude of spurious signal
ω_s = subcarrier angular frequency, = $2\pi \times 38$ kHz
ω_n = angular frequency of spurious signal

and the switching signal $S(t)$ is given by

$$S(t) = \tfrac{1}{2} + \frac{2}{\pi}(\sin\omega_s t + \tfrac{1}{3}\sin 3\omega_s t + \tfrac{1}{5}\sin 5\omega_s t + \cdots) . \qquad (2)$$

Then the audio output $A_0(t)$ is

$$\begin{aligned} A_0(t) &= C(t)S(t) \\ &= \tfrac{1}{2}(L + R) + \frac{1}{\pi}(L - R) \\ &\quad + \frac{X_n}{3\pi}\cos(3\omega_s - \omega_n)t \\ &\quad + \frac{X_n}{5\pi}\cos(5\omega_s - \omega_n)t + \cdots . \end{aligned} \qquad (3)$$

Thus the undesired spectra $(3\omega_s - \omega_n)$, $(5\omega_s - \omega_n)$, . . . , appear in the audio frequency range and are usually called birdie noise. In order to avoid this interference, an antibirdie noise filter is inserted between the FM detector and the stereo decoder. However, this filter is not ideal so that the amplitude and the delay time are not flat over the required bandwidth. Then the separation characteristic at high frequency worsens because of the phase and the amplitude difference between main channel and subchannel. Fig. 1 shows the separation characteristic of the FM receiver employing

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this filter, and Fig. 2 illustrates the stereo distortion characteristic.

2 NEW STEREO DECODER

A block diagram of the direct stereo decoder is shown in Fig. 3. The intermediate frequency of 10.7 MHz is converted to 1.26 MHz in order to meet the operating speed of switching devices and the one-shot multivibrator. A trigger circuit detects the zero-crossing point of the FM signal and generates the trigger pulse. In response to this trigger pulse, a one-shot multivibrator generates a constant-width pulse train. In the conventional pulse-counting detector audio signals are repro-

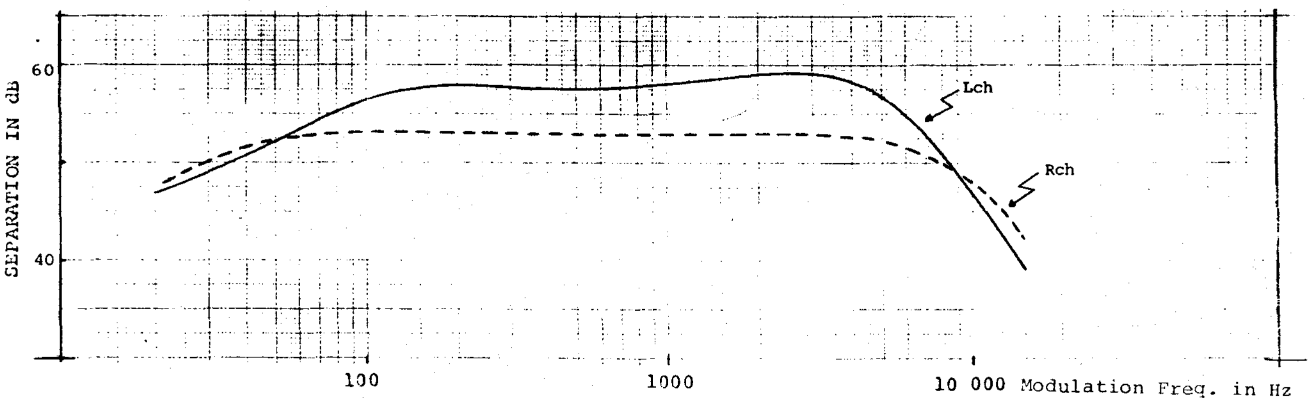

Fig. 1. Stereo separation versus modulation frequency. Carrier frequency 98 MHz; antenna input level 80 dBf; deviation: $L + R$ = 67.5 kHz, pilot 7.5 kHz.

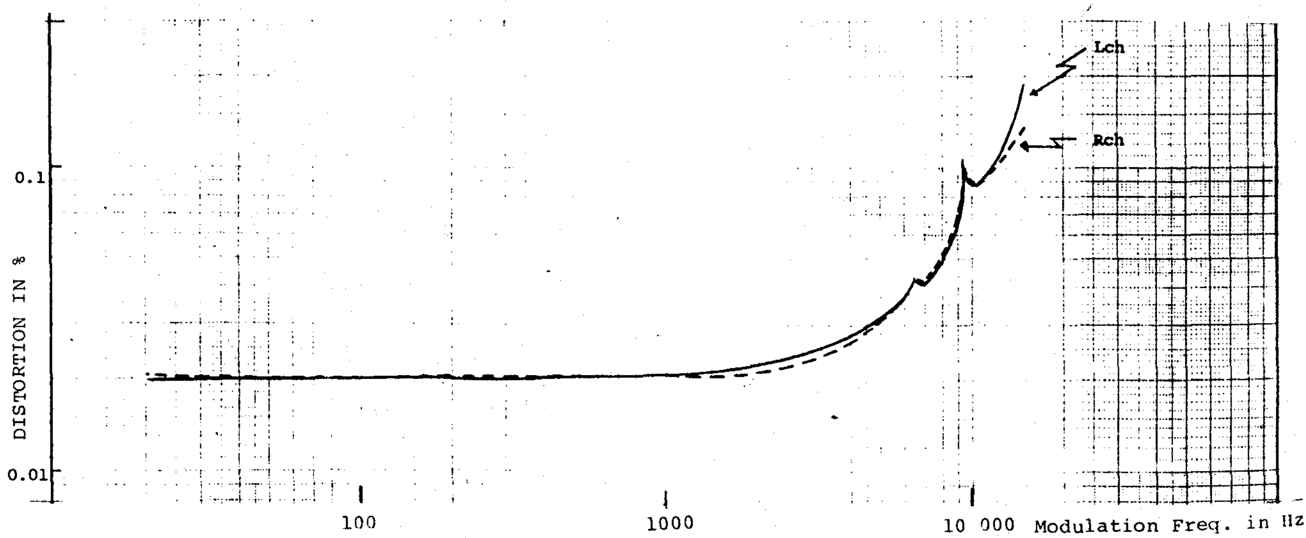

Fig. 2. Stereo distortion versus modulation frequency in conventional system. Carrier frequency 98 MHz; antenna input level 80 dBf; deviation: $L + R$ = 67.5 kHz, pilot 7.5 kHz.

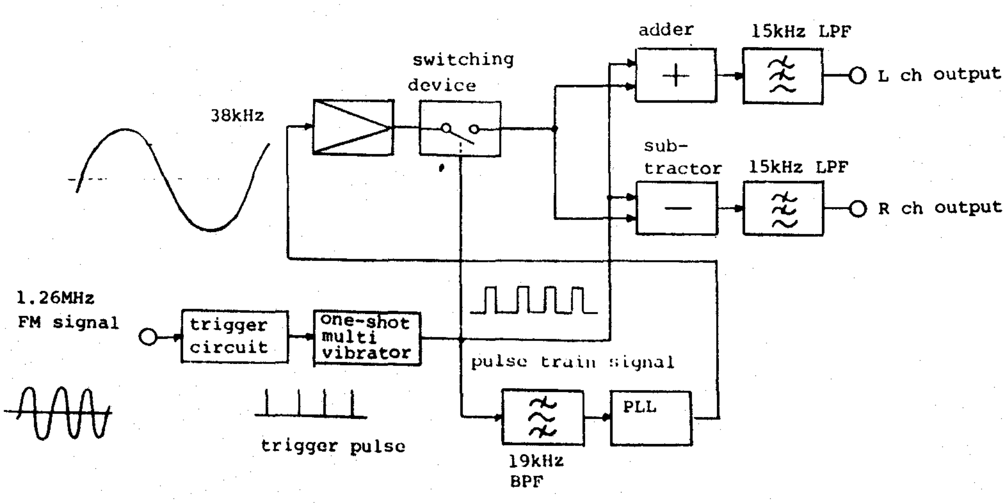

Fig. 3. Block diagram of direct stereo decoder.

duced by integrating this pulse train, while in the direct stereo decoder this pulse train is not integrated but is used as a control signal for the switching devices that switch the 38-kHz subcarrier corresponding to the high and low levels of the pulse train. This subcarrier is a sinusoid rather than the rectangular wave used in the conventional stereo decoder. This switched subcarrier and pulse train are added and pass through a 15-kHz low-pass filter to obtain left-channel output. Subtracting the switched subcarrier from the pulse train recovers right-channel information. Fig. 4 shows the spectrum of the pulse train, which includes spurious signals. Composite signals exist in the lower frequency region and spurious signals exist around 100 kHz, with the rest of the spectrum existing around 1.26 MHz and its harmonics. The product of the 38-kHz subcarrier and each of the spectra in Fig. 4 can be written as

$$U(t)S \sin \omega_s t$$

where

$$\begin{aligned} U(t) &= \text{function of pulse train} \\ S \sin \omega_s t &= \text{38-kHz subcarrier} . \end{aligned}$$

Therefore the output of the adder $A(t)$ is

$$A(t) = (1 + S \sin \omega_s t)U(t) . \tag{4}$$

Because the pulse train is multiplied by the sinusoidal subcarrier, the spectrum around 1.26 MHz and its harmonics and spurious signals do not affect the audio signal. Thus the beat noise is not generated. Consequently the spectrum components of the pulse train are considered to be the composite signal only. If we assume the amplitude of subcarrier S to be 2, then $A(t)$ becomes

$$\begin{aligned} A(t) = {} & 2L + 2P \sin\left(\frac{\omega_s}{2} t + \frac{\pi}{4}\right) \\ & + (3L + R) \sin \omega_s t - P \cos \frac{3\omega_s}{2} t \\ & - (L - R) \cos 2\omega_s t . \end{aligned} \tag{5}$$

After filtering Eq. (5), the left-channel audio signal $2L$ is obtained. Right-channel audio signal $2R$ is obtained in a similar manner. Since the sinusoidal subcarrier is applied in the new decoder, the construction of phase-locked-loop circuitry is different from that of conventional circuits. A block diagram of the new phase-locked loop is shown in Fig. 5. The output frequency of the voltage-controlled oscillator is 76 kHz and is counted down to 38 kHz at the frequency divider. A band-pass filter converts the rectangular wave to a sinusoidal wave. The sinusoidal wave is again converted to a rectangular wave by the zero-crossing detector and counted down

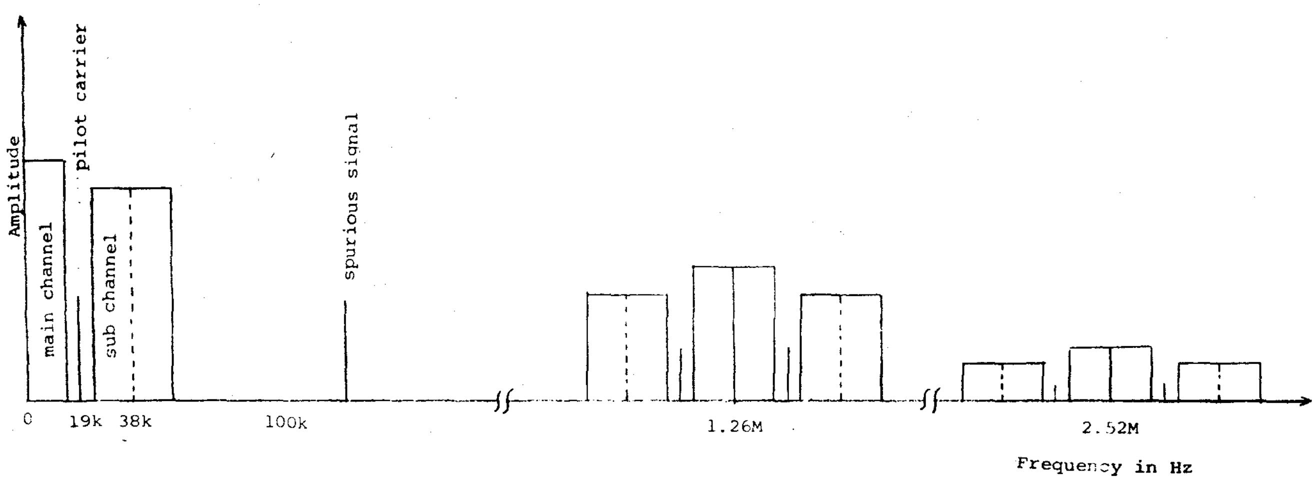

Fig. 4. Spectrum of modulated pulse train.

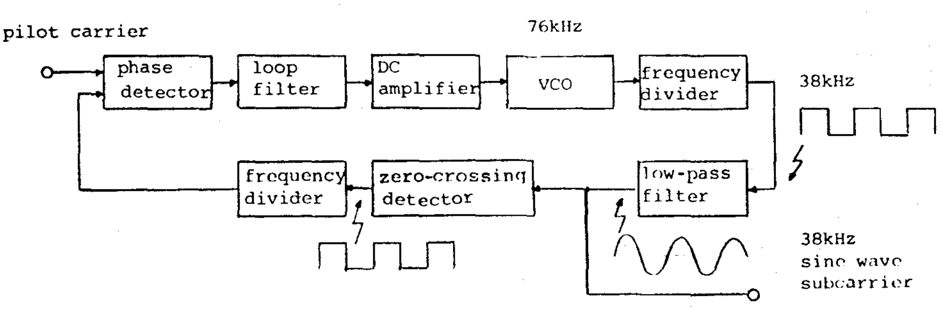

Fig. 5. Block diagram of phase-locked-loop circuit for new system.

to 19 kHz. The phase difference between this 19-kHz signal and the pilot carrier in the pulse train is detected at the phase detector. The phase-locked loop is completed by feeding the output of the phase detector through the loop filter and dc amplifier to the voltage-controlled oscillator. The deterioration of the separation characteristic due to the phase error is eliminated since the drift of the bandpass filter in the phase-locked loop is corrected.

3 DEVELOPMENT OF THE NEW SYSTEM INTEGRATED CIRCUIT AND ITS CHARACTERISTICS

Two system integrated circuits, PA5006 and PA5007, have been developed in order to obtain high performance and reliability in the new system. Fig. 6 shows the block diagram of these new circuits. PA5006 is a 22-pin circuit which involves trigger circuit, one-shot

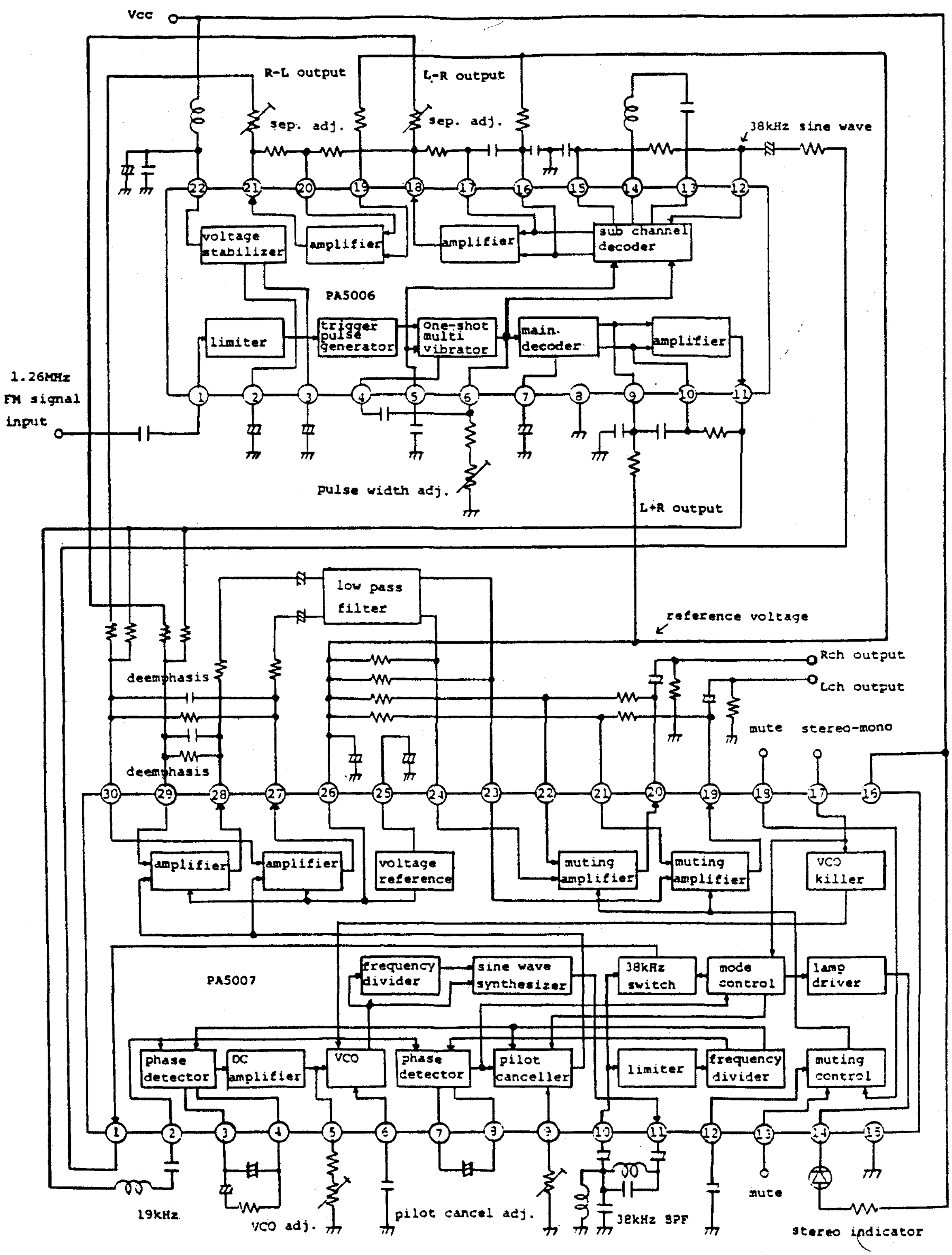

Fig. 6. Block diagram of integrated circuits for new system.

multivibrator, main- and subchannel decoder, and amplifiers. For the switching devices, emitter-coupled differential circuitry is used in order to meet the high speed required. Feeding the 1.26-MHz FM signal to pin 1 and the 38-kHz sinusoidal subcarrier to pin 12 results in the main-channel and subchannel signals appearing at pins 11 and 18, respectively. PA5007 is a 30-pin circuit which involves amplifiers, muting circuits, a pilot cancellation circuit, and a phase-locked loop that generates the sinusoidal subcarrier to be used in PA5006. A 38-kHz subcarrier is generated by a sinusoidal synthesizer and passed through a tank circuit in order to purify it. The muting circuit suppresses the pop noise and the interstation noise. The pilot cancellation circuit eliminates the pilot carrier in order to improve the audio signal frequency response. Figs. 7–10 show the characteristics of an FM receiver using these system integrated circuits. Fig. 7 illustrates distortion versus modulation frequency and Fig. 8, separation versus modulation frequency. Both performances are close to the limit of the measuring equipment. Fig. 9 shows the spurious response compared with the desired one. Spurious response around 114 and 190 kHz can be neglected in the new system. However, high-level response due to the harmonics of the switching signal is observed in the conventional system. Fig. 10 shows the stereo signal-to-noise ratio versus desired input signal level when adjacent channel interference exists. The interference signal is 200 kHz from the desired signal and is modulated with white noise at ±75 kHz deviation. A CCIR peak-indicating meter is used to measure the signal-to-noise ratio. The signal-

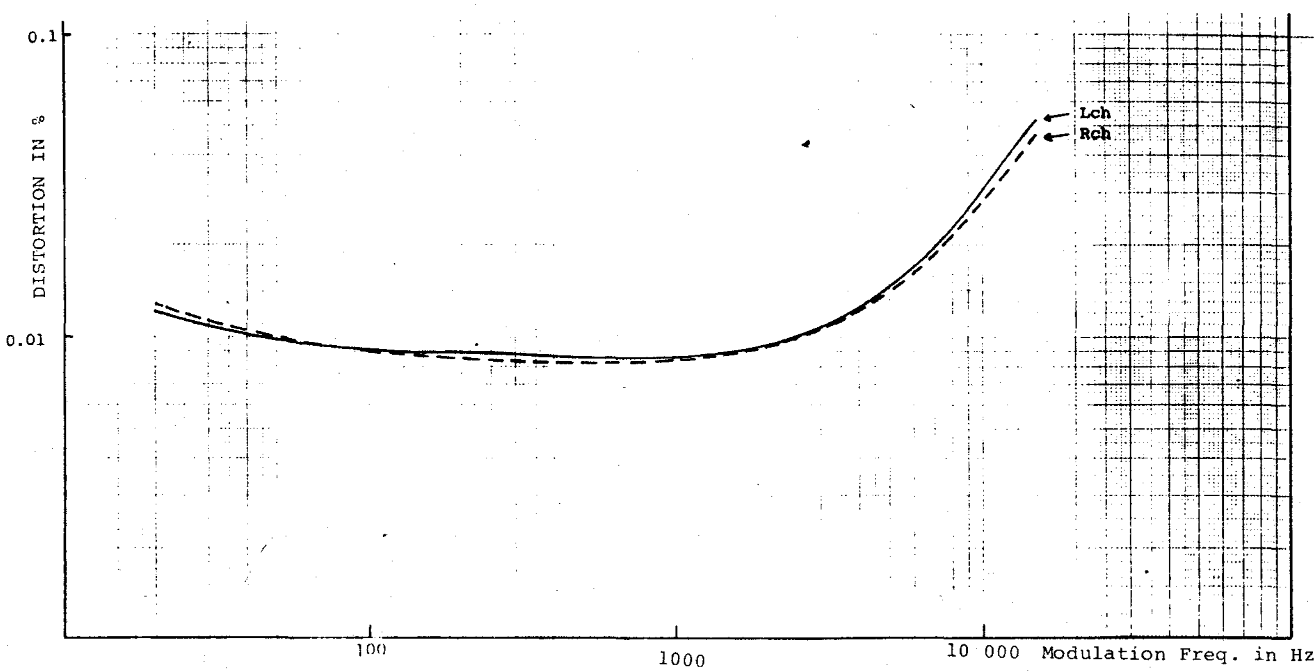

Fig. 7. Stereo distortion versus modulation frequency in new system. Carrier frequency 98 MHz; antenna input level 80 dBf; deviation: $L + R = 67.5$ kHz, pilot 7.5 kHz.

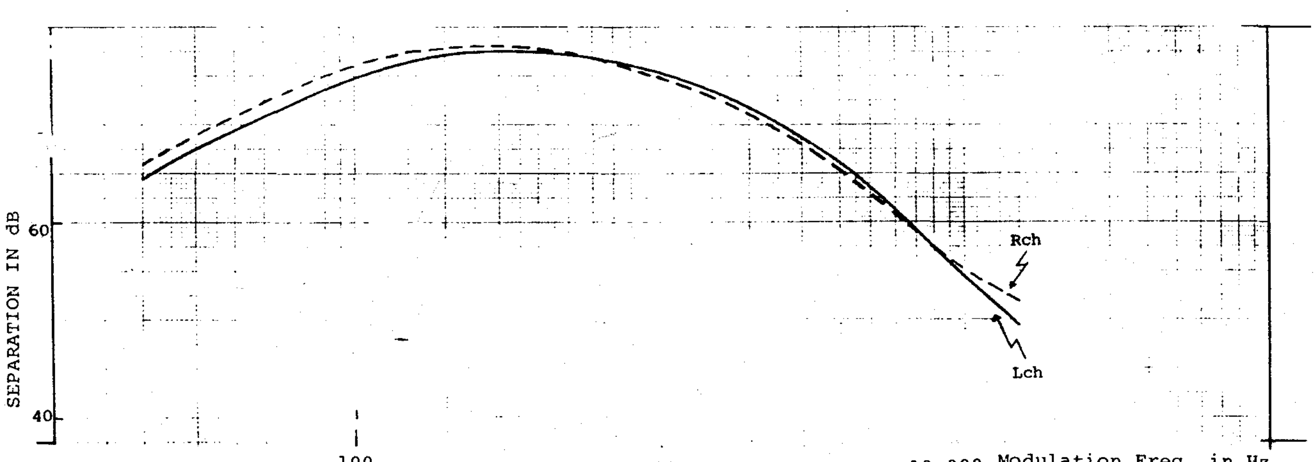

Fig. 8. Stereo separation versus modulation frequency in new system. Carrier frequency 98 MHz; antenna input level 80 dBf; deviation: $L + R = 67.5$ kHz, pilot 7.5 kHz.

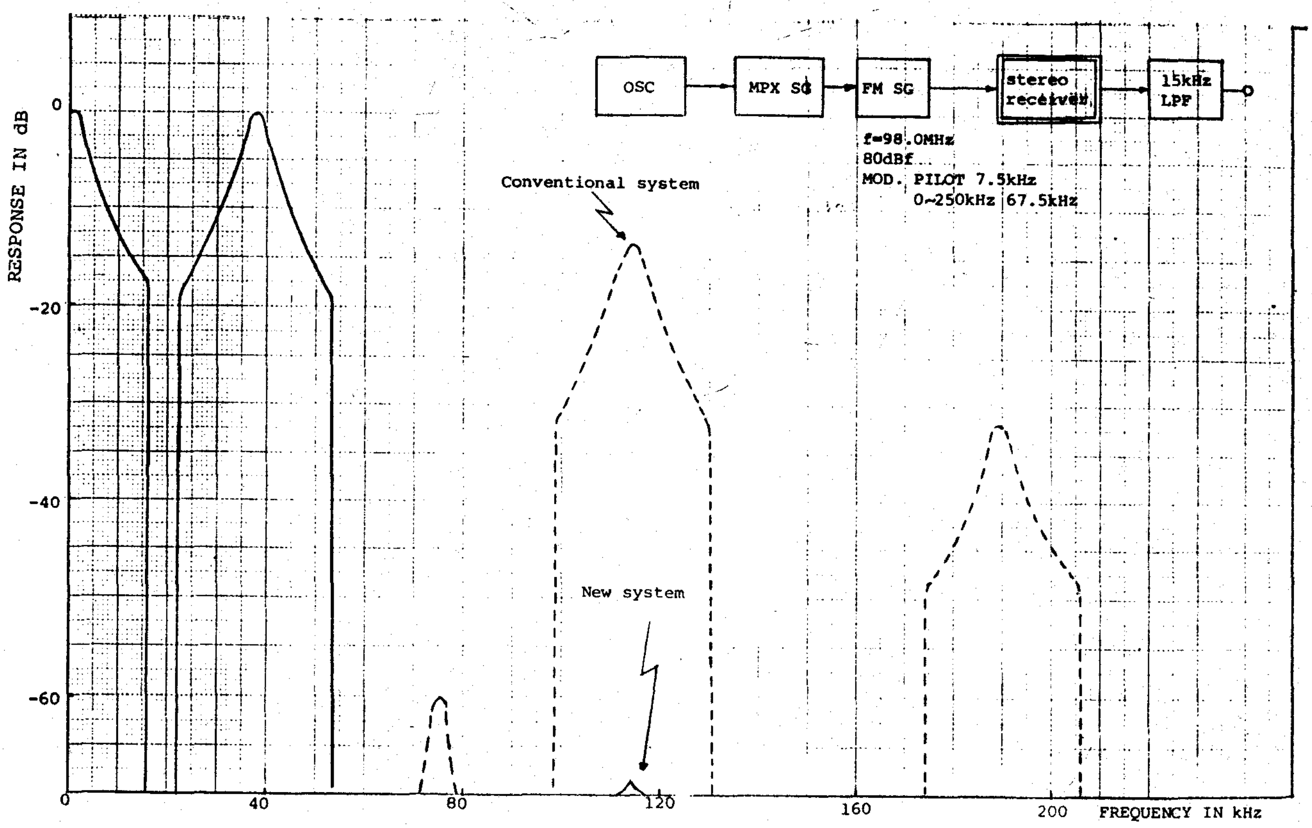

Fig. 9. Stereo spurious response.

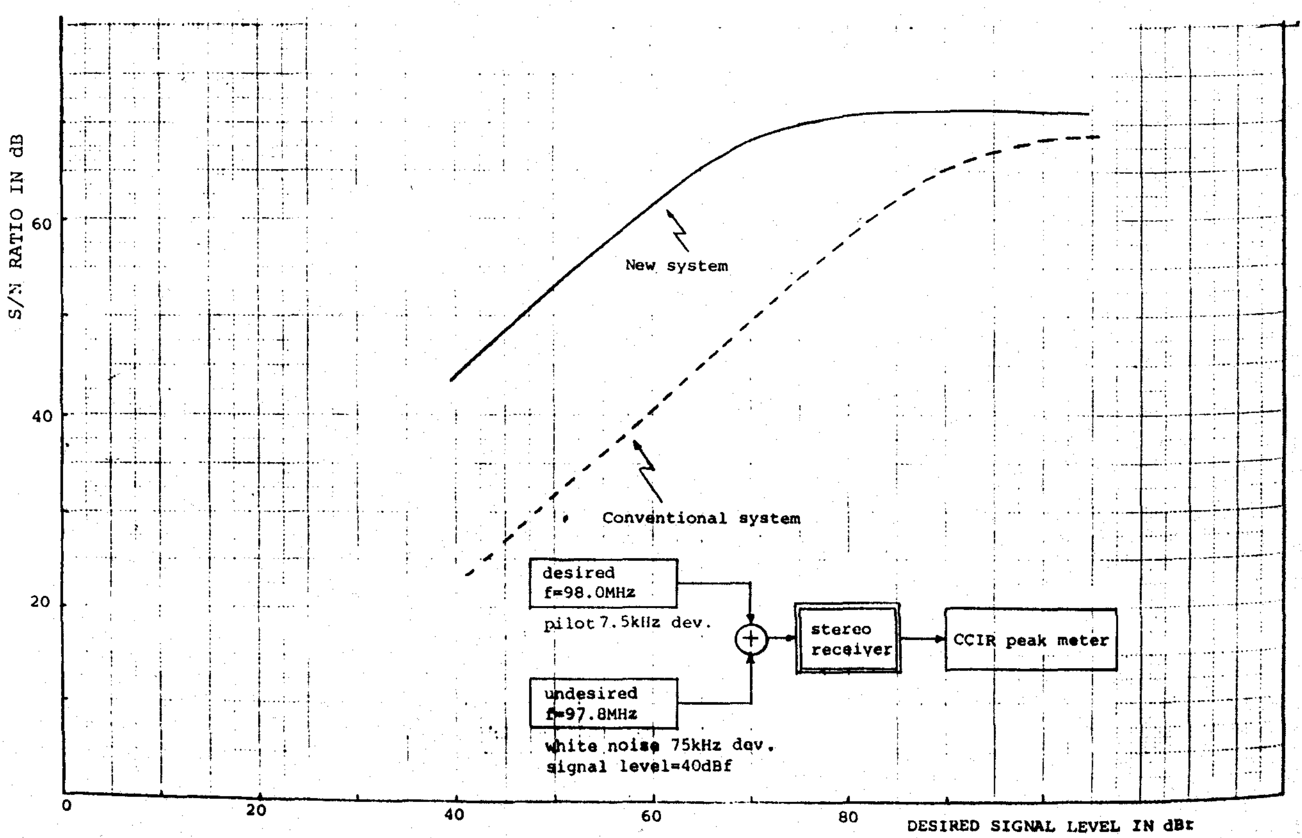

Fig. 10. Stereo signal-to-noise ratio in adjacent channel interference.

to-noise ratio in the new system is improved by more than 20 dB compared with that in the conventional system.

4 CONCLUSION

In the conventional receiver, stereo signals are decoded by switching the composite signal with a rectangular subcarrier, whereas in the new direct stereo decoder, stereo signals are decoded by switching the sinusoidal subcarrier with the pulse train which has information from the composite signal. The beat noise caused by the harmonics of the switching signal is eliminated and, accordingly, no antibirdie noise filter is required. Moreover the switching devices are operated at 1.26 MHz, so their nonlinearity does not affect the decoded audio signal. Thus in the new FM direct stereo decoder, anti-interference and high-fidelity reception are realized at the same time.

5 REFERENCES

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