APPLICATION OF WALSH FUNCTIONS TO AN FM STEREO DEMODULATOR

2031 (B-6)

Susumu Takahashi and Hiroshi Iida
Sansui Electric Company, Ltd.
Tokyo, Japan

**Presented at the 74th Convention
1983 October 8-12
New York**

AN AUDIO ENGINEERING SOCIETY PREPRINT

APPLICATION OF WALSH FUNCTIONS TO AN FM STEREO DEMODULATOR

By

Susumu Takahashi Hiroshi Iida

Sansui Electric Co., Ltd.

Tokyo, Japan

Abstract

In the conventional FM stereo demodulator, interference from adjacent channels is demodulated by the harmonics of a square wave switching signal resulting in a high pitched whistle type noise. In our new demodulator such noise is not generated because the switching square waves are derived from Walsh functions. We can therefore eliminate the low-pass filter, producing a phase-linear composite signal characteristic with high separation and low distortion.

1. Introduction

Most modern FM stereo decoders are square wave switching type and they make use of a phase-locked loop (PLL). The left and right audio channels are obtained by switching the composite signal with a 38kHz square wave produced in the PLL circuit. A simple double balanced mixer is employed for this purpose. While the decoder itself is not subject to harmonic distortion or deterioration in signal-to-noise ratio, it suffers from "beat" problems. The switching square wave contains not only the fundamental frequency but also its harmonics. Usually the composite signal contains frequencies close to the harmonics of the switching signal due to interference from adjacent channels. These adjacent channel frequencies when decoded, deteriorate the s/n ratio and cause beat interference making listening extremely unpleasant. In order to avoid such phenomenon, a low-pass filter often referred to as an "anti-birdie" filter, is used to filter out the unwanted high frequency components in the composite signal before demodulation. The use of this low-pass filter however creates phase errors between the main and sub-channels which degrades stereo separation.

Various demodulation methods have been developed to avoid such degradation. One method is to use a sinusoidal wave instead of a square wave as the switching signal. Such a decoder demodulates only the desired frequency range, thus no beat interferenced problem is encountered. Drawbacks of this method, are that it requires an analog multiplier, making the decoder expensive. This technique still falls short of performance in THD, s/n ratio and dynamic range because of analog processing. We are yet to see a commercial application of this demodulation method.

Another demodulation method is to eliminate the anti-birdie filter by cancelling beat frequency components. The composite signal is separately switched by the third-order harmonic of the 38kHz (114kHz) while it is demodulated by the 38kHz switching signal [1]. This method is good for cancellation of the beat frequency at 114kHz, but no other frequencies are cancelled.

Yet another method known to us is to generate a 38kHz sinusoidal wave from a composite signal derived from a pulse-counting type FM detector. Demodulation is accomplished by switching the sine wave by the output pulses of the pulse-counter [2]. This is a very effective method of eliminating the beat problems and avoids performance degradation of a tuner. One drawback however, is that the detector and the decoder have to be combined, making its application rather limited.

In this paper, we describe a new demodulation system which does not produce the "birdie" noise found in conventional processing. Our technique still uses square waves as switching signals but multiples of different width pulses are derived from Walsh functions. Two switching circuits are employed where the composite signal is switched by 2 different square waves based on Walsh functions, resulting in virtually no harmonics up to the 6th-order of the 38kHz. In practice, this is equivalent to a composite signal multipled by a sinusoidal wave but, we still retain the advantages of digital processing. This new demodulation system makes receivers immune to interference from adjacent channels up to 228kHz apart including the most harmful harmonic at 114kHz. Unlike the demodulation method referenced above, the new demodulator can be used independently of the detector.

2."Birdie" Noise Generation Mechanism

Due to the harmonics of the switching square wave, conventional demodulators suffer "birdie" noise. The noise is generated as follows.

Fig. 1a shows a frequency spectrum of an FM detected composite signal when a desired and adjacent channels are FM modulated. Suppose the adjacent channel is located 200kHz away from the desired station and is deviated by $\pm$ 75 kHz. The desired station is also deviated by $\pm$ 75kHz. The relative frequency deviation between the two channels becomes $\pm$ 150 kHz, making wide frequency range of the detector output subject to beat problems. In practice, the high frequency area is attenuated by the IF filter and the frequencies in the range of about 50kHz to 200kHz in Fig. 1a (shaded area) become the beat components. These frequencies are outside the audible range and an ideal stereo demodulator does not cause any problem. The switching square wave however,poses problems here. The odd harmonics of the square wave demodulate the ultrasonic beat components (Fig. 1b) making them audible.

Fig. 1c shows spectra that are converted to the audible frequencies. The spectrum around 38kHz is necessary for stereo demodulation but those around 114kHz and 190kHz cause unnecessary noise. Such noise varies with the amount of frequency deviation between the desired and the adjacent channels and resembles twittering of birds. Thus the name "birdie" noise.

3. New Demodulation System

Our objective was to develop a demodulator which is basically a square wave switching type, but with beat rejection capability equivalent to the sinusoidal wave demodulator. This is achieved by applying the Walsh functions to conversion of a sine wave into square waves for switching.

3.1 Walsh Transform of a Sinusoidal Wave

As an arbitrary waveform can be resolved into multiples of sinusoidal waves by the Fourier transform, the Walsh transform makes it possible for such waveforms to be resolved into multiples of square waves (Walsh waves). What this means is that the original sinusoidal waveform can be reconstructed by composing the Walsh waves after adjusting their amplitude. The Walsh waves are composed of positive and negative binary values and they can be generated by a simple digital circuit. They are suitable for use as switching signals of a demodulator. Table 1 shows examples of a sine wave transformed by the Walsh functions. The first example is a sine wave transformed to the 8th-order Walsh functions, and the second example to the 16th-order.

3.2 Fourier Transform of a Walsh Wave

Let us now confirm how closely the results of the Walsh transform in Table 1 approximate a sinusoidal wave. Fig. 2 shows the Walsh waves of W1, W7 and 0.65W1-0.27W7 shown in the first example in Table 1. Fourier transform of the 0.65W1-0.27W7 reveals how accurately a sine wave is approximated. The same method is applied to the second example. From Fig. 3a it is clear that with the 8th-order Walsh transform, the 7th-order harmonic of the 38kHz, or 266kHz, is the lowest harmonic component and there are no other harmonics lower than that frequency. With the 16th-order Walsh transform (Fig. 3b) the 15th-order harmonic or 570kHz is the lowest harmonic component that exists.

4. Circuit Configuration

For practical application of the Walsh functions to the decoder we assumed that the 8th-order Walsh transform is sufficient. With this transform beat frequencies above 266kHz are demodulated, but the interfering frequencies higher than that from adjacent channels are attenuated by the IF filter. Those frequencies are further reduced in level at the FM demodulation stage, or when they are turned to a composite signal and, if necessary, a simple filter can be added for further attenuation.

4.1 **Walsh Functions and Hardware Realization**

A 3-bit counter and EXOR circuit generate the Walsh waves of the 8th-order Walsh transform (Fig. 2). Fig. 4 shows its diagram and Fig. 5, its timing chart. The input clock frequency for the counter is 8 times the fundamental (WI) and it is successively divided to obtain 1/2, 1/4 and 1/8 the clock frequency. They are denoted as W4, W2 and W1 respectively, and the following relationship is established: $W6=W4 \oplus W2$, $W7=W6 \oplus W1$ and also $W3=W1 \oplus W2$. Thus W1 to W7 waveforms can be all expressed by the combination of the three divided frequencies. The $\oplus$ indicates EXOR between the signals. When a 4-bit counter is used, Walsh waves of up to the 16th-order Walsh transform are generated and with a 5-bit counter up to the 32nd-order. W0 represents DC component in the Walsh functions.

4.2 **Actual Circuit**

With the switching type demodulator, the composite signal is multiplied by a demodulation carrier. There are two different ways for separating the left and right channels, one is called a switching method and the other a matrix method. With the former the left and right channels are obtained directly by switching,and with the latter a difference signal is first demodulated by switching before it is fed to a matrix circuit to obtain the necessary separation. The new demodulation system we have developed can be applied to either types of demodulation and the following explanation refers to the matrix method.

In Fig. 6, DBMI and DBM2 are double balanced mixers, ATT is an attenuator, ADD is an adder and MTX is a matrix circuit. The composite signal is fed to one of the inputs of DBM1 and DBM2, while their other inputs are fed with W1 and $\overline{W7}$ signals that are derived from the Walsh functions (Fig. 5). The output of DBM2 is multiplied by $1/(1+\sqrt{2})$ by ATT and its output is summed by ADD and the difference signal L-R is obtained. The difference signal and sum signal of the composite are matrixed and the left and right separation is achieved. Although it appears that the above function is similar to the conventional switching demodulator, the result of demodulation is equivalent to multiplying the composite signal by a sinusoidal wave which does not contain the 2nd- to the 6th-order harmonics (Fig. 2c), and therefore no birdie noise is generated. Fig. 7 shows a block diagram of the complete demodulator.

4.3 **Performance**

The new demodulator is compared with a conventional square wave switching demodulator for their ability to cancel birdie noise. Fig. 8

shows the measured results. The frequency response, separation and THD of the new demodulator are shown in Fig. 9.

5. Conclusion

Modern FM receivers are required to be free from beat problems. This becomes particularly important to cope with the additional channel allocations being considered, new SCA uses or FM discrete quadraphonic transmissions which require wide bandwidth. It is our belief that the new demodulation system based on the Walsh functions will demonstrate its advantages more than ever. Birdie noise level caused by the 3rd and 5th-order harmonics of the switching signal is reduced by more than 40dB compared with those of a conventional square wave switching demodulator. We have confirmed that the harmonics higher than the 7th-order do not appear in the composite signal when the demodulator is combined with an FM tuner. The demodulator is similar in its construction to the conventional PLL type multiplex demodulator and no special circuit is required other than an additional double balanced mixer and a change in VCO oscillation frequency to 304kHz. The new demodulator is quite suitable for IC integration.

6. Acknowledgement

The authors would like to express their sincere gratitude to Masaya Ishikawa of Central Research Laboratory of the company for his assistance in preparing and translating the paper.

7. References

[1] S. Inoue, Y. Iso and M. Ienaka, "High Quality FM Stereo Decoding IC with Birdie Noise Canceling Circuit", IEEE Transaction on Consumer Electronics, CE-27 No. 3, August 1981.

[2] K. Ishida and T. Numata, "FM Direct Stereo Decoder". presented at the 72nd Convention of the Audio Engineering Society, preprint no. 1933.

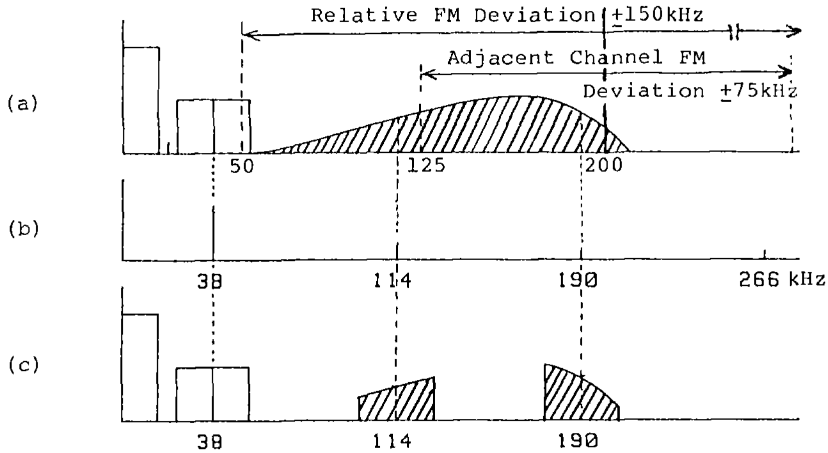


Fig. 1   Birdie Noise Generation

	8th-Order Transform	16th-Order Transform
W0	0	0
W1	0.65	1.00
W2	0	0
W3	0	0
W4	0	0
W5	0	0
W6	0	0
W7	-0.27	-0.41
W8	-	0
W9	-	0
W10	-	0
W11	-	-0.20
W12	-	0
W13	-	0.08
W14	-	0
W15	-	0

Table 1 Walsh Transformations of a Sine Wave

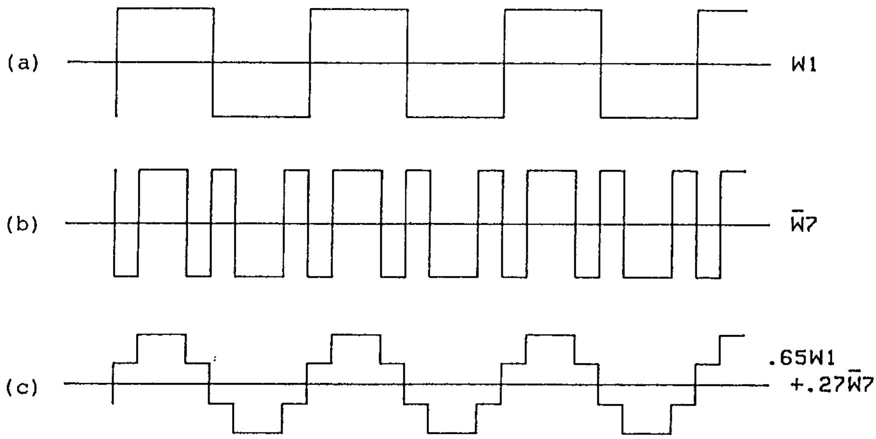


Fig. 2 Walsh Waves

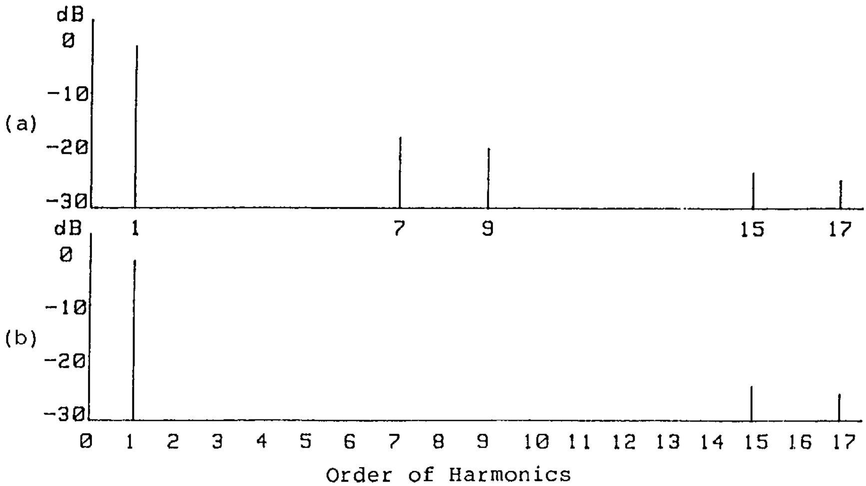
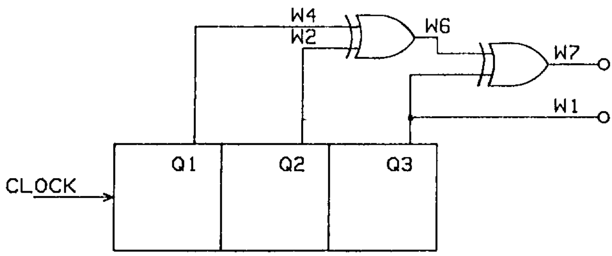


Fig. 3 Fourier Transforms of Walsh Waves



3-bit Counter

Fig. 4 Walsh Wave Generator

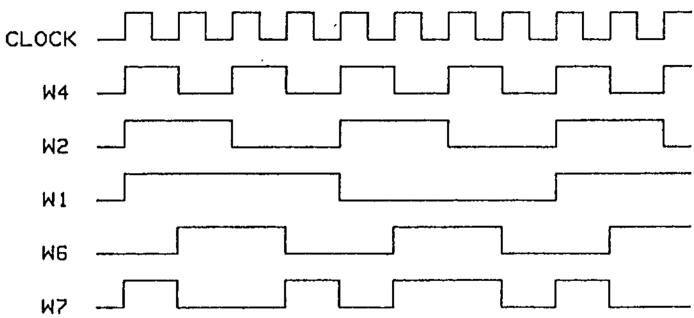


Fig. 5 Timing Chart of Walsh Waves

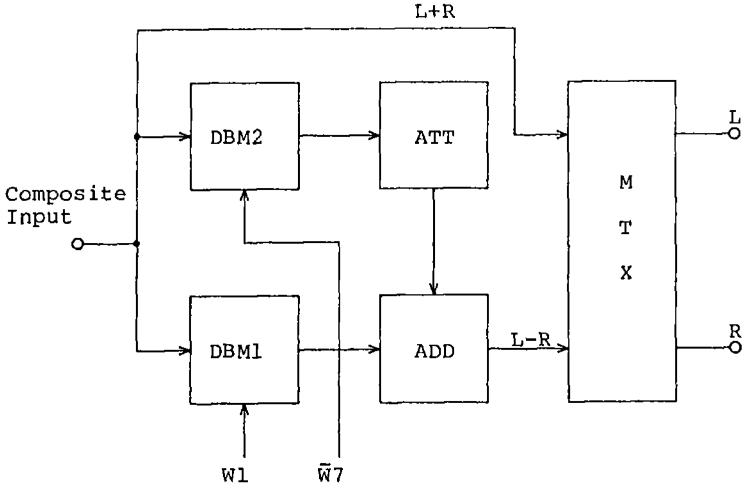


Fig. 6 Decoding Block

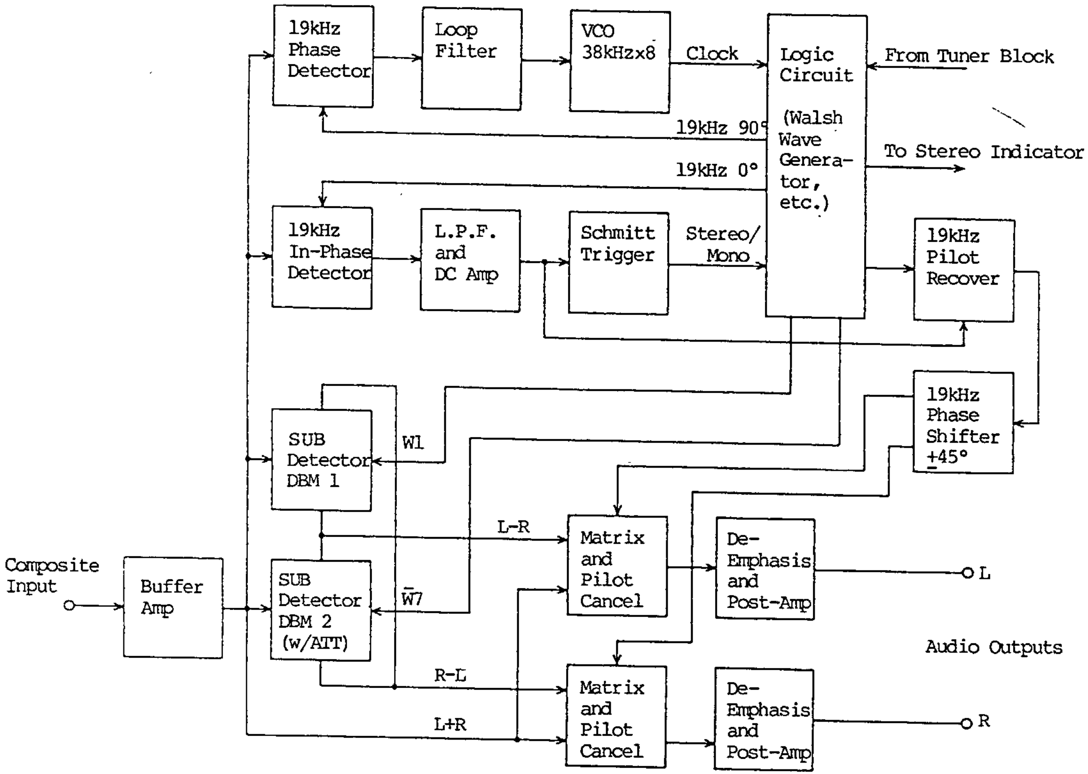


Fig. 7 Demodulator Block Diagram

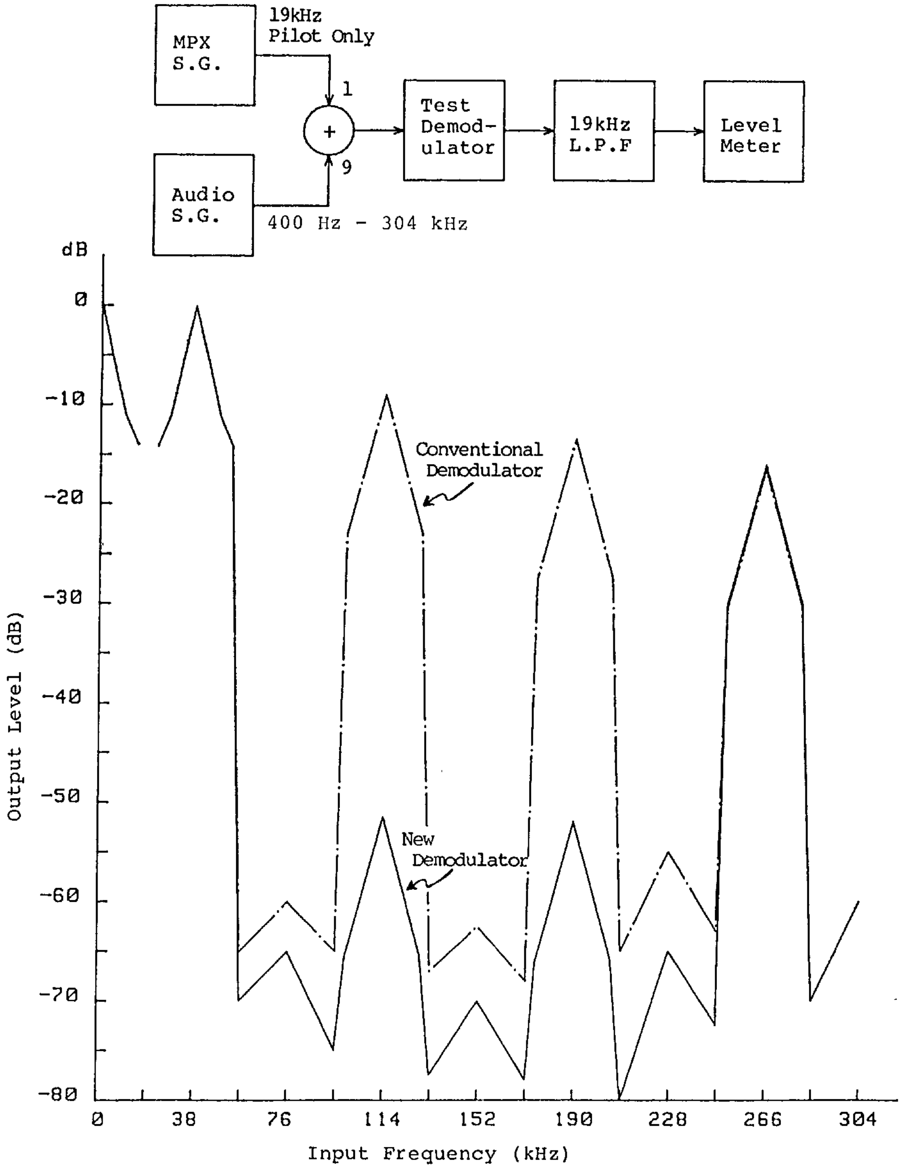


Fig. 8 Comparison of Birdie Noise Cancellation

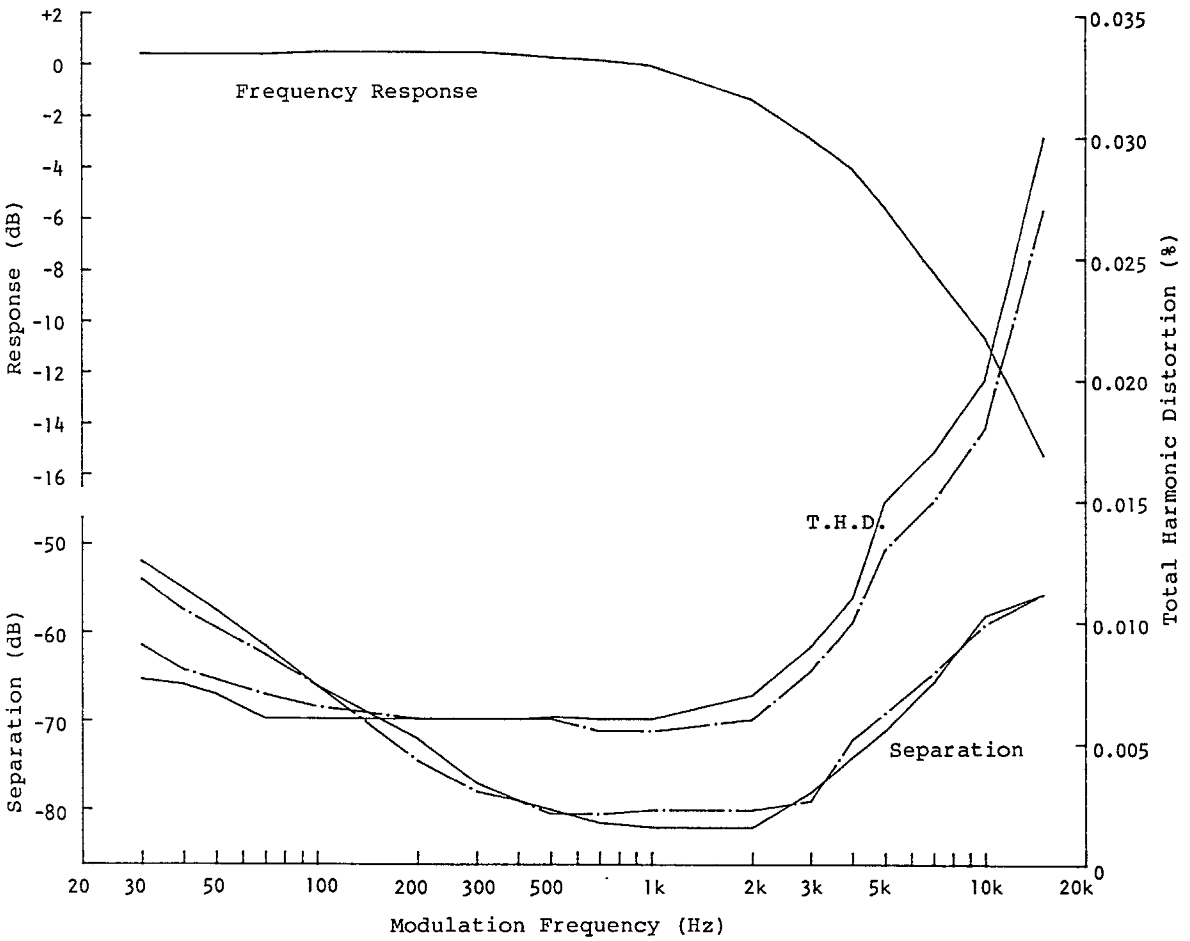


Fig. 9 Performance of the New Demodulator