AM STEREO

National Semiconductor

AM Stereo NOW with National Semiconductor's LM1981 Stereo Decoder

The industry's first AM stereo decoder, the LM1981, is being made available to designers by National Semiconductor Corporation. This exceptional device, developed using National's proprietary linear circuit technology, decodes, or separates, the AM-IF signal into left and right channels. For the first time designers are able to directly modify existing mono designs for stereo operation.

The FCC is presently re-reviewing its April 1980 selection of the Magnavox system as a standard AM stereo manufacturing approach. The LM1981 is the integral part of at least three of the five proposed systems currently under evaluation. By designing-in the LM1981 now, manufacturers can be in production quickly once the FCC decision is announced.

According to National Semiconductor applications engineers, most AM radio front ends will probably not meet the requirements for AM stereo without substantial improvements in at least two major areas:

— phase noise of the local oscillator

— IF (intermediate frequency) symmetry

National is nearing introduction of an AM radio chip that will substantially improve performance in these areas. The LM1981 AM stereo decoder is available in a 20-lead, dual-in-line molded package. The device is fully developed and ready to enter volume production. Samples are available now on a restrictive consignment basis.

CLAU— 124 (Edition B)

Product: LM1981

Topic: Compatible AM Stereo

Date Issued: October 1980

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Don Sauer

National
Semiconductor

LINEAR APPLICATION UPDATE

LM1981 AM Stereo Decoder IC

This CLAU 124 introduces the industry's first AM Stereo Decoder IC: the LM1981. The basic operation is described while decoding the Magnavox system. Also, included is a description of how to decode the Motorola and Belar systems with this same chip. We believe that the LM1981 may also be usable with the Harris and perhaps even the Kahn systems, although at this time we'll have to leave the implementation as a project for the very interested reader (who also has an encoder).

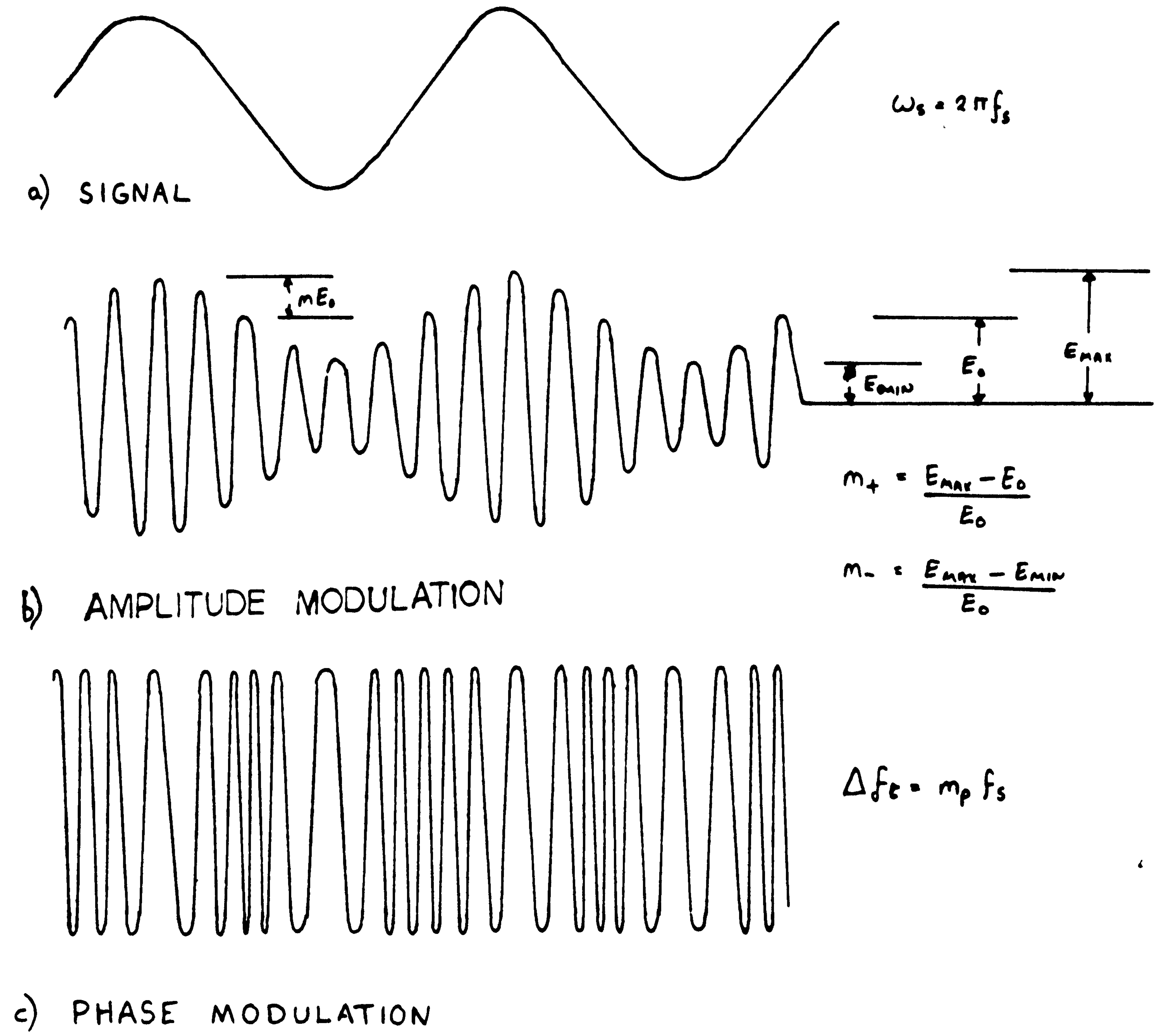

FIGURE-1. CARRIER AMPLITUDE AND ANGLE MODULATION

Introduction

Why stereo AM broadcasts? After all, FM stations have been broadcasting stereo for many years now, so what additional benefit is obtainable by putting AM signals into stereo?

For the AM broadcaster, upgrading the quality of the AM service by introducing stereo clearly has an economic impact, since stereo is already a preferred listening source for many people. Also, since the FM stereo service suffers from a limited geographic coverage line of sight transmission is needed, and multipath - the simultaneous arrival of a second, time delayed signal at the receiving antenna caused by reflection from tall buildings etc. - can cause serious distortion of the received signal. These factors can severely limit good quality FM stereo reception, as many listeners to automobile radios in particular will attest to. By introducing AM stereo, it is anticipated that much wider stereo coverage, free from multipath, will be available to the travelling listener.

The AM Stereo Signal

In the U.S. a conventional AM broadcast signal consists of a carrier frequency between 535kHz and 1605kHz that is amplitude modulated with the intelligence(!) being transmitted. The extent to which the carrier envelope varies is expressed by the degree of modulation M+ or M- as shown in Figure 1. If the modulation is symmetrical, then $M+ = M- = M$ where M is known as the modulation index. Full modulation occurs when $M = 1.0$. Therefore for $M \leq 1$, the monophonic AM signal can be written as

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$$e_c = E_0\left[1 + m(\sin\omega_m t)\right]\sin\omega_c t \quad (1)$$

where ω_c is the carrier angular velocity and ω_m represents the audio signal angular velocity. For a stereo audio source, a compatible mono signal is composed of the sum of Left and Right components and can be written $(L + R) = \sin\omega_m t$

To transmit the stereo signal for the benefit of receivers equipped to decode it, in the Magnavox system the radio frequency carrier is deviated in phase (PM) by the instantaneous amplitude of the stereo component $(L - R)$ of the signal. If $\phi(t)$ is the angular displacement of the carrier at time t then

$$\frac{d\phi(t)}{dt} = \omega_c \quad (2) \qquad \therefore \phi(t) = \omega_c t + \theta \quad (3)$$

(where θ = displacement at $t=0$)

For phase modulation the reference phase θ is varied with the instantaneous amplitude of the modulating signal

$$\theta = \theta_0 + m_p \sin\omega_s t$$

θ_0 = phase with no modulation
ω_s = signal angular velocity
m_p = phase modulation index
$(\sin\omega_s t = (L-R))$

Substituting this is Eq. (3) and assuming for convenience that $\theta_0 = 0$

$$\phi(t) = \omega_c t + m_p \sin\omega_s t \quad (3)$$

$$e_c = E_0 \sin(\omega_c t + m_p \sin\omega_s t) \quad (4)$$

This is the equation for a phase modulated carrier waveform. Therefore, combining Eq. (1) and Eq. (5) we obtain the AM/PM stereo signal

$$e_c = E_0\left[1 + m\sin\omega_m t\right]\sin\left[\omega_c t + m_p \sin\omega_s t\right] \quad (5)$$

To limit the spectrum utilization, the Magnavox system limits the peak deviation produced by the (L - R) modulating waveform to 1 radian (57.3°). This means that for 100% modulation in both the AM and PM channels, $M = M_p = 1$.

Unlike FM stereo, where the (L - R) or stereo information is modulated on a suppressed subcarrier requiring the simulataneous transmission of a harmonically related pilot carrier, the AM stereo signal does not require a pilot for decoding. Nevertheless, a stereo identification signal is also included, to facilitate automatic switching into the stereo mode and to provide visual identification of stereo broadcasts. This identification signal is a low frequency tone (5Hz) which modulates the carrier phase by 4 radians. Therefore the total signal equation, for the Magnavox system, becomes

$$e_c = E_o\left[1+\underbrace{m\sin\omega_m t}_{\substack{(L+R)\\ \text{MONO}}}\right]\cos\left[\omega_c t+\underbrace{m_p\sin\omega_s t}_{\substack{(L-R)\\ \text{STEREO}\\ \text{DIFFERENCE}}}+\underbrace{4\sin 10\pi t}_{\substack{\text{PILOT}\\ \text{TONE}}}\right] \qquad (6)$$

The Transmitter

The conversion of a conventional monophonic AM transmitter to AM/PM is relatively straightforward, as shown by the block diagram of Figure 2. Instead of a fixed frequency r.f. carrier, the carrier is phase modulated by a 5Hz stereo identification signal with a peak deviation of 4 radians.

It can be shown that the peak frequency deviation of a phase modulated carrier waveform is given by Equation (7)

$$\Delta f_c = m_p \times f_s \quad (7)$$

f_s = modulating frequency
m_p = modulation index
Δf_c = carrier deviation

The 5Hz tone produces a 20Hz peak frequency deviation of the carrier.

Left and Right audio components are matrixed to form (L + R) and (L - R) signals. The (L + R) signal is the normal monophonic signal and is used to amplitude modulate the carrier in the normal way. Because the carrier is simultaneously being phase modulated with the (L - R) signal, the carrier instantaneous amplitude cannot be allowed to diminish to zero by large negative modulation indices ($M- \geq 1$). Therefore the AM section of the transmitter is restricted to 95% negative modulation. Positive modulation peaks are no problem, allowing up to 125% carrier level increase as in current practice.

Regarding the permitted spectrum utilization, Figure 3, an (L + R) single tone cannot produce sideband amplitudes greater than -25dB compared to the unmodulated carrier amplitude from 15kHz to 30kHz. Above 30kHz the sidebands have to be below -35dB. Clearly the occupied spectrum standards allow the broadcast of a "HiFi" signal. In practice these sideband amplitudes are unlikely to be encountered. AM transmitters are required to have a frequency response that is ± 2dB from 200Hz to 5kHz, and to be "proofed" out to 7.5kHz. Where the studio and transmitter are physically separated, an 8kHz telephone line link is common. Even when broadcasting occurs at frequencies out to 15kHz (30kHz bandwidth at r.f.)

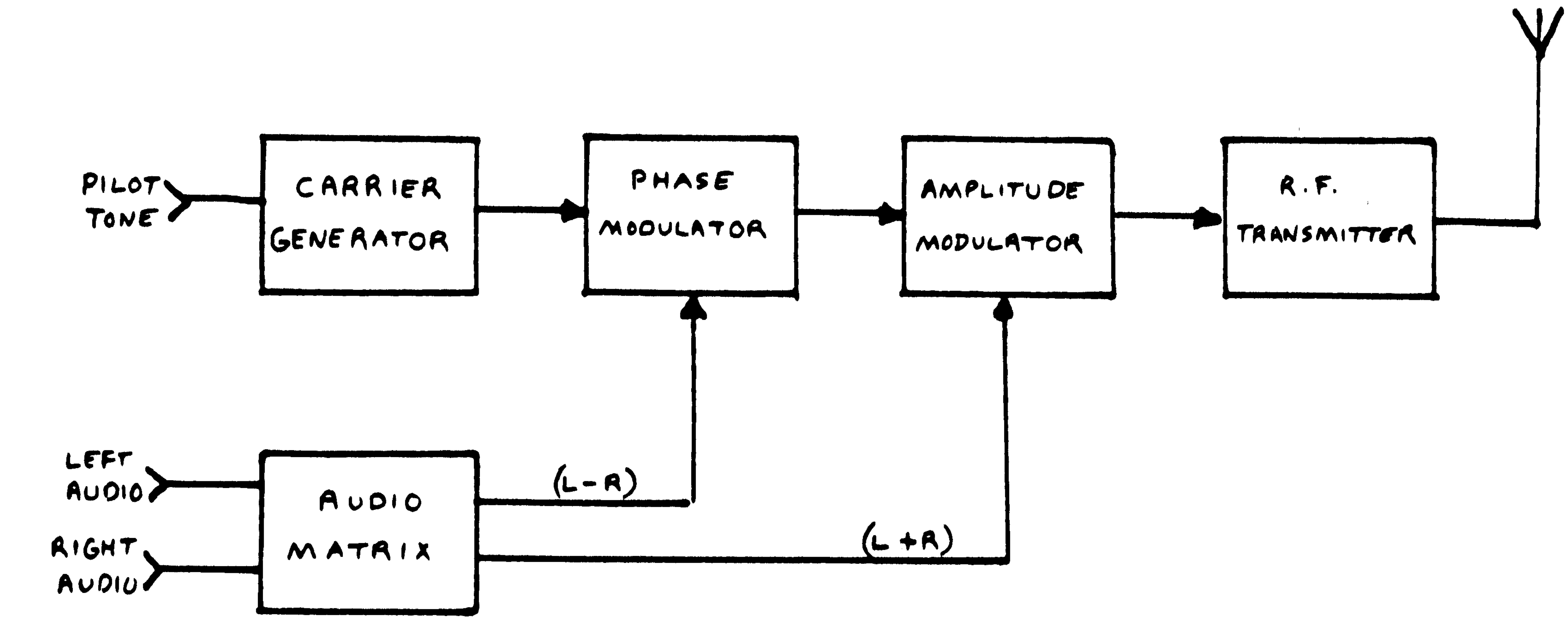


FIGURE 2. AM STEREO TRANSMITTER

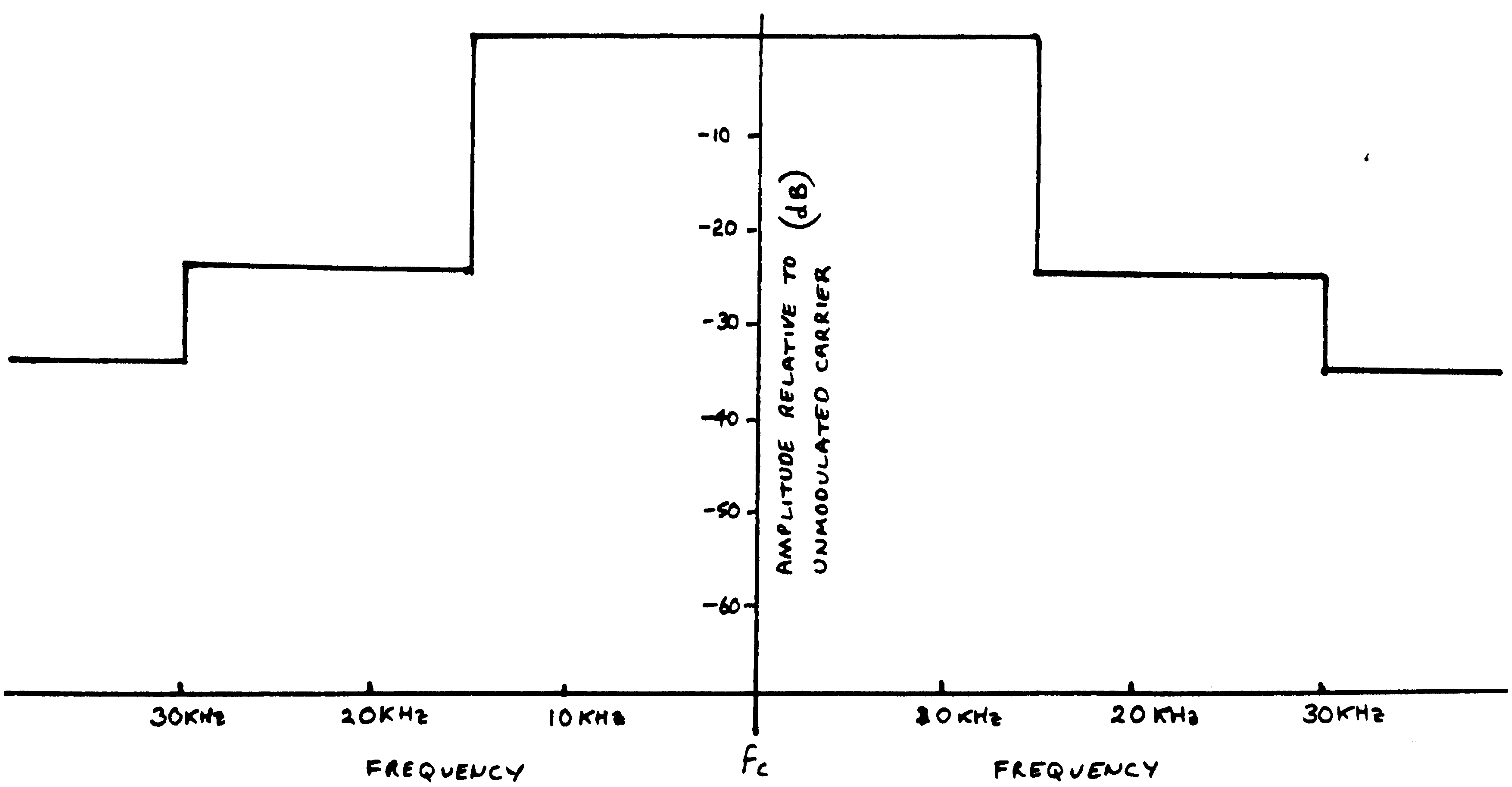


FIGURE 3. AM BROADCAST SPECTRUM LIMITS

most of the energy in the broadcast signal is at frequencies well below 10kHz - fortunately so for receivers tuned to adjacent channels (10kHz away). Introduction of AM stereo is not likely to encourage an increase in the broadcast frequency response, particularly when the spectrum of the (L - R) phase modulating signal is considered.

Neglecting for the moment the 5Hz stereo identification tone, the (L - R) signal can cause a carrier phase peak deviation of 1 radian. Because the PM modulation index is constant for all modulating frequencies, the sideband amplitudes produced by any audio frequency can be calculated by reference to the appropriate Bessel functions. Table 1 shows the sideband amplitudes for 30%, 68% and 100% PM. By comparing these amplitudes with the permitted spectrum of Figure 3 we can see that 100% modulation is practical up to about 7.5kHz. Above this frequency the maximum modulation with a single tone is limited to 68%.

Another factor to be considered with the increased spectrum utilization is the impact on protection ratios. Current standards for AM stations place limitations on the day/night co-channel radiation, and on groundwave signal (day) radiation to the 1st, 2nd and 3rd adjacent channels. These standards specify the maximum ratio of overlapping field strength contours with a threshold level of interference taken to be at least -26dB below the desired groundwave signal, Measurements taken by the NAMRSC on receivers shows that the stereo signal tends to raise the interference level by about 3dB on co-channel, 1dB to 4dB on the 1st adjacent channel, and by 14dB on the second adjacent channel, compared to the interference produced by a monophonic signal. The second

TABLE I

MODULATION INDEX	SIDEBAND ORDER	LEVEL REFERRED TO UNMODULATED CARRIER	LEVEL (dB)
0.30	0	0.977626	-0.2
	1	0.148319	-16.6
	2	0.011166	-39.0
	3	0.000559	-65.0
	4	0.000021	-93.6
	5	0.000001	-129.0
	6	0.000000	-156.0
0.68	0	0.887698	-1.0
	1	0.320723	-9.9
	2	0.055605	-25.1
	3	0.006369	-43.9
	4	0.000544	-65.3
	5	0.000037	-88.6
	6	0.000002	-113.5
1.00	0	0.765198	-2.33
	1	0.440051	-7.1
	2	0.114903	-18.8
	3	0.019563	-34.2
	4	0.002477	-52.1
	5	0.000250	-72.1
	6	0.000021	-93.6

PHASE MODULATION SIDEBAND AMPLITUDES

adjacent channel interference for stereo is actually at the -26dB threshold level. Further work by Magnavox shows that limiting the (L - R) signal with an 8kHz filter substantially reduces the 2nd adjacent channel interference. Limiting the (L - R) and (L + R) signals to 5kHz produces comparable interference levels to the monophonic condition.

It might be wondered how much stereo information is available if the audio signals are going to be restricted to something less than 8kHz?

Fortunately (for the recognition of stereo) there is data that shows separation is best perceived in the frequency range from 200Hz to 5kHz. As a result the transmitted separation is proposed to be $\geq$ 26dB over the frequency range 600Hz to 5kHz.

A further aspect of transmitter performance is that of incidental PM. If the carrier frequency varies with the level of amplitude modulation, then stereo separation will suffer and the distortion in the (L - R) channel will be increased. An example of this type of problem can be provided by a typical laboratory AM/FM signal generator used to simulate AM stereo signals. This particular generator is specified to have no greater than 60Hz carrier deviation when amplitude modulated by 30%. An accuracy of 0.006% is more than adequate for standard AM but if this deviation actually occurs when modulating the carrier with a 1kHz (L + R) signal, a simultaneous 1kHz PM signal is generated. Since, from Equation (7), a 30% modulated 1kHz (L - R) signal produces a carrier peak deviation of $1 \times 10^3 \times 0.3 = 300$Hz our unwanted carrier deviation of 60Hz is only 14dB down. Quite obviously, this generator is totally unsuitable for making measurements on an AM stereo receiver. The trasmitters are expected to produce less than 2% phase modulation with a 400Hz tone at 85% amplitude modulation. This implies a carrier accuracy of 8Hz at 1MHz.

One final aspect of the AM/PM stereo signal to be considered is that of the S/N ratio - will the stereo signal suffer as much degradation in S/N compared to the mono signal as does an FM stereo signal ($\simeq$ -23dB)? This is not the case. In FM stereo the additional noise comes from the (L - R) subchannel which occupies a different part of the spectrum to the (L + R) signal. For the AM/PM stereo signal, the (L - R) signal occupies the same frequency spectrum as does the (L + R) signal. It can be shown that the AM stereo quiescent noise level is from 1.8dB to to 3dB higher than the conventional monophonic quiescent noise level.

The Receiver

A block diagram of a receiver suitable for decoding the AM stereo signal is shown in Figure 4. This resembles a conventional AM receiver in several respects and it would be possible to convert a standard AM/FM/ FM Stereo receiver to one capable of AM stereo reception by the addition of a limiter, PM detector, stereo identification tone detector and audio matrix. However, as we shall see, it is more likely in practice that complete re-design or modification of the entire AM section will occur.

Briefly, the entire AM/PM modulated carrier is mixed down in a superhet front end, either to a 455kHz or to a 262kHz intermediate frequency. At the I.F. amplifier output the signal splits in two directions. A typical envelope detector may be used to extract the amplitude modulated intelligence and to detect the r.f. carrier level for

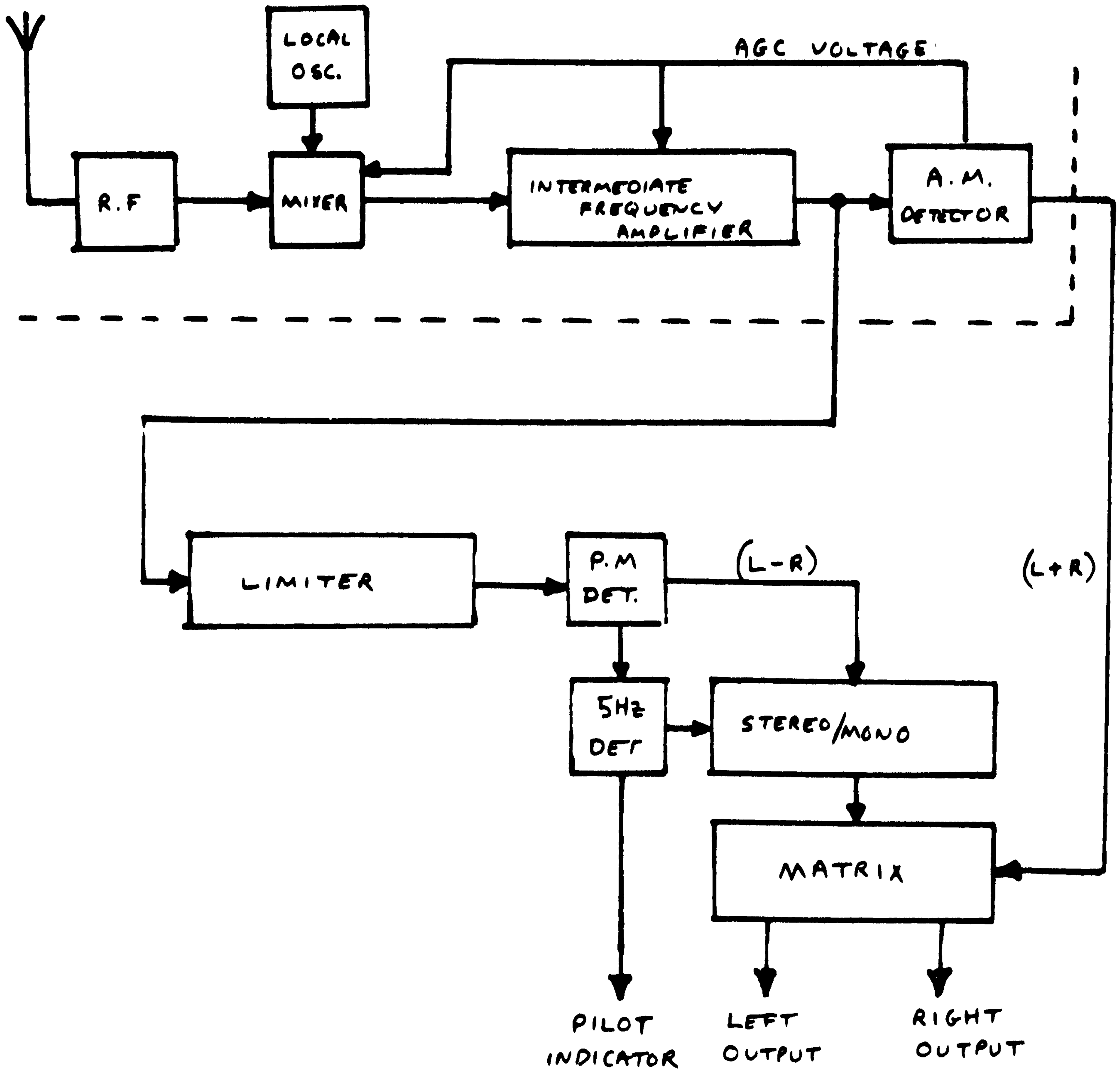


FIGURE 4. AM STEREO RECEIVER

an a.g.c. function. At the same time the I.F. amplifier output is limited to remove amplitude modulation, and a PM detector supplies the (L - R) audio signal and the 5Hz stereo identification tone. Both AM and PM channel outputs are applied to the matrix which will drive the Left channel and Right channel audio signals. If indeed an (L - R) signal is present, indicated by the 5Hz tone detection, the stereo/mono switch is enabled in the stereo mode. Absence of the tone, or unsuitable reception conditions, will result in the switch being in the mono mode, and (L + R) is applied to both Left and Right channel outputs. In the stereo mode

$$\text{LEFT} = \frac{(L+R)+(L-R)}{2} \qquad \text{RIGHT} = \frac{(L+R)-(L-R)}{2}$$

Practical circuits rarely operate as smoothly as block diagrams would suggest, and decoding the AM/PM signal is no exception. To begin with, we can examine more closely the requirements of the AM receiver front end, up to the 455kHz I.F. output and including

a) the r.f. antenna,

b) Local oscillator stability,

c) I.F. Amplifier Bandwidth and Symmetry,

and d) Agc characteristics.

a) The Antenna Circuit.

Two types of AM antenna are most common. In automotive radios a whip antenna is used, whereas for portable and table radios a built-in ferrite rod antenna is most popular. Both types of antenna are coupled to the input stage of the radio through a tuned circuit and there may or may not be any active gain stage at radio frequencies before the mixer stage (see Section 3 of National's AUDIO/RADIO HANDBOOK). For most present day designs the front end bandwidth is not wide - an input tuned circuit loaded Q of 100 being typical. Because of the short effective height of the ferrite rod antenna, this high Q helps increase the radio sensitivity but also means that the input bandwidth is about ±5kHz at 1MHz. At lower radio frequencies near 600kHz the bandwidth is even less a little over ±2kHz for -3dB response. If we have any intention of accurately reproducing the broadcast stereo signal it will be necessary to redesign the receiver front end for less selectivity.

b) Local Oscillator (L.O.) Stability.

A similar situation exists for the (L.O.) in a superhet front end receiver as does for the transmitter carrier frequency generator. Incidental phase modulation will cause loss of separation and/or raise the "noise floor" in the PM channel. For example, in synthesized AM frequency tuners, the noise floor caused by phase variations in the V.C.O. frequency (either intrinsically in the V.C.O. or in the phase detector control voltage of the P.L.L.) seems to be typically at about -60dB. Conventional discrete or I/C mixer/oscillator combinations can easily produce higher incidental PM since this has not been a design consideration for mono AM receivers. Isolation between the mixer and the L.O. is important, to prevent "pulling" of the L.O. by the mixer signal. Even the tuning capacitor can cause audible problems (i.e. microphonics). Typical solid dielectric tuning capacitors are unsuitable from this viewpoint - any slight vibration of the p.c.b. will produce an output in the PM detector. For table radios that use air dielectric tuning capacitors ganged for AM/FM operation the problem will be less severe, but compact portable designs are likely to pose difficulties.

c) I.F. Amplifier Bandwidth and Bandpass Symmetry.

In their evaluation of the proposed systems for AM stereo, the National AM Stereophonic Radio Committee (NAMSRC) stated

"The choice of I.F. bandwidth is probably one of the largest variables affecting the NAMSRC tests of AM stereo receivers".

This can be true for either stereo or monophonic broadcasts but the addition of the phase modulated (L - R) channel does create some new performance differences.

As mentioned earlier in the description of the transmitter signal, current AM transmitters are able legally (although unlikely) to broadcast over a full 15kHz audio bandwidth, occupying a 30kHz r.f. bandwidth. If a receiver were designed to take full advantage of this and had a 30kHz received bandwidth, only transmitters much stronger than the adjacent channel transmitters (which are separated by only 10kHz) would provide a satisfactory signal. This is because the AGC developed by the strong transmitter r.f. signal strength reduces the receiver sensitivity to the adjacent and potentially interfering transmitters. However when the receiver is tuned to a weaker transmitter, or for an automobile receiver moving away from the transmitter, this situation will not necessarily exist and objectionable adjacent channel interference could occur. The use of protection ratios defining the permitted level of fieldstrength overlap between adjacent channels helps minimize interference problems and the fact that most of the radiated energy is within 10kHz of the carrier frequency enables receivers with severely limited bandwidths to be used. Even so, protection ratios apply only to groundwave signals or normal daytime broadcasting. At night, radio propagation at medium frequencies by skywave becomes a significant factor, multiplying the number of potentially interfering transmitters. Because of this, satisfactory daytime reception with a relatively wide receiver bandwidth may be totally unsatisfactory at night. Current AM receiver designs are dominated by restricted bandwidth I.F. amplifiers and antenna circuits (which also results in fewer components and more economic designs) but consideration of the ability to recieve a stereo broadcast should change this. Because of the day/night difference, it is likely that there will be a return to dual bandwidth receivers with a "Wide/Sharp" (or Fidelity-Normal!) front panel switch.

The actual bandwidths that are used will depend on the quality (and price) of the radio as well as what will be considered an acceptable level of performance. Referring back to the transmitter signal description, the need to provide comparable adjacent channel performance and keeping the spectrum within permitted limits means that the stereo signal will already be limited in terms of the audio bandwidth. A "Fidelity" position giving a bandwidth of ±6kHz to ±8kHz, decreasing to something less than ±3kHz in the "Normal" position is likely.

Before leaving the subject of the I.F. amplifier bandwidth, it is worth noting that limiting the bandwidth to something less than the transmitted signal bandwidth has a different effect on the (L−R) channel than on the (L + R) channel. When a single 5kHz tone is amplitude modulated on the carrier, this will produce two equal amplitude sidebands at 5kHz on either side of the carrier frequency. A ±6Hz bandwidth receiver will detect this with no problem and produce an undistorted 5kHz output. If this same tone is now considered to be an L = - R stereo signal which is phase modulating the carrier, the transmitted spectrum will include sidebands at 5kHz, 10kHz, 15kHz and 20kHz (See Table 1) if high modulation indices are being used. Now the ±6kHz bandwidth in the receiver will cause attenuation and complete loss of the higher frequency sidebands which will result in distortion of the detected 5kHz tone. This will occur only for large modulation indices - for low carrier phase deviations the transmitter spectrum will more closely resemble the AM case with a single set of significant sidebands. Since in an actual broadcast the total carrier modulation is a function of the prescence of a number of tones, any individual frequency is unlikely to have large amplitude higher sidebands. Even so, the receiver designer will have to be aware of the sources or potential for high distortion numbers and not necessarily attribute them to faults in transmission or decoding.

The actual I.F. amplifier bandwidth is not the only factor to be considered. Assymetry of the passband about the center frequency can also cause distortion and loss of separation. For example, separation loss because of mismatched (L + R) sidebands is shown in the curve of Figure 5. This mismatch can be caused by passband assymetry through a.g.c. action or mistuning.

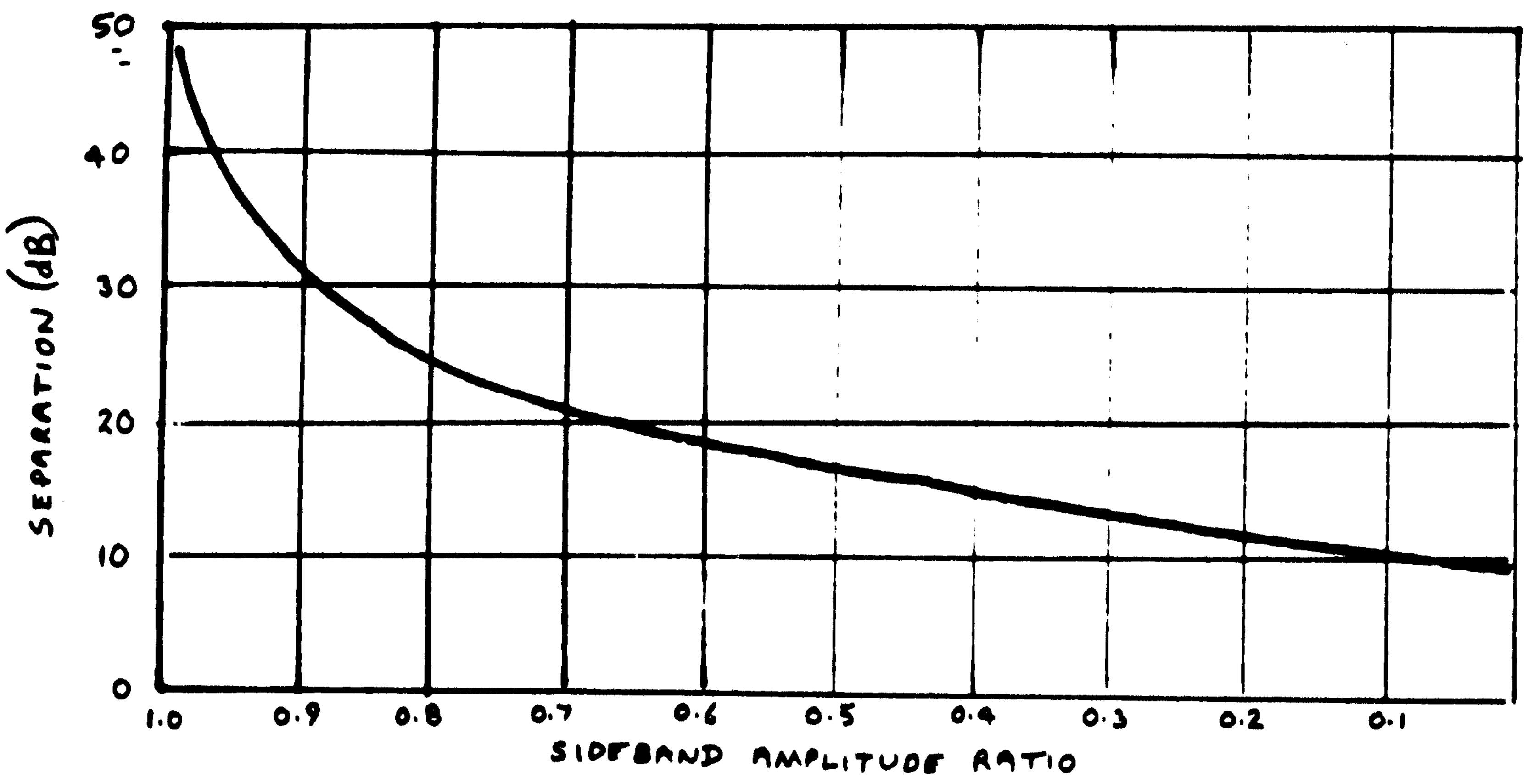


FIGURE 5. CALCULATED SEPARATION LOSS VS SIDEBAND AMPLITUDE IMBALANCE (L+R ONLY)

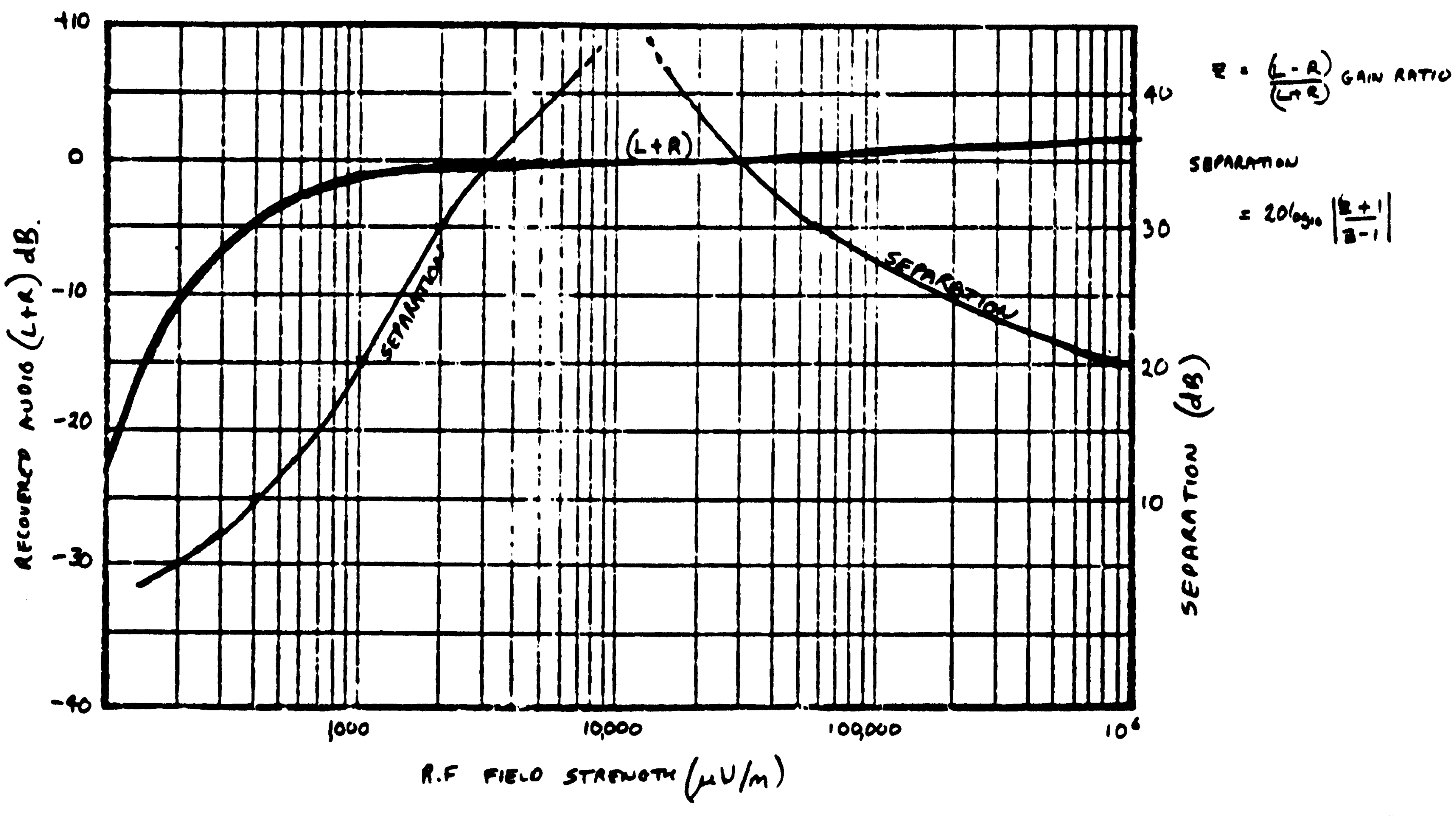


FIGURE 6. CALCULATED SEPARATION LOSS WITH (L+R) AMPLITUDE CHANGE (AGC CHARACTERISTIC) VS RF FIELD STRENGTH

d) A.G.C. Characteristics.

When the receiver is properly tuned the phase modulated (L - R) signal is detected after passing through a limiter stage to remove amplitude variations. The (L + R) signal is detected and the average detector level is used as an indication of the r.f. signal strength for A.G.C. purposes. Therefore the actual detected (L + R) level can be expected to vary depending on the mixer/i.f. amplifier a.g.c. characteristics. Comparable variations in detected level will not be occuring in the PM channel so that the signals (L - R) and (L + R) that are applied to the matrix will not have the correct amplitude ratios. The loss in separation is shown graphically in Figure 6. For example, if we wish to maintain the separation better than 20dB from an r.f. signal strength of 1000μV/m to 1V/m the (L + R) level must change less than 4dB. This implies an A.G.C. figure of merit better than 60dB. Using high a.g.c. F.O.M. is not always possible or desireable. An alternative is to control the output amplitude of the PM channel with a control voltage derived from the (L + R) detected output.

The Decoder

Assuming that any deficiencies or distortions in the RF/IF stages are either eliminated or accounted for, we can move on the the new section of the receiver, the AM stereo decoder.

The I.F. amplifier output is first passed through a limiting amplifier to remove the amplitude modulation. Obviously this limiter needs sufficient gain to provide limiting in the presence of a 95% negative modulated I.F. carrier frequency and some method for handling the times when over modulation of the carrier will occur. If the limiter is capable of limiting on this and lower input signal levels, the limiter will also switch on noise inputs. If the r.f. carrier S/N ratio is less than 26dB then the phase channel input will be simply noise on large negative modulation swings. Normally, of course, the carrier/noise ratio will be better than 26dB for acceptable listening. For example a 20dB S/N with 30% AM is actually a carrier/noise ratio of 30dB. Nevertheless temporary carrier fading can mean that the PM channel is producing bursts of noise (maximum detected amplitude because of the random noise phase). Therefore the decision to remain in the stereo mode can depend in part on the average modulation depth and field strength of the received signal.

There are two basic ways to detect the signal on an angle modulated carrier. The first, possibly most direct way, is to use a phase locked loop as shown in Figure 7. For an FM signal the error voltage detected at the phase detector filter output is that necessary to keep the V.C.O. in precise phase lock with the incoming signal - i.e. the modulating signal. Hence the L.P.F. determines the detected audio signal bandwidth or the range of frequencies over which the V.C.O. can stay in exact phase with the incoming carrier. For a PM signal, if the V.C.O. is locked to the carrier frequency but stable (non-varying) in phase, then the output of the phase detector is proportional to the phase difference between the V.C.O. and the carrier - or the modulating signal. The L.P.F. must not pass any desired modulating frequency through to the V.C.O. While this technique is direct, it does present several problems of a practical nature.

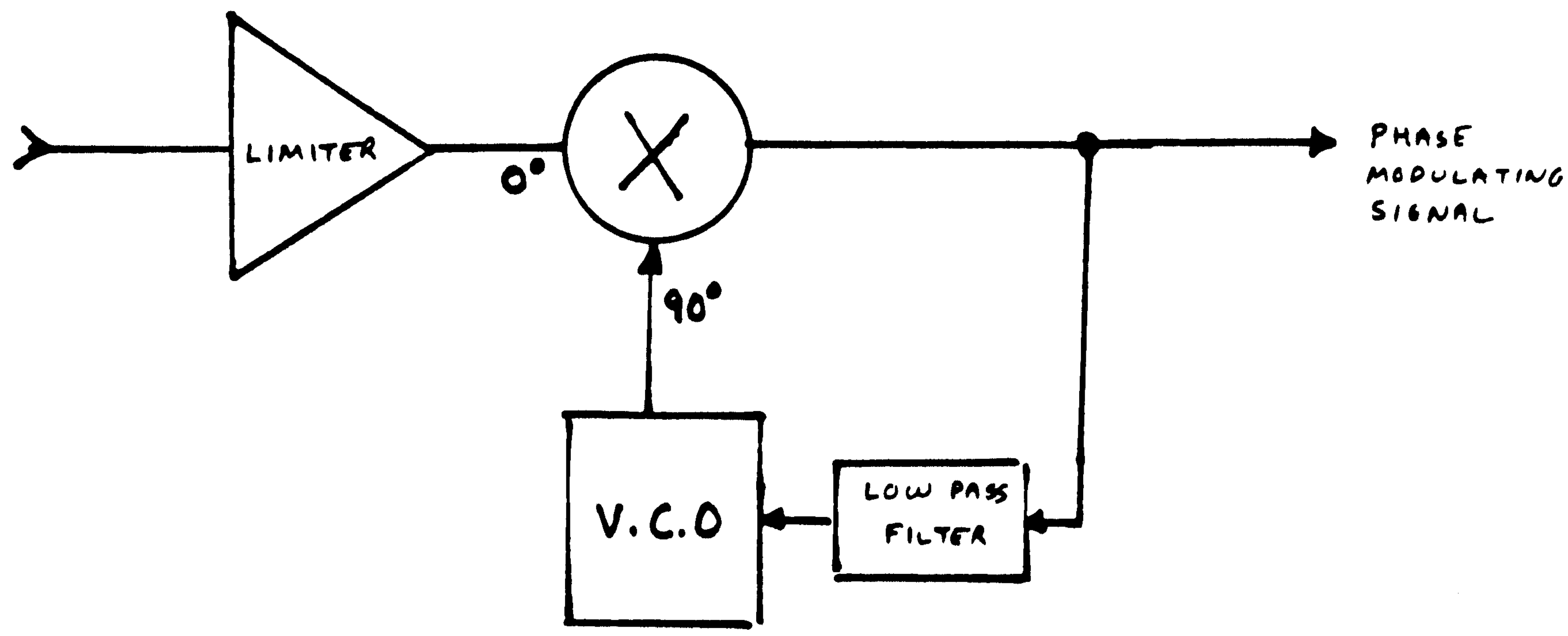


FIGURE 7 P.L.L DECODER

1) V.C.O. Radiation.

Since the V.C.O. must be operating at the intermediate frequency (455kHz or 262kHz), the radio front end must have good I.F. rejection and the V.C.O. itself have low radiation and coupling to the mixer or I.F. input.

2) Muting.

As the V.C.O. comes into lock during tuning to the transmitter frequency, the instantaneous frequency difference between the V.C.O. and the I.F. amplifier output can produce beat notes or a whistling tone that decreases in frequency as the V.C.O. approaches lock. This means that muting circuits are needed to prevent this tuning effect from being heard.

3) Stereo Identification Tone Detection.

The Magnavox AM stereo signal presents a particular problem in that the stereo identification tone is low frequency and has a much higher deviation than does the audio, 229° compared to 57° maximum. If the V.C.O. is stable in phase, then output signal inversion will occur at each 90° change in phase of the stereo identification tone. To avoid this the L.P.F. must pass some proportion of the 5Hz tone through to the V.C.O. in order to keep the multiplier detected phase difference less than 90°. As the detected 5Hz tone level is reduced by this technique, the low frequency pole in the main signal path goes higher, which will tend to reduce the low frequency stereo separation. However the acquisition time of the loop will be enhanced since the V.C.O. control voltage frequency response is improved. This latter point is probably more significant than loss of separation at low frequencies, particularly since long acquisition times are not very acceptable in consumer products. Thus a trade-off will occur between locking time, low frequency stereo separation and the 5Hz detected level unless more complex circuitry is resorted to.

The other, and more general method for detecting angle modulation is to apply the amplitude limited carrier to a differentiating network which will produce an envelope modulation proportional to the carrier instantaneous frequency. This envelope modulation can be amplitude detected (for FM) and then integrated to recover the original modulating information (for PM).

There are several possible ways to differentiate and detect the angle modulated carrier and the most popular (at the consumer level) have been either frequency domain or time domain types. Frequency domain differentiation is obtained by a linear network with a sloping magnitude response over the relevant band of frequencies. Resonant circuits tuned above or below the carrier frequency can be used, particularly when the frequency deviation is small compared to the carrier frequency, but these simple circuits are sensitive to incidental amplitude variations.(Actually the standard AM/IF amplifier tuned circuits are capable of slope detecting the angle modulation of the carrier if the carrier is tuned over to the passband skirts). In the time domain, the differentiation function is obtained by the time delay of a tuned circuit resonant <u>at</u> the carrier frequency. If the limited signal is applied to one input of a multiplier, then the tuned circuit will present a second delayed signal with amplitude

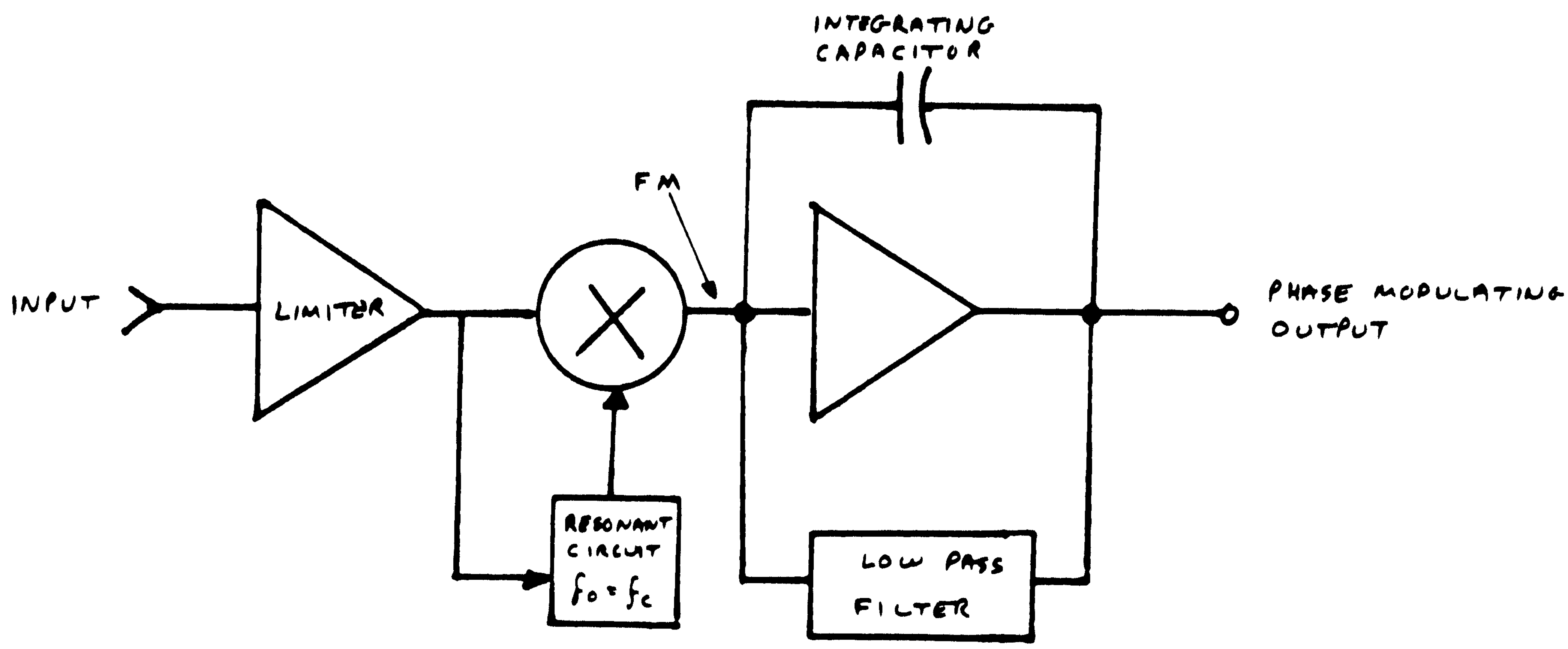


FIGURE 8 F.M DETECTOR AND INTEGRATOR FOR PM

and phase variations proportional to the carrier instantaneous frequency to the other input. This is the basis of the most popular I/C FM detectors such as the LM2111 and LM3089. Since this is a frequency modulation detection scheme, for recovering the PM stereo signal we simply integrate the detected output. Expressed mathematically we have

$$e(t) = E_0 \sin\left(\omega_c t + m_p \sin \omega_m t\right) \tag{4}$$

Differentiating,

$$\frac{d}{dt} e(t) = K_1 \underbrace{\left(\omega_c + \omega_m m_p \cos \omega_m t\right)}_{\text{Envelope proportional to carrier instantaneous frequency}} \cos\left(\omega_c t + m_p \sin \omega_m t\right) \tag{8}$$

The detector recovers the envelope in Eq. (8) and the carrier frequency and higher harmonics are filtered or balanced out. Therefore, integrating the detected output yields the modulating signal.

$$e_m = \int K_1 K_2 \, \omega_m m_p \cos \omega_m t$$

$$= -K_1 K_2 \, m_p \sin \omega_m t \tag{9}$$

where $K_1 K_2$ are differentiation and detector conversion constants

Using conventional FM techniques to detect the PM signal offers several advantages. The detector technology is widely available, low cost and simple to adjust. Absence of a V.C.O. means that radiation problems are eliminated and there are no audible tuning transients. While the quad coil needs to be precisely tuned to the carrier frequency (IF), millions of FM receivers show that this is not difficult or unreliable.

In the case of the AM/PM stereo signal, the integrated output includes the 5Hz pilot identification tone which has four times the amplitude of the maximum audio signal - in essence the audio will be riding up and down at a 5Hz rate. This can simply be decoupled from the matrix or, as shown in Figure 8, a low pass filter back around the integrator reduces the amplitude to limit the dynamic swing requirement. After detection the PM signal path goes in two directions. First a 5Hz tone detector identifies the stereo signal and provides an input to the stereo/mono mode switch and a front panel stereo indicator light. Secondly the (L - R) signal is applied to the matrix along with the amplitude detected (L + R) signal. At this point several practical considerations need to be taken into account.

While the identification tone will indicate the presence of a stereo broadcast, its detection may not be a sufficient reason to switch to the stereo mode or to maintain this mode. For example a badly noise contaminated signal may be preferable in mono (when the PM channel is experiencing modulation depths producing outputs that are totally random noise). Since noise produces phase deviations greater than 57°, detection of excess phase may also be used to help determine the stereo/mono switching function. On the other hand, since detection of the low frequency pilot can take a significant time period, if temporary loss of pilot detection occurs it may not be desirable to promptly switch out of the stereo mode. Provision of a blend function will help, in that stereo to mono switching can be done gradually (and less perceptibly) as the r.f. input signal deteriorates.

As mentioned earlier, I.F. amplitude variations caused by the signal strength/a.g.c. performance of the receiver will mean that the matrix inputs will be incorrect since the (L - R) signal is unaffected by carrier signal amplitude changes. To compensate for this, rather than improve the a.g.c. characteristics, the decoder can include a variable gain amplifier controlled by the detected average level of the (L + R) signal. Now the level of the (L - R) signal can track the amplitude changes of the (L + R) ensuring proper decoding with no shift in stereo image. In passing, it is worth noting that having the ability to modulate the (L - R) detected level also means that correction signals can be applied - in order to properly decode a quadrature modulated signal as used in the Motorola system for example. A block diagram of the LM1981 incorporating these features is shown in Figure 9.

Conclusion

AM stereo broadcasting, with the stereo information conveyed on an angle modulated carrier looks practical from a Transmitter and Receiver design viewpoint. However, to obtain full benefit of the stereo information being transmitted, redesign or modification of the AM receiver front end is necessary - simply tracking on a decoder to a conventional AM/IF output is not likely to be satisfactory.

Some emphasis has been placed on describing the ability of the decoder to detect angle modulation. While the Magnavox system uses PM, other methods of encoding the stereo information, such as FM or quadrature, will require the same general building blocks. It seems reasonable, therefore, that the LM1981 I/C can be configured to decode most of the proposed AM stereo systems.

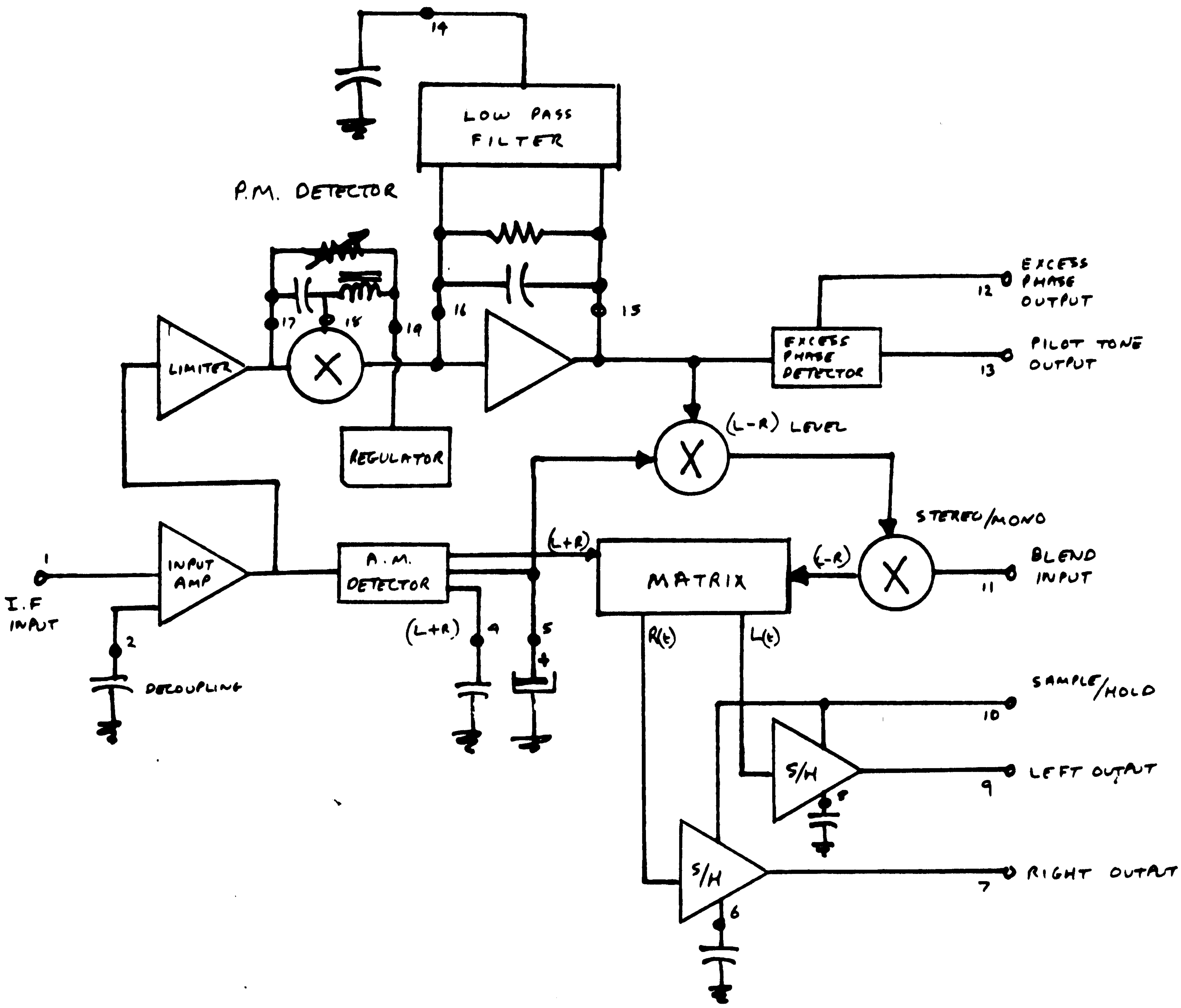


FIGURE 9. AM STEREO DECODER

LM1981 AM STEREO DECODER

Introduction

The LM1981 is an I/C designed to decode the stereo information that is amplitude and angle modulated on an AM Stereo broadcast carrier. It is capable of accepting the 455kHz (or 262kHz) I.F. amplifier output and amplitude detecting the (L + R) mono signal; limiting, detecting and conditioning the (L - R) stereo difference signal; and combining these signals in a suitable matrix to form the Left and Right channel audio outputs. Other features include an excess phase detector, stereo pilot tone output, stereo/mono blend function, output sample and hold circuits and an internal regulated reference voltage. This note describes the various functions of the LM1981, typical operating parameters, and details external component selection for a working AM stereo decoder. Preliminary ac/dc characteristics of the LM1981 are given in Table 3.

LM1981 Circuit Description

The basic features of the LM1981 are shown in the block diagram of Figure 9, which includes typical external component values used when the signal format is the Magnavox AM/PM system.

Usually the signal input to the LM1981 will be extracted from the final AM/IF Amplifier tuned circuit. In this particular design example, the tank circuit impedance is required to be 15kΩ and the 455kHz carrier level is approximately 400mVrms when the r.f. signal strength is above the a.g.c. threshold. From Table 3 we see that the

TABLE III

Vcc = 8V, Vin = 200mVrms, MODULATION DEPTH 45%

PARAMETER	NOTES		UNITS
SUPPLY CURRENT	Vcc = 8V	19	mA
REFERENCE VOLTAGE	PIN 19	4.26	Volts
REFERENCE CURRENT CAPABILITY	PIN 19	±1	mA
MONAURAL GAIN	PIN 11 SHORTED TO Vcc I/P 200mV @ 45% MOD	200	mV rms
MONAURAL DISTORTION	45% MOD	1	%
STEREO SEPARATION LEFT	LEFT ONLY 45% MOD	30	dB
RIGHT		30	dB
STEREO DISTORTION	45% MOD	1	%
S/N RATIO		50	dB
OUTPUT CURRENT CAPABILITY	PINS 7 AND 9	±1	mA
INPUT IMPEDANCE	PIN 2	15	kΩ
PILOT AMPLITUDE	PIN 13	28	μA(p-p)
INPUT DYNAMIC RANGE	PIN 2	+6, -20	dB
EXCESS PHASE OUTPUT	PIN 12	100	μA(p-p)
OPERATING SUPPLY VOLTAGE			
MAXIMUM		18	Volts
MINIMUM		7.5	Volts

LM1981 input resistance is 15kΩ and the nominal input level is 200mVrms. The input stage is capable of handling signal levels +6dB greater or -20dB less than 200mVrms which will ensure proper stereo operation until the signal S/N ratio is too low to be satisfactory, as shown by Figure 10.

It is necessary for the I.F. amplifier final stage bandwidth to be relatively wide so that the I.F. bandwidth, which may be selectable, is determined by the previous stages. A suitable circuit with a take-off point for the LM1981 is shown in Figure 11, along with the equivalent circuit. If the I.F. amplifier output resistance (R_O) is 100kΩ, the tank will be damped by this and the parallel input resistance (R'_{IN}) presented by the input resistance of the LM1981 reflected across the tank. Since the tap ratio will be 2:1 to provide the proper signal level to the LM1981, the reflected resistance R'_{IN} is given by

$$R'_{IN} = n^2 R_{IN} \quad (10) \qquad \therefore R'_{IN} = 60K\Omega$$

$$\text{and} \qquad R_t = R'_{IN} \| R_O = \underline{37.5K\Omega}$$

If we set the tuned circuit bandwidth at 20kHz, the unloaded circuit Q is

$$Q_L = \frac{f_o}{\Delta f} = \frac{455 \times 10^3}{20 \times 10^3} = \underline{23}$$

Now the dynamic resistance (R_D) of the tuned circuit in parallel with R_t must present a 15kΩ load to the I.F. amplifier output

$$\text{ie} \quad R_D \| R_t = 15K\Omega, \quad \therefore \underline{R_D = 25K\Omega}$$

Since R_D also determines the unloaded Q of the tuned circuit

$$\frac{Q_L}{Q_u} = \frac{R_t}{R_t + R_D} \quad (11) \qquad \therefore \underline{Q_U = 38.3}$$

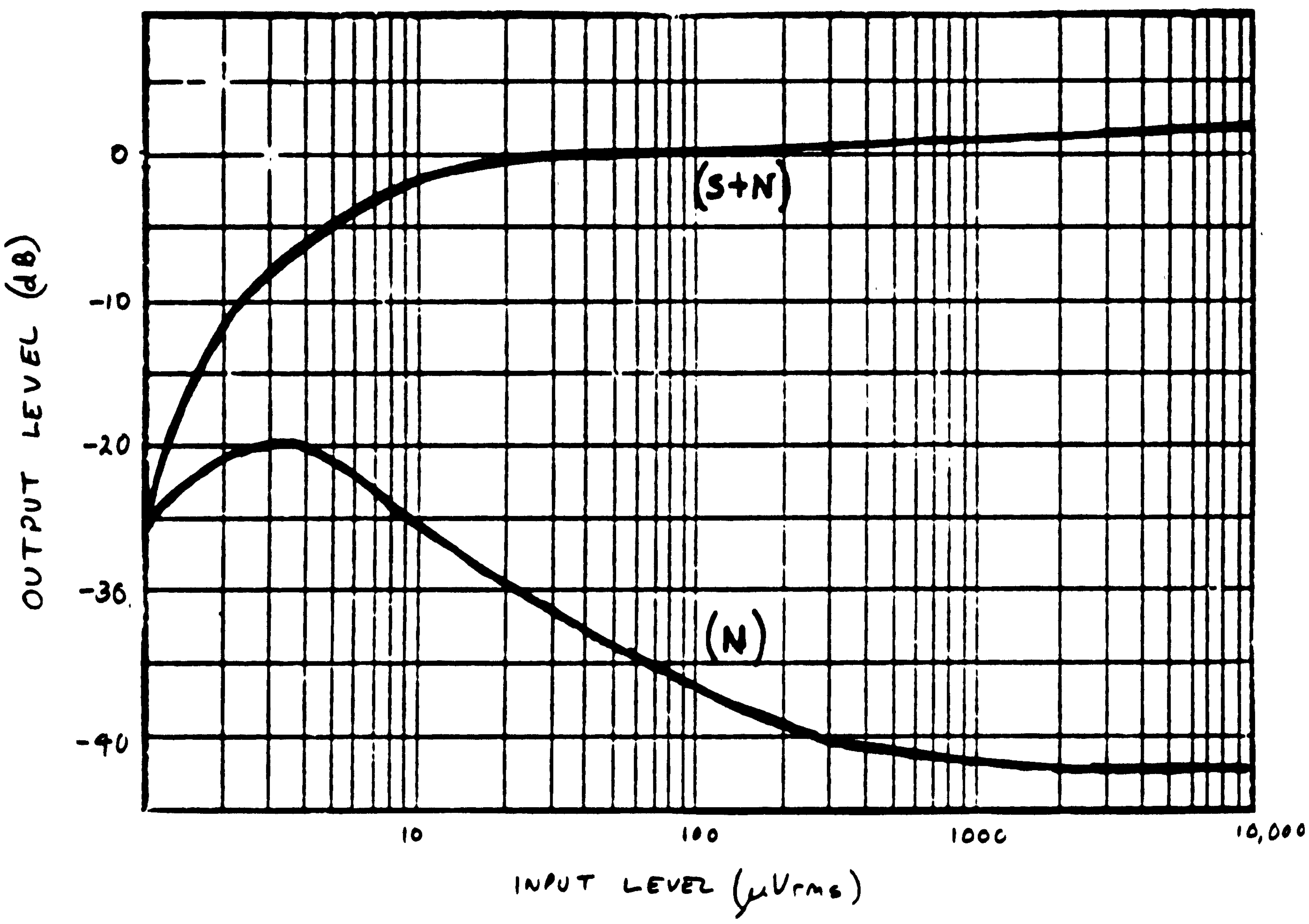


FIGURE 10 AM RADIO SIGNAL/NOISE PERFORMANCE

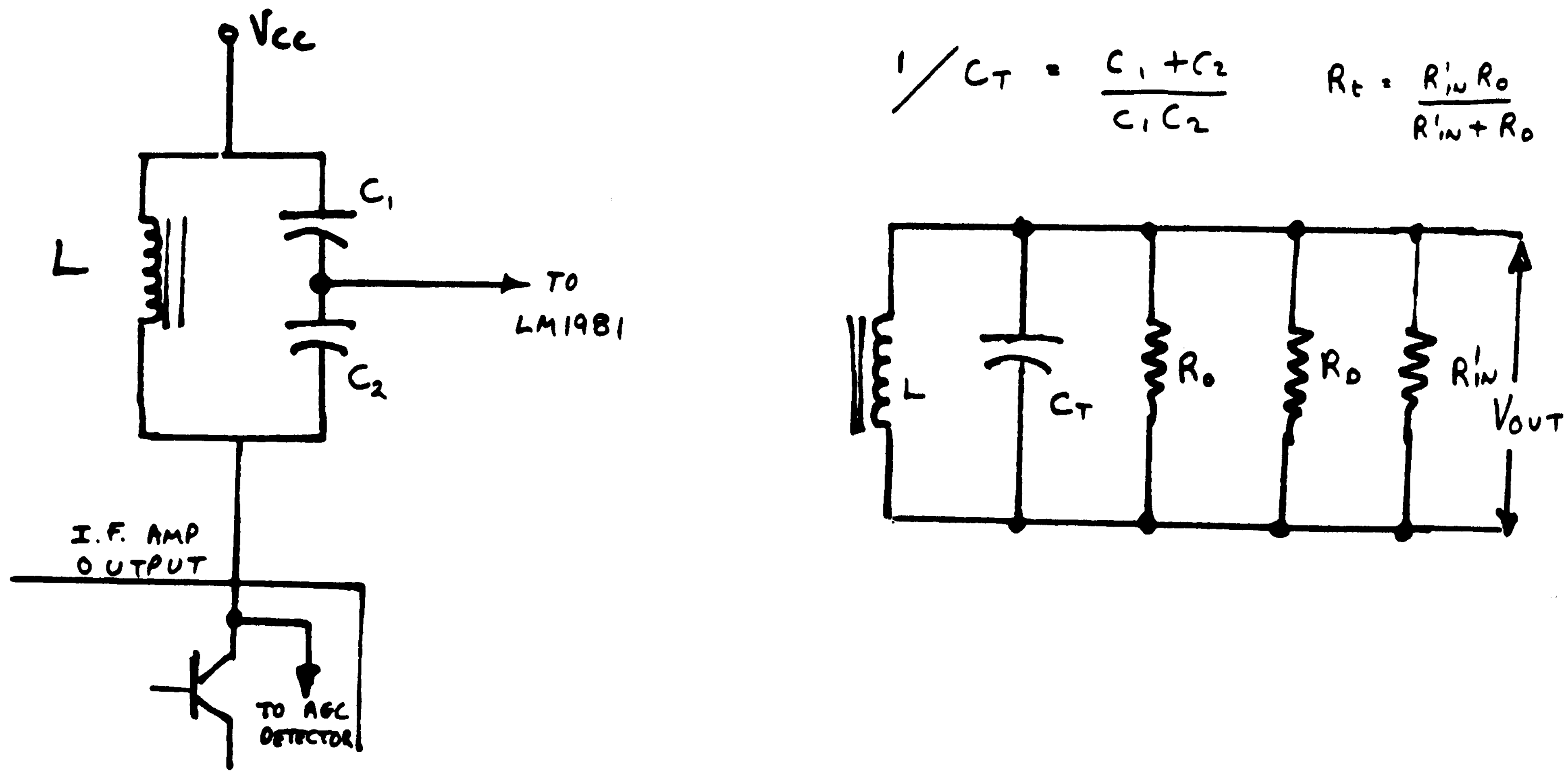


FIGURE 11 LM1981 INPUT CIRCUIT

For a parallel resonant circuit we have $L = \frac{R_D}{\omega Q_u}$ (12)

$$L = \frac{37.5 \times 10^3}{2\pi\, 455 \times 10^3 \times 38.3} = 228\mu H$$

The total capacitance necessary to tune this inductance to 455kHz is 536pF, so each capacitor is put at 1000pF.

Following the input stage, the signal is split up into two paths one to a limiter for the angle modulated information and the other to an envelope detector for the amplitude modulated information. This latter detector is a full wave rectifier shown in more detail in Figure 12. The balanced input will produce differential signal currents at the carrier frequency and the difference between these currents must be supplied by the emitters of Q_5. Therefore Q_5 collector current is proportional to the absolute magnitude of the carrier frequency. Filtering the carrier component at Pin 4 leaves the amplitude modulating information. Internally the resistance at Pin 4 is 2kΩ so that a 3900pF capacitor sets the -3dB bandwidth at 20kHz. The output level for 30% modulation (200mVrms I.F.) is 200mVrms.

Although the (L + R) monophonic signal is available at Pin 4, normally the signal for the audio amplifiers will be taken from the matrix which follows the detector. Also the detected output is heavily filtered at Pin 5 to provide a dc voltage proportional to the average value of the i.f. carrier. Because the absolute value detector will not peak detect on noise, this voltage is an accurate indication of the i.f. carrier level and is used to compensate the detected level of the (L - R) signal, which is insensitive to changes in the average carrier level caused by the a.g.c. not perfectly tracking the r.f. signal strength.

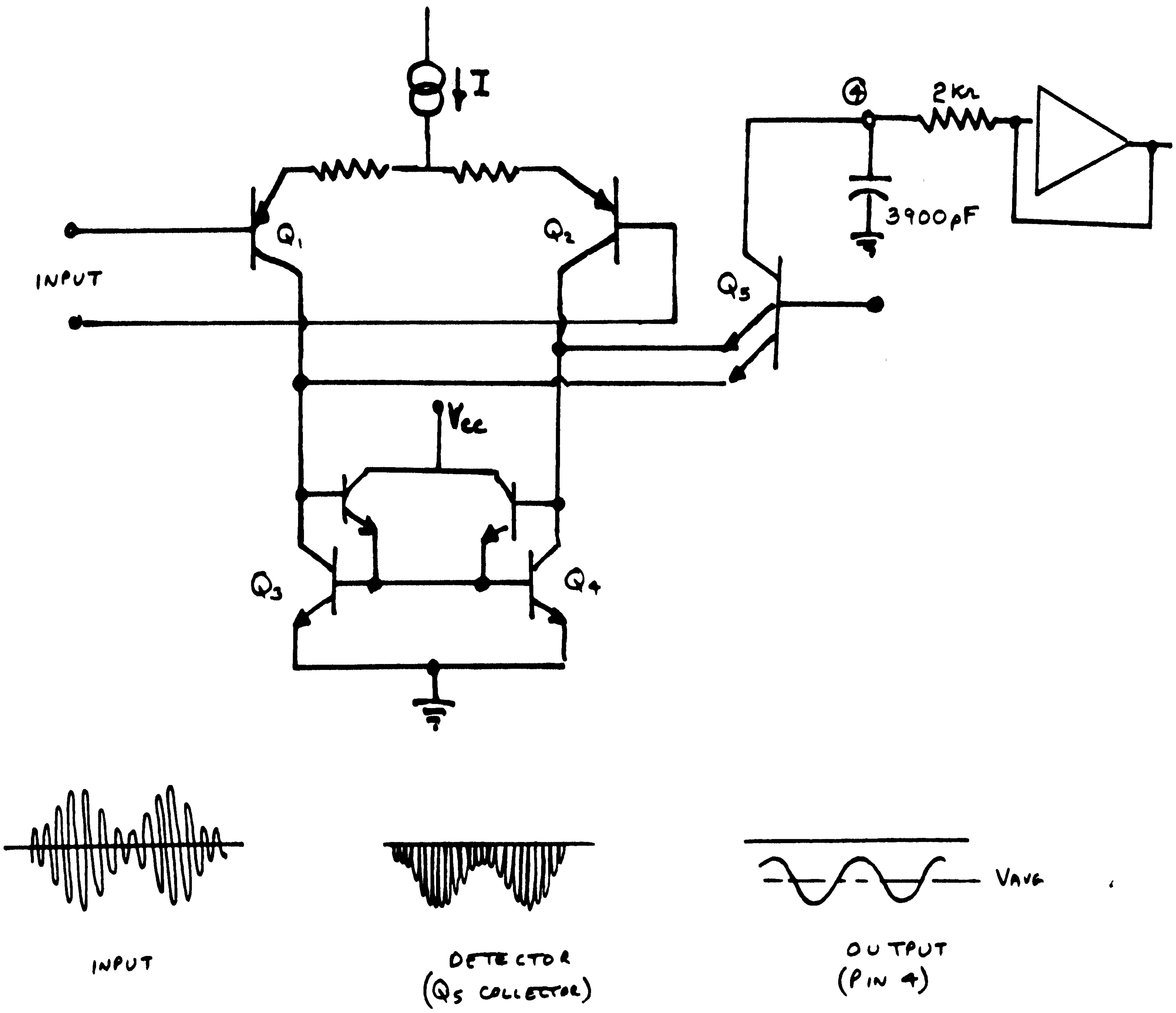


FIGURE 12(a) LM1081 AM DETECTOR

The internal load resistance at Pin 5 is 9kΩ so that a 10μF capacitor gives a 90mSec time constant.

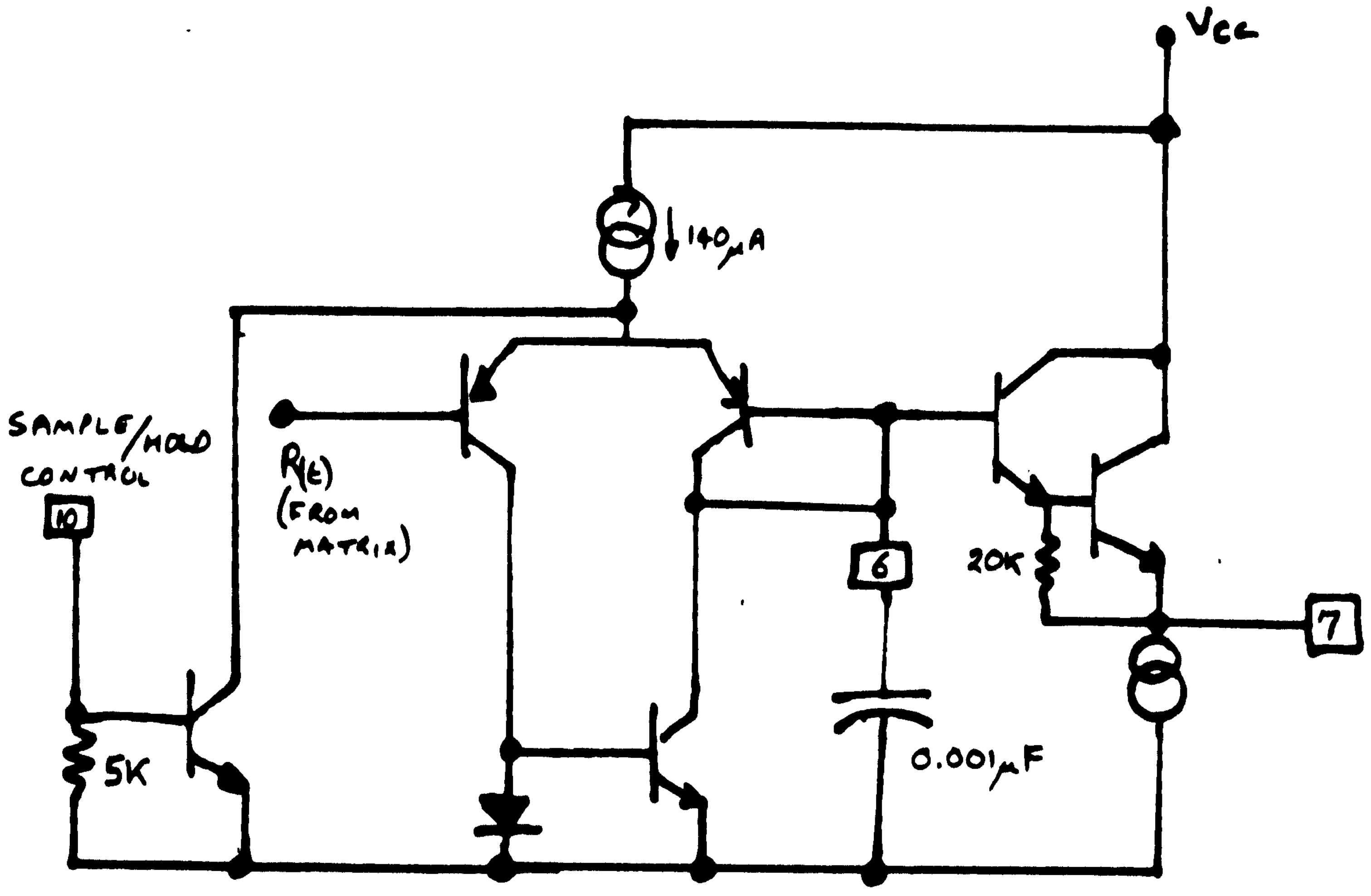
At the matrix the (L - R) signal is added to the (L + R) signal. The actual amplitude of the (L - R) signal will not only depend on the control voltage developed at Pin 5, but also on the voltage at the blend input, Pin 11. If Pin 11 is held above the internal reference voltage (4.26V on Pin 19) the (L - R) signal is completely muted and the Left and Right channel outputs will be (L + R) information only.

Both Left and Right channel outputs (Pins 9 and 7 respectively) are buffered with Sample/Hold circuits, Figure 12(b), which can be used to hold the signal level in the presence of a detected noise burst. If Pin 10 is left floating, the S/H circuits will pass the signal. Pulling Pin 10 to a V_{BE} above ground holds the Pin 7 and 9 output levels. The capacitors for the S/H circuits are at Pins 6 and 8, and these capacitors will slew limit the audio signal. The charge/discharge current is 140μA so that for a 1V swing at 20kHz

$$C_{PIN\ 6,8} \leq 0.0022\mu F$$

In the PM channel, the signal passes through 5 stages of limiting before being applied to the detector circuit. The limiter is stabilized with dc feedback decoupled at Pin 2. Although the carrier switching frequency is relatively low, the capacitor at Pin 2 is not large to enable it to track dynamic offset voltages in the limiter caused by the simultaneous large amplitude modulation of the carrier.

The signal is FM detected in a quadrature demodulator and then integrated as shown in Figure 13. The limited amplitude r.f. carrier

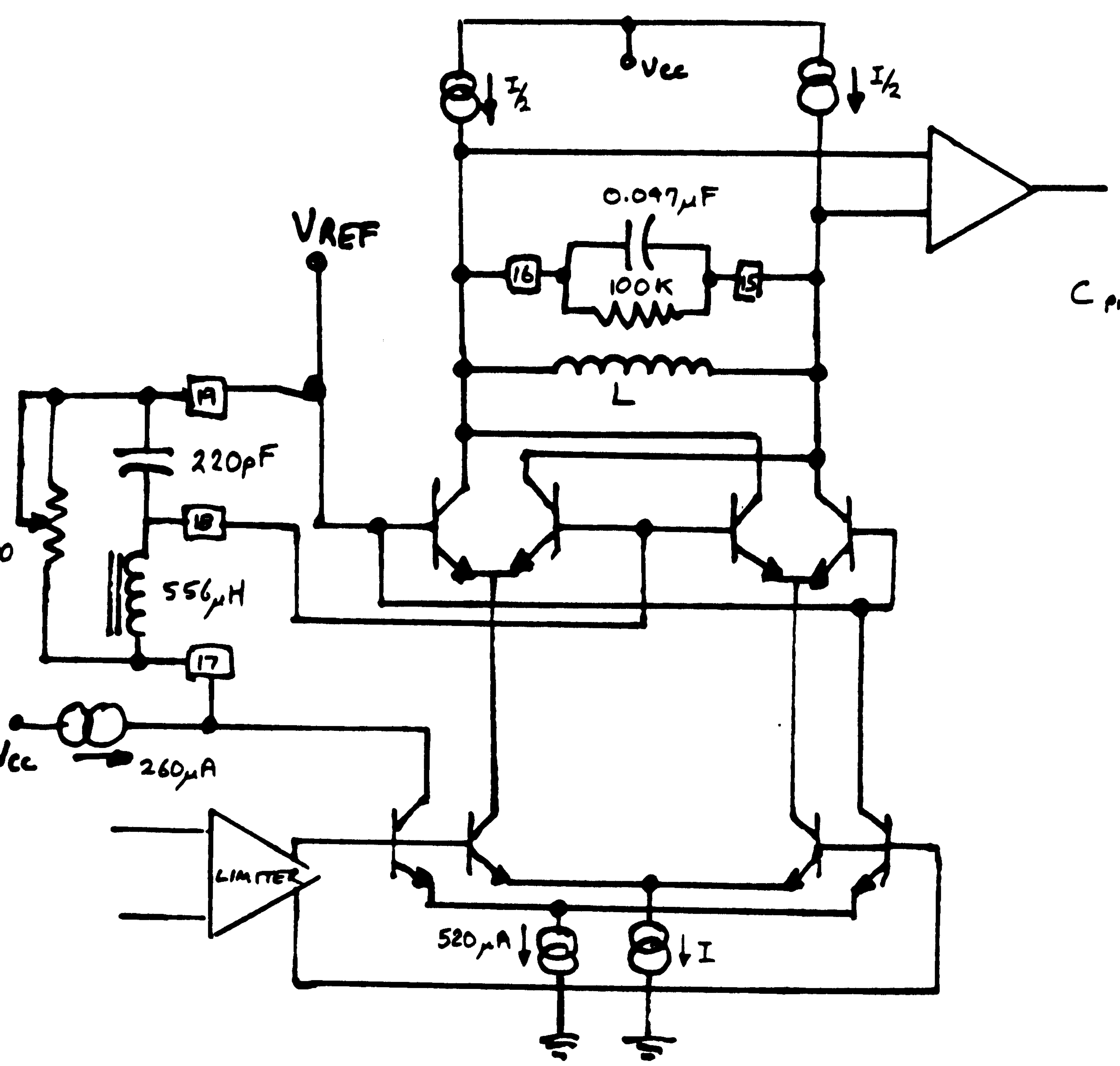


$$I_{PIN\,10} \simeq 1mA \text{ (HOLD)}$$

$$R_{PIN\,10} = \frac{V_{cc} - 0.7}{1 \times 10^{-3}}\ \Omega$$

$$C_{PIN\,6} \leq 0.002\mu F$$

FIGURE 12(b) OUTPUT SAMPLE AND HOLD CIRCUIT



$$C_{PIN(18-19)} \leq 227pF$$

$$L_{(17-18)} = \frac{1}{\omega^2 C_{(18-19)}}$$

$$\frac{Q_L}{Q_U} = \frac{R_D}{R + R_D}$$

FIGURE 13 P.M DETECTOR CIRCUIT

signal switches one set of balanced input ports to the multiplier, while the other ports are receiving the signal after it passes through a resonant circuit tuned to the carrier frequency. A series resonant circuit is used, driven from a current source in order to minimize radiation at the carrier frequency. Since this available signal current is typically 130μA (peak), the capacitor is chosen to ensure switching amplitude (about $400mV_{(p-p)}$) at the multiplier inputs.

$$|\omega C| \geqslant \frac{260 \times 10^{-6}}{400 \times 10^{-3}} \geqslant 0.65 \times 10^{-3}$$

$$\therefore C \leq 227 pF$$

Choosing C to be 220pF gives the inductor nominal value as

$$L = \frac{1}{(2\pi \times 455 \times 10^{3})^{2}\, 220 \times 10^{-12}} = \underline{556 \mu H}$$

Although both sets of input ports to the multiplier are switched, the conversion gain can change since the multiplier operating currents are internally set. In order to avoid different levels of (L - R) from part to part we need to provide some external means for adjusting the detector conversion gain. This is most conveniently done by a potentiometer connected across the tuned circuit which will dominate the circuit Q if the inductor series resistance (R_o) is low. With a coil Q_u of 110, R_D = 14.5Ω at 455kHz. If we choose a tuned circuit bandwidth of ±30kHz then the loaded Q will be 7.5. For a series tuned circuit with a shunt resistance R we have

$$\frac{Q_L}{Q_u} = \frac{R_D}{R + R_D} \qquad (13)$$

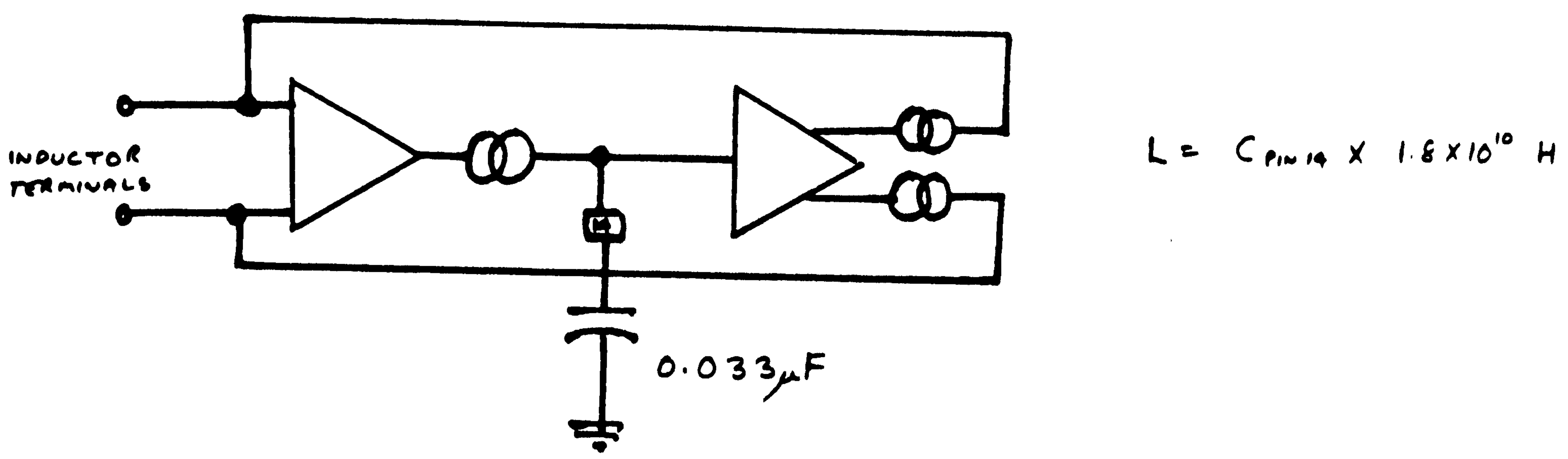


FIGURE 14 SIMULATED INDUCTOR FROM TWO O.T.A's

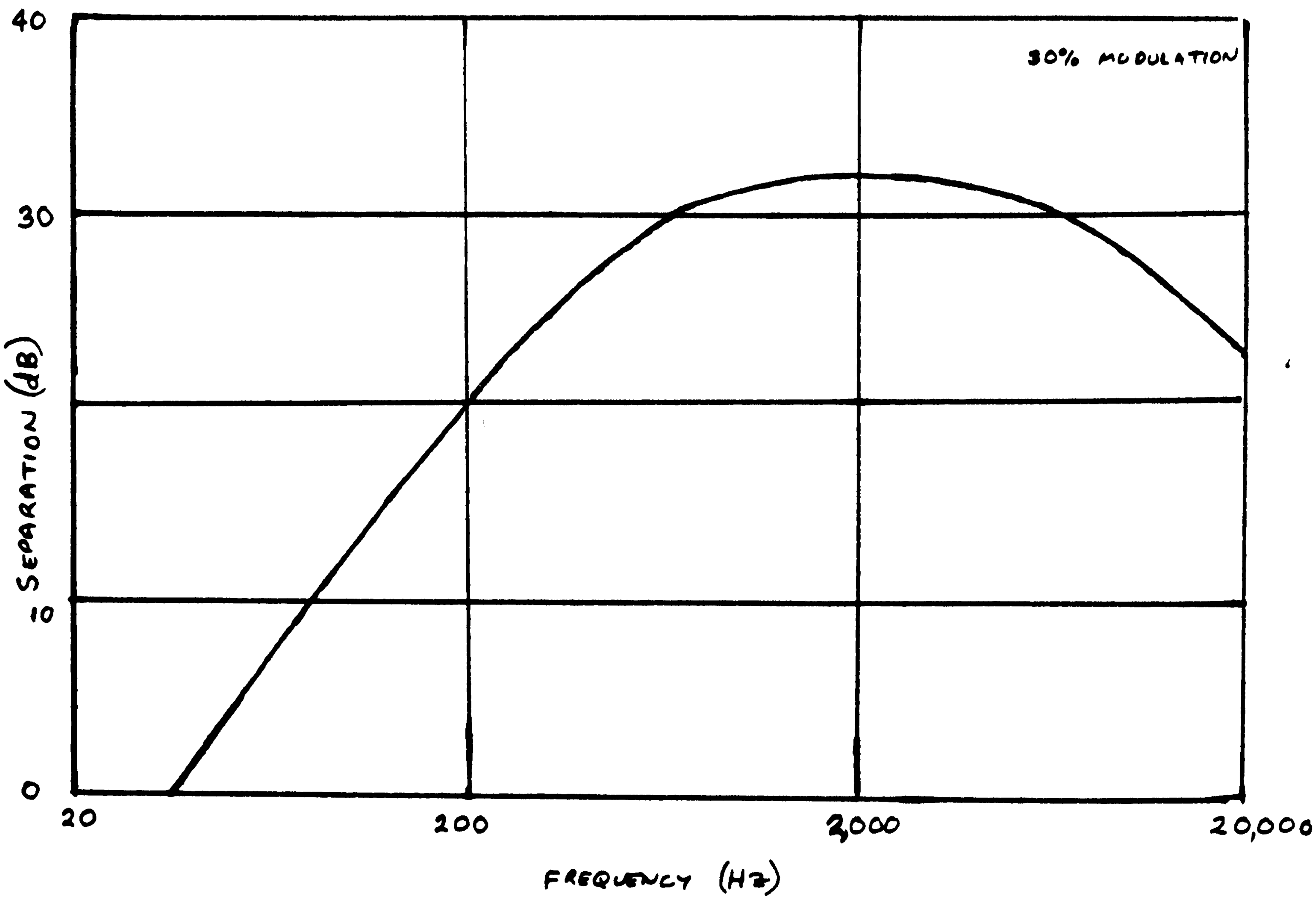


FIGURE 15 SEPARATION VS FREQUENCY

and the potentiometer resistance will be approximately 193Ω. Now that suitable components have been chosen for the resonant circuit that yield a gain adjustment and the proper bandwidth, it becomes possible to select the integrating capacitor that gives the nominal (L - R) detected level necessary for best separation. With a 500Ω pot set to 200Ω the capacitor value is 0.047μF.

Also connected across the multiplier output is an active inductor which serves a two-fold purpose. First, it places a dc short across the multiplier outputs preventing offset voltages exceeding the dynamic range of the next stage. These offsets can occur because of mismatching in the multiplier active devices but, more importantly, as the tank circuit is tuned for resonance (or the input signal is mistuned) dc terms are present in the multiplier output which must be cancelled if the output linear operating range is not to be exceeded. Secondly the inductor is tuned with the integrating capacitor to produce a low frequency pole in the output. Since we would like some means for detecting excess phase signals for noise suppression purposes, the detected stereo identification tone peak voltage must be reduced below the peak audio voltage (remember, in the Magnavox system, the detected audio is riding up and down on a 5Hz waveform that is 4 times the peak audio amplitude). This is done by choosing the pole frequency to be around 30Hz. The capacitor has already been determined as 0.047μF, so a 30Hz pole requires that the inductor be almost 600H!! Hence an active inductor is used.

Two operational transconductance amplifiers (OTA's) are used to realize the simulated inductor, Figure 14. The actual inductance value is proportional to the size of the capacitor at Pin 14 such that

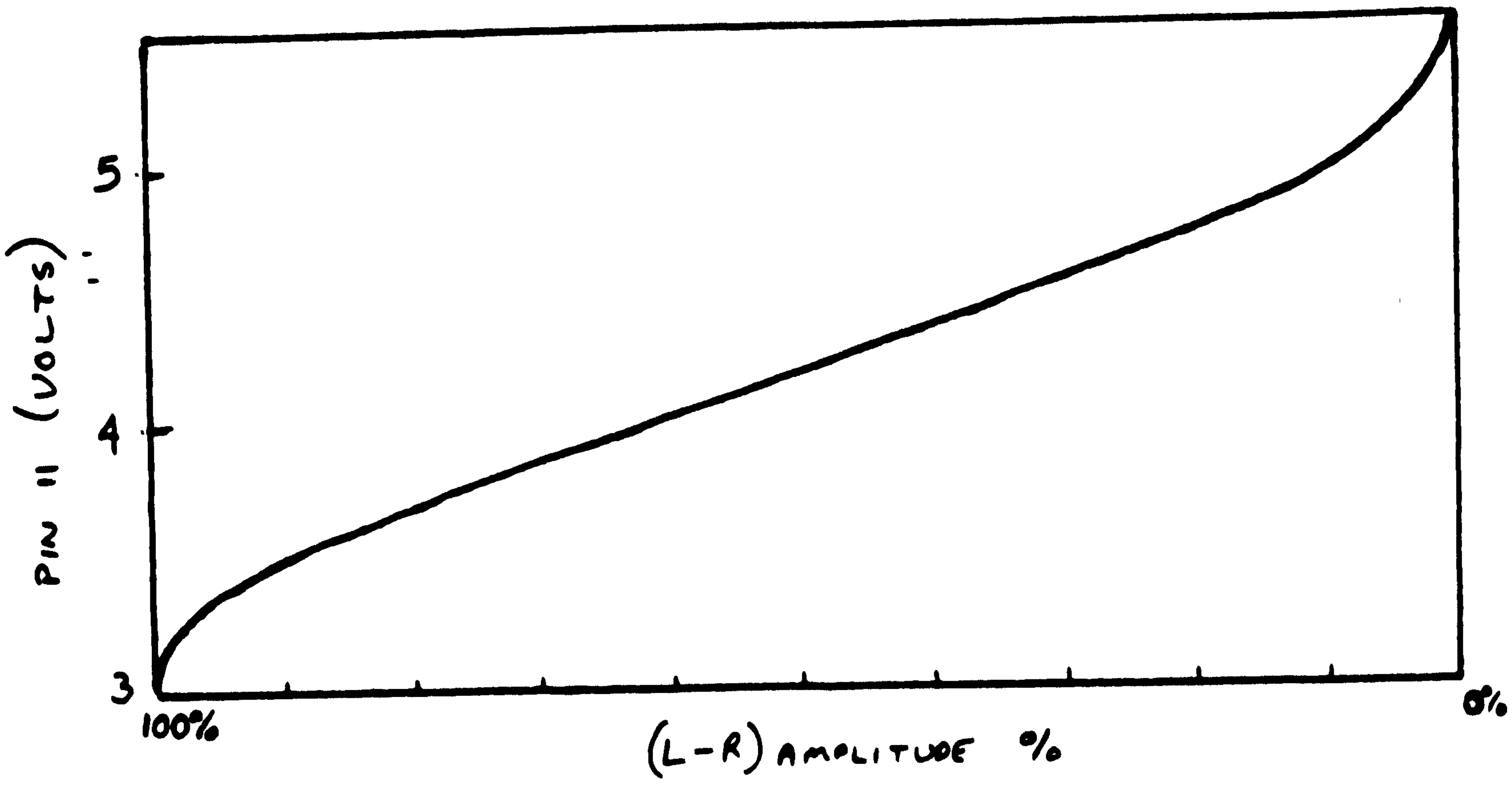


FIGURE 16 STEREO/MONO BLEND CONTROL CHARACTERISTIC

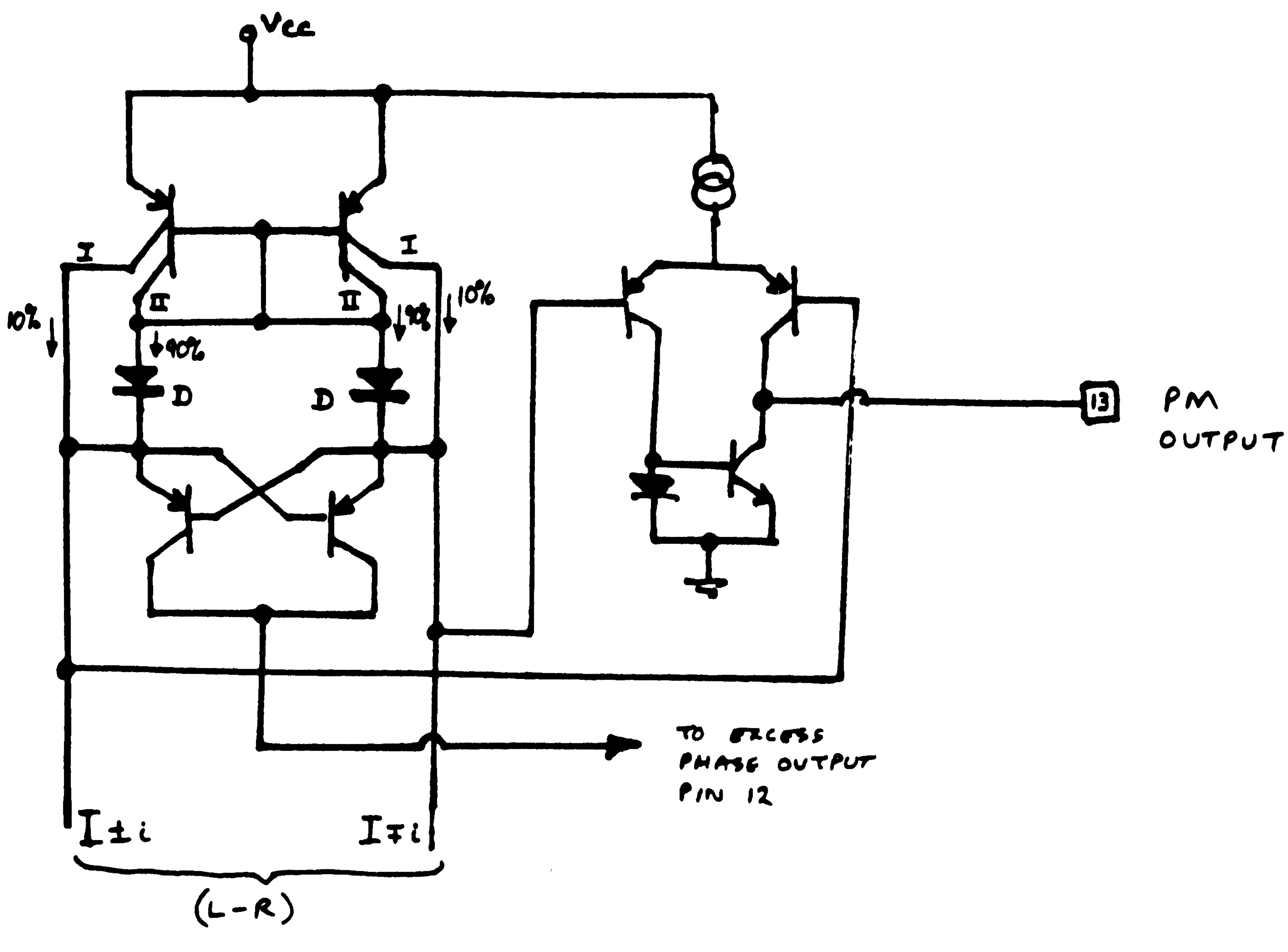


FIGURE 17 EXCESS PHASE DETECTOR

$$L_{(\text{HENRIES})} \simeq C k \qquad (14) \qquad k = 1.8 \times 10^{10}\ H/\mu F \quad C \text{ in } \mu F$$

For 600H, we need a 0.033µf capacitor. A 100kΩ resistor is shunted across Pins 15 and 16 to prevent peaking in the low pass filter.

Alignment of the detector (in fact the entire alignment for the decoder) is straightforward. The blend input is shorted to ground and the coil is tuned for resonance at 455kHz by adjustment until the voltage on the capacitor at Pin 14 matches the reference voltage at Pin 19. Because the charge current for the capacitor is only 1µA, this point is easily loaded and the best way to connect a DVM is <u>between</u> Pins 14 and 19 rather than from Pin 14 to ground. Next, with Left only or Right only information modulating the carrier, the potentiometer between Pins 17 and 19 is adjusted for maximum separation. For example, with Right only signals, the potentiometer is set to minimize the Left output signal at Pin 9. At 1kHz, 30% PM the Left signal level should be -30dB down, Figure 15.

The detected (L - R) signal is connected internally to the matrix through a variable gain block controlled by the average level of the I.F. carrier - the voltage developed at Pin 5. By this means the separation adjustment carried out above will not be impaired by the a.g.c. characteristics of the receiver. A second gain control is placed between the (L - R) signal and the matrix. This gain block is controlled by the mute/blend input at Pin 11 and enables stereo/mono switching or a gradual blend from stereo to mono as the r.f. signal deteriorates. The control amplifier is equivalent to a differential pair biassed at V_{REF} with 5kΩ between the bases and a series 50kΩ to Pin 11. For a 100mV differential across the pair

$$V_{PIN\,11} \geqslant V_{REF} \pm 100 \times 10^{3} \times \left(\frac{50+5}{5}\right)$$
$$= V_{REF} \pm 1.1\,\text{Volts}$$

A control curve is shown in Figure 16 and above 5.36V the decoder will be in the mono mode; below 3.16V the decoder will be in the stereo mode.

The last part of the decoder to be described is the PM output and excess phase detector, Figure 17. A buffered differential (L - R) signal current from the PM integrating capacitor is supplied to this circuit which has a dc to ac ratio such that each degree of detected phase deviation causes a 1.2% change in the current drawn from each side of the detector. For each side, with no signal, 10% of the total current is supplied by collector I of the upper PNP device. The remaining 90% is supplied by collector II through a diode D. When a signal is present, as the peak phase deviation increases, one side will draw more current and one side correspondingly less. The differential current component is buffered to Pin 13 to provide an output for the stereo identification tone detector. A 30% modulated audio signal will cause Pin 13 to sink and source 54µA and the stereo identification tone, reduced in level by the PM detector low pass filter, produces 14µA peak.

When the peak phase deviation exceeds 75° this indicates that noise must be present and the peak current drawn in one side is less than 10% of the quiescent value. Collector II will be transferring all its current to the other side (which will be drawing greater than 190% of the quiescent

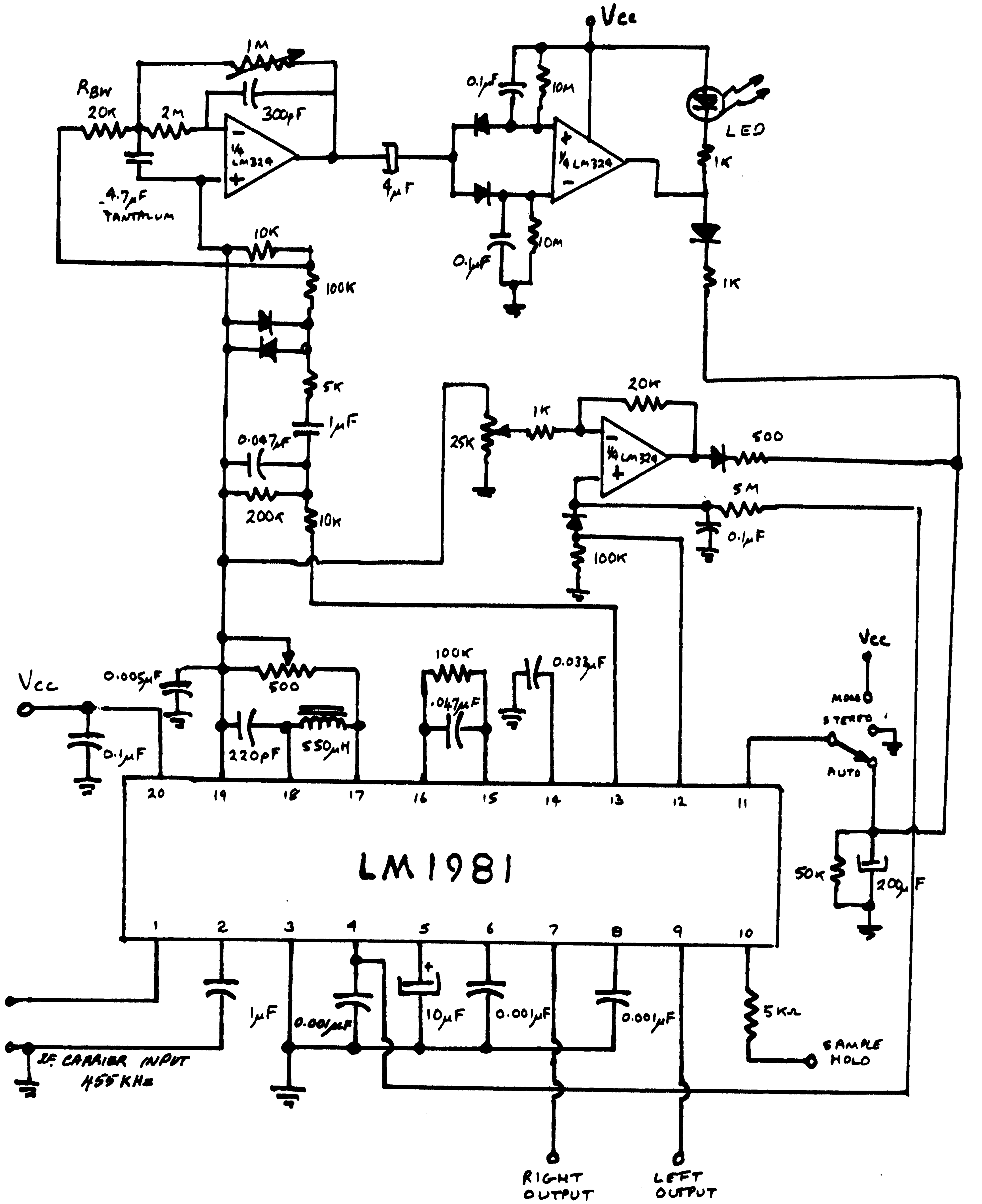


FIGURE 18 MAGNAVOX SYSTEM DECODER WITH 5Hz DETECTOR

current) and the diode for collector II will cut off. The difference between the current demanded, and that delivered by collector I is coupled over to Pin 12 which then provides an excess phase indication.

A complete Magnavox system decoder is shown in Figure 18, including a suggested stereo tone detector. This consists of a 5Hz filter followed by a full-wave rectifier connected to the stereo/mono blend input Pin 11. The excess phase output is filtered and applied to Pin 11, as is a filtered voltage from the absolute value detector. Excess phase outputs will force Pin 11 above V_{REF}, giving automatic switching into mono in the presence of noise in the PM channel. Similarly a weak r.f. signal will cause switching into mono. As the r.f. signal improves, the dc level at Pin 4 will go below V_{REF} enabling the stereo mode if a stereo tone is simultaneously present.

Decoding other AM Stereo Systems

This note has described the Mangavox AM/PM system in some detail in order to show the system requirements for adding stereo information to the present monophonic AM broadcast signal. Alternative compatible AM stereo systems have been proposed by Motorola, Belar, Harris and Kahn. Since all these alternate proposals depend on some form of angle modulation of the r.f. carrier to convey the additional information, the LM1981 can be used as a decoder. The following is not intended as a complete description of such decoders, or to imply that a universal decoder is practical, but more as a suggestion for the circuit modifications that will be necessary if the AM stereo test signal source is other than the Magnavox AM/PM signal.

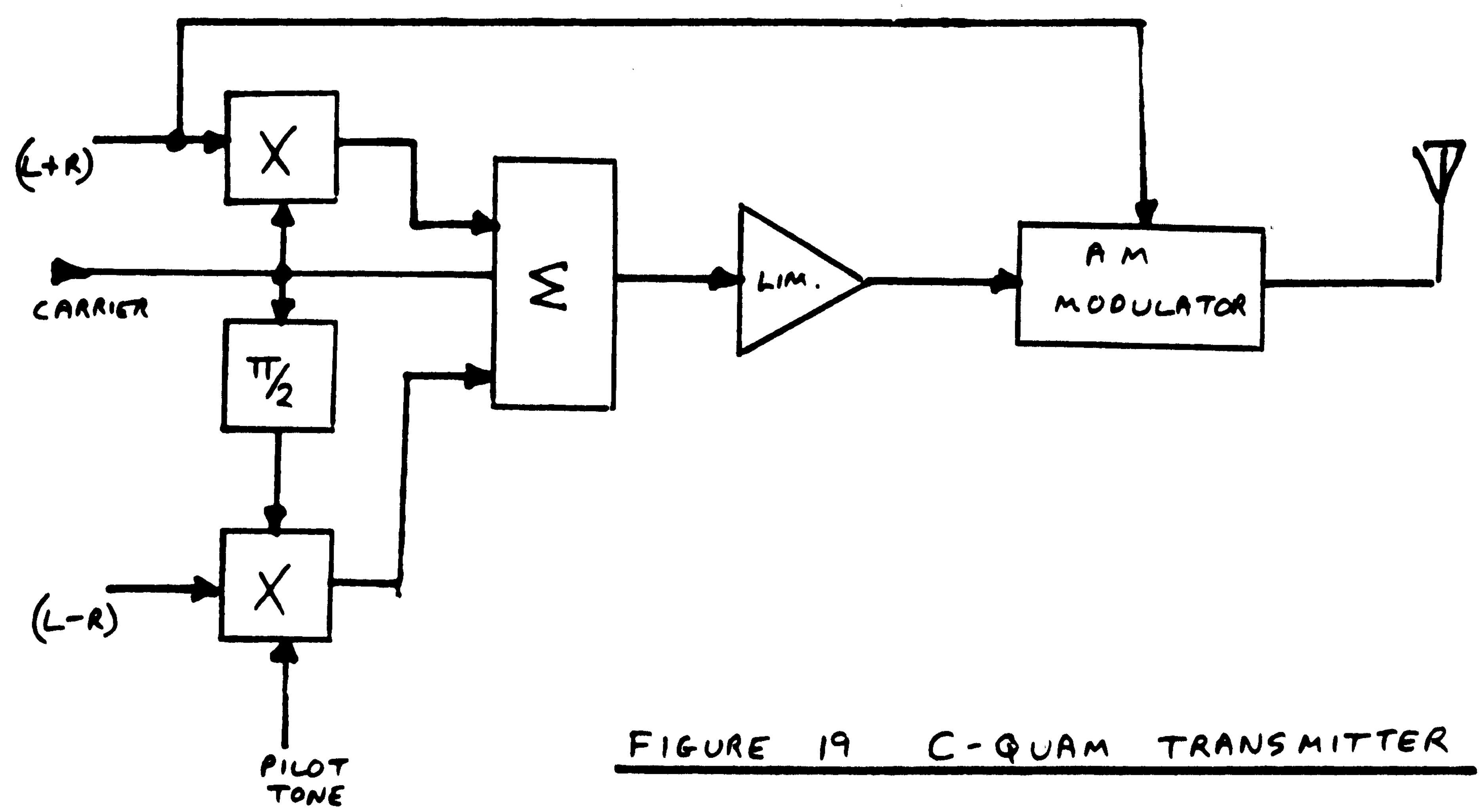


FIGURE 19 C-QUAM TRANSMITTER

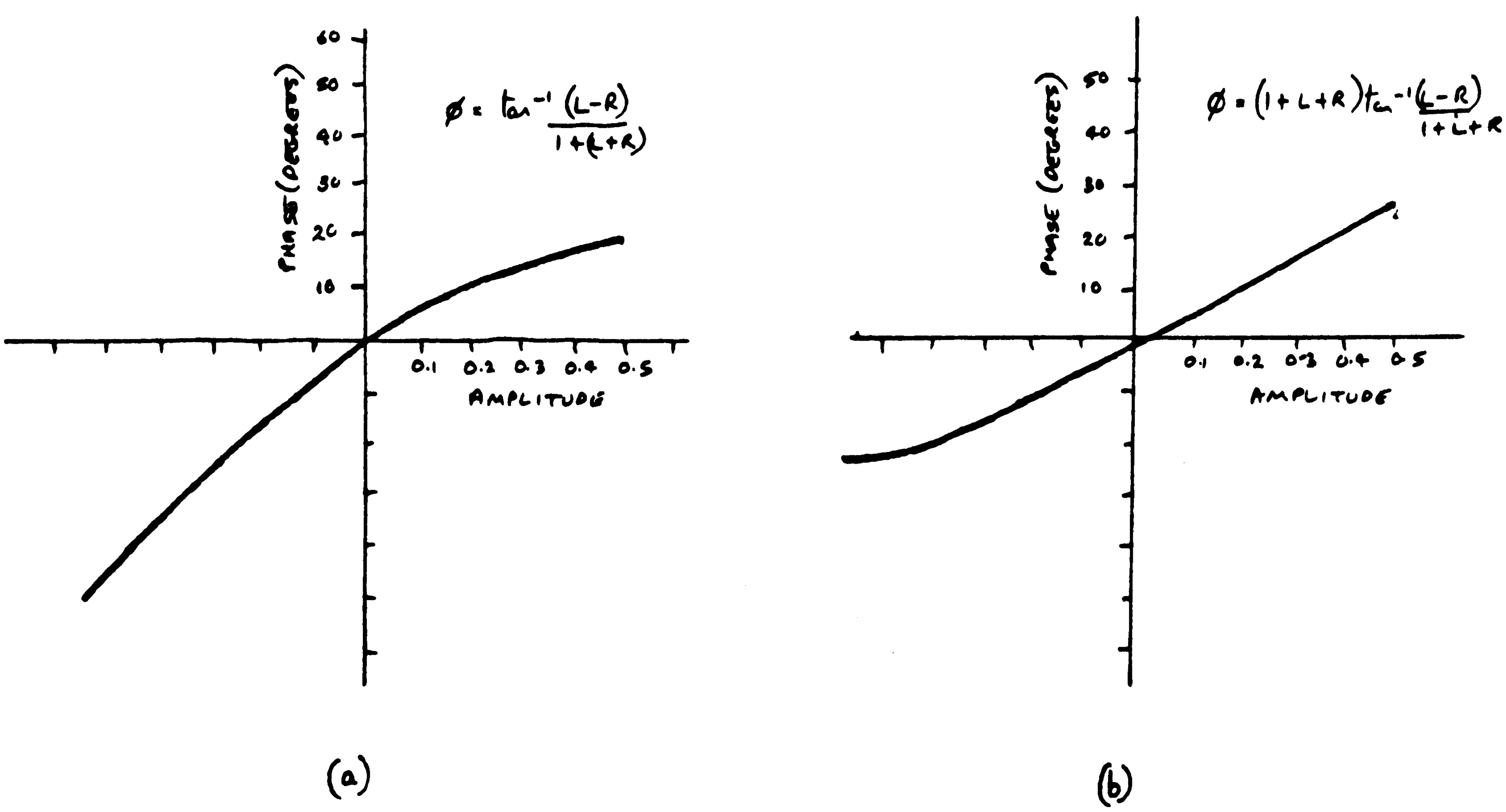


FIGURE 20. DETECTOR TRANSFER FUNCTIONS

Motorola C-QUAM

This system for compatible AM stereo uses a technique similar to that employed in color television for the simultaneous transmission of two independent color signals on the same color subcarrier. The (L - R) and (L + R) signal components are quadrature modulated onto the r.f. carrier as shown in the block diagram of Figure 19. After limiting to retain only the phase variations, the carrier is amplitude modulated with the monophonic (L + R) signal. A stereo identification tone of 25Hz is included with the (L - R) modulating signal. The equation of the C-QUAM carrier is

$$e_c = E_o\left[1 + m\cos\omega_m t\right]\cos\left[\omega_c t + \tan^{-1}\left(\frac{m\cos\omega_s t + 0.05\sin 50\pi t}{1 + m\cos\omega_m t}\right)\right] \qquad (15)$$

Notice the similarity between Equation (15) and Equation (6) with the exception of the carrier phase modulating term. Ignoring the pilot term we have

$$\text{For Magnavox} \qquad \theta = m\cos\omega_s t \qquad (16)$$

$$\text{For Motorola} \qquad \theta = \tan^{-1}\frac{m\cos\omega_s t}{(1 + m\cos\omega_m t)} \qquad (17)$$

Therefore, if we use the decoder of Figure 18, for a Left only or Right only signal the detector transfer function would be similar to Figure 20(a) and the (L - R) audio signal would be distorted if the signal source is the Motorola system. Now consider the function

$$\theta' = (1 + m\cos\omega_m t)\tan^{-1}\frac{m\cos\omega_s t}{(1 + m\cos\omega_m t)} \qquad (18)$$

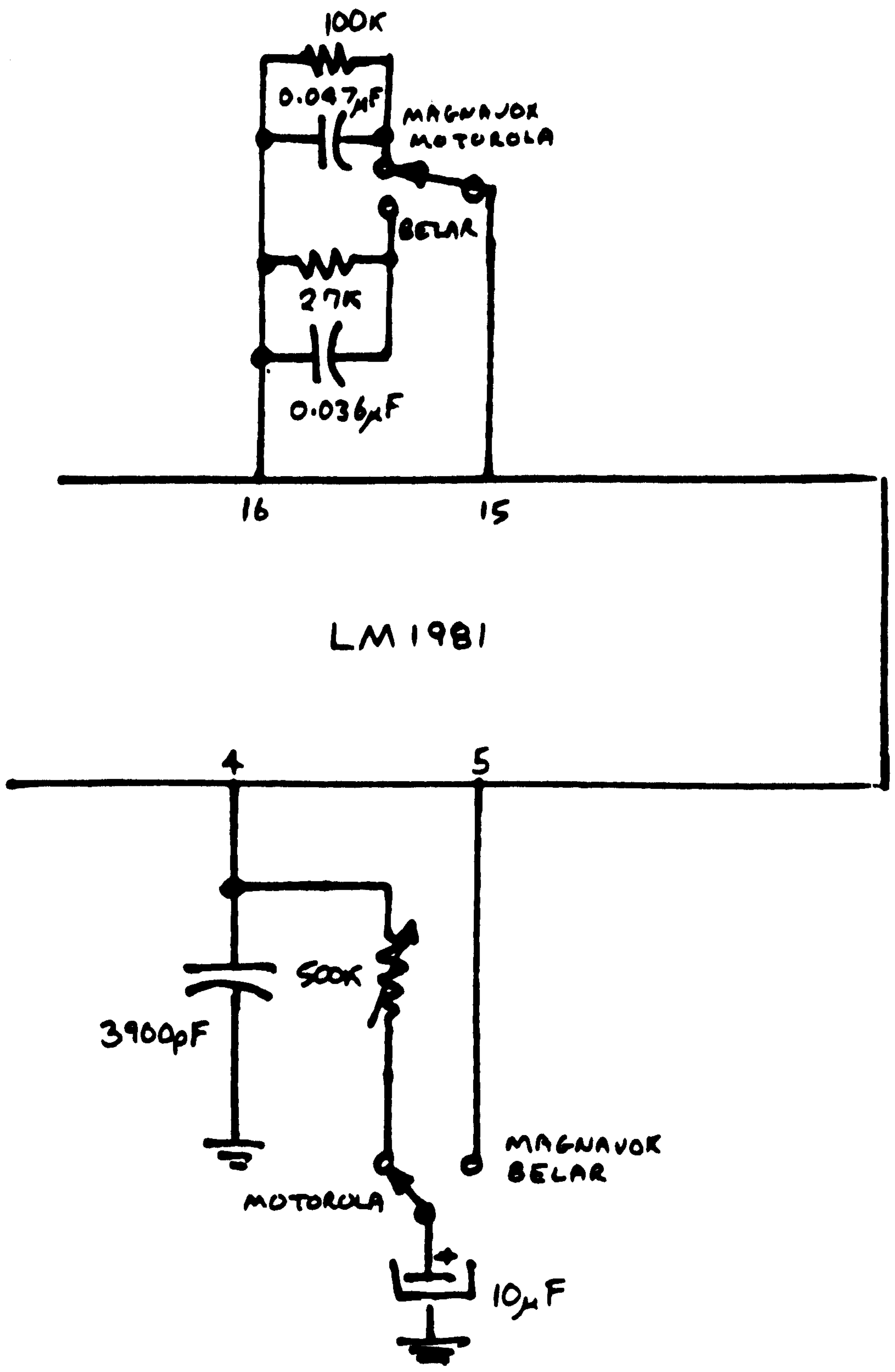


FIGURE 21 DECODER MODIFICATIONS FOR MOTOROLA/BELAR SYSTEMS

The transfer function changes to that of Figure 20(b) which is much more linear, at least up to 50% modulation. Above 50% modulation the distortion, predominantly second harmonic, begins to increase. Recalling that extremely high stereo modulation depths are unlikely to be encountered in practice, if we modify the detected PM function of the LM1981 with a $(1 + M \cos \omega_{mt})$ term, then this decoder will do a reasonable job of detecting a Motorola compatible AM stereo signal. Since $(1 + M \cos \omega_{mt})$ is the amplitude modulation signal (L + R), our correction term is already available at Pin 4 of the I/C. Instead of adjusting the (L - R) detected level in response to the average carrier level, we can remove the filter from Pin 5 allowing the (L - R) level to change with the amplitude modulation. Practical experiments with this circuit have shown that the distortion level can be reduced if the correction term is actually $(1.14 + M \cos \omega_{mt})$. This modification is implemented by decreasing the a-c term as shown in Figure 21. Re-adjustment of the potentiometer across Pins 17 and 19 restores the stereo separation.

BELAR AM-FM

The Belar system uses the same (L + R) and (L - R) signal components but with the (L - R) signal frequency modulating the carrier. A maximum frequency deviation of ±1.25kHz and a pre-emphasis time constant of 100µS (f_o = 1.6kHz) are the significant parameters that need to be considered for the decoder modification. The Belar carrier equation is

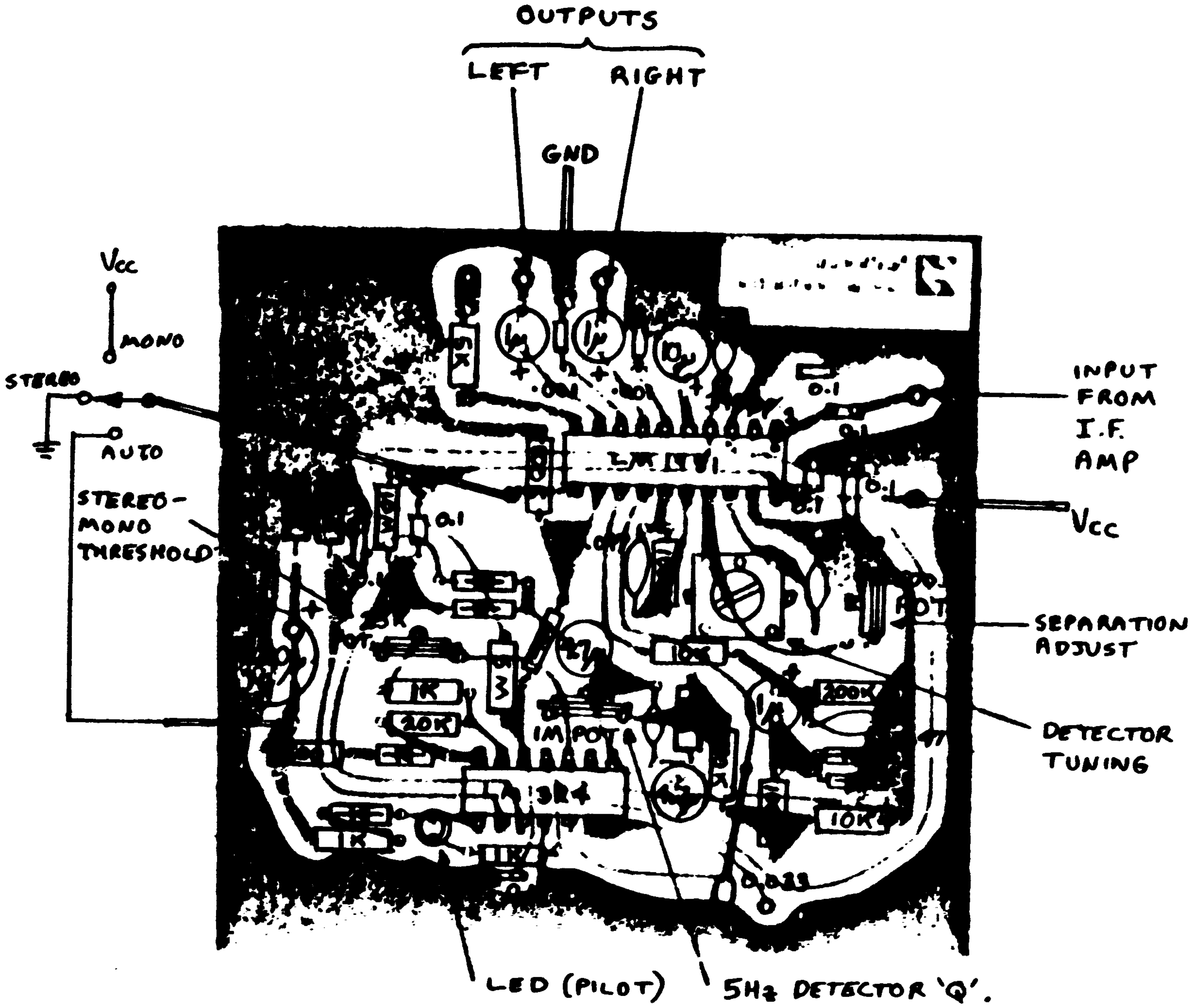
$$e_c = E_o \left[1 + m \cos \omega_m t \right] \cos \left[\omega_c t + \frac{1250}{f_s} \sqrt{1 + \left(\frac{f_s}{f_o}\right)^2} \times m \sin \left(\omega_s t + \tan^{-1} \frac{f_s}{f_o} \right) \right]$$

For this signal, the LM1981 will operate as a conventional FM demodulator with de-emphasis above 1.6kHz as shown in Figure 21. The detected signal level with this modulation method is reduced (compared to the Magnavox system) by the factor

$$\frac{1250}{f_s}\sqrt{1+\left(\frac{f_s}{f_0}\right)^2} = 0.75 \ @ \ 1.6\text{KHz}$$

To compensate for this the capacitor between Pins 15 and 16 is reduced to 3600pF, thus determining the resistor value as 27kΩ for a 1.6kHz corner frequency.

A complete decoder p.c.b. for the Magnavox system, based on the circuit of Figure 18 is shown in Figure 22 along with a component stuffing guide. Set-up is straightforward, following the procedure outlined earlier. Time constraints have prevented a full evaluation of the board modified for the Motorola system and the development of the detector for the Motorola pilot tone. Operation of the I/C with the other AM stereo systems appears feasible but time and the lack of appropriate encoders has prevented practical implementation at present.



COMPONENT SIDE

COPPER SIDE

FIGURE 22 MAGNAVOX DECODER P.C.B.

National Semiconductor

2900 Semiconductor Drive
Santa Clara, California 95051
Tel.: (408)737-5000
TWX: (910)339-9240

AM Stereo Generator

Below is an AM Stereo Generator designed by National's Central Applications Group for demonstration of AM stereo receivers. The generator uses an Armstrong phase modulator at one third carrier frequency for audio difference channel and direct FM for the 5 Hz pilot. The LED bar-graph display modules show left and right or (left + right) and (left − right) signal levels.

The industry's first AM stereo decoder, the LM1981, is being made available to designers by NSC. Most existing AM radio front ends will probably not meet the requirements for AM stereo without substantial improvement in at least two major areas:

- phase noise of the local oscillator
- IF symmetry

By using the LM1981 now, designers can begin conversion of mono systems to stereo in order to evaluate these areas.

Thomas B. Mills
Staff Engineer
10 February 1981