

6. Equation (2.11.3):

$$C_1 = \frac{3180 \times 10^{-6}}{360 \times 10^3} = 0.0088 \mu\text{F}$$

Put $C_1 = 0.01 \mu\text{F}$

7. At 1 kHz the feedback network impedance (z) = 37.6k \angle 49°. Equation (2.11.7):

$$\text{0dB reference gain} = 100 = \frac{37.6 \times 10^3}{R_6} + 1$$

$$\therefore R_6 = \frac{37.6 \times 10^3}{99} = 379 \Omega$$

Put $R_6 = 390 \Omega$

8. From Equation (2.6.4):

$$\left(\frac{V_{CC}}{2.6} - 1\right) R_5 = R_1 + R_2$$

$$\therefore R_5 = \frac{390 \times 10^3}{8.23} = 47 \text{ k}\Omega$$

Note: This value of R_5 will center the output at the mid supply point. However, for symmetrical clipping it is worth noting that the LM387 can swing to within 0.3V of ground and 1.7V of V_{CC} . To put the output midway between these points (11.2V DC with $V_{CC} = 24\text{V}$), put $R_5 = 56 \text{ k}\Omega$.

9. From Equation (2.6.10):

$$C_4 = \frac{1}{2\pi f_0 R_6}$$

$$= \frac{1}{2\pi \cdot 10 \cdot 390}$$

$$= 40.8 \times 10^{-6}$$

Put $C_4 = 47 \mu\text{F}$

The completed design is shown in Figure 2.11.5 where a 47k Ω input resistor has been included to provide the RIAA standard cartridge load.

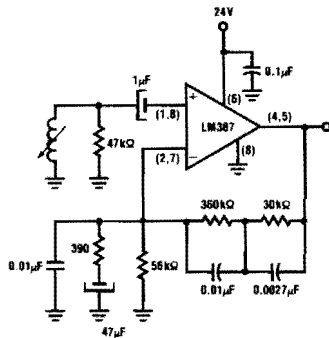


FIGURE 2.11.5 LM387 Phono Preamp (RIAA)

The LM381 integrated circuit may be substituted for the LM387 in Figure 2.11.5 by making the appropriate pin number changes.

2.11.5 LM382 Phono Preamp

By making use of the internal resistor matrix, a minimum parts count low noise phono preamp is possible using the LM382 (Figure 2.11.6). The circuit has been optimized for a supply voltage equal to 12-14 V. The midband 0dB reference gain equals 46dB (200V/V) and cannot easily be altered. For designs requiring either gain or supply voltage changes, the required extra parts make selection of a LM381 or LM387 more appropriate.

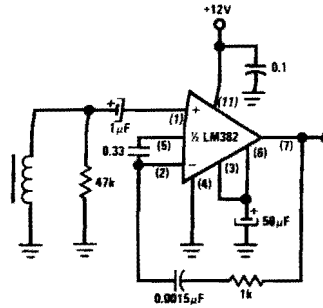


FIGURE 2.11.6 LM382 Phono Preamp. (RIIA)

2.11.6 LM1303 Phono Preamp

The LM1303 allows a convenient low noise phono preamp design when operating from split supplies. The circuit appears as Figure 2.11.7. For trimming purposes and/or gain changes the relevant formulas follow:

$$\text{0dB Ref Gain} = 1 + \frac{R_2}{R_3} \quad (2.11.5)$$

$$f_1 = \frac{1}{2\pi R_1 C_1} \quad (2.11.6)$$

$$f_2 \approx \frac{1}{2\pi R_2 C_1} \quad (2.11.7)$$

$$f_3 = \frac{1}{2\pi R_2 C_2} \quad (2.11.8)$$

As shown in Figure 2.11.7, the 0dB reference gain (1kHz) equals about 34dB and the feedback values have been altered slightly to minimize pole-zero interactions.

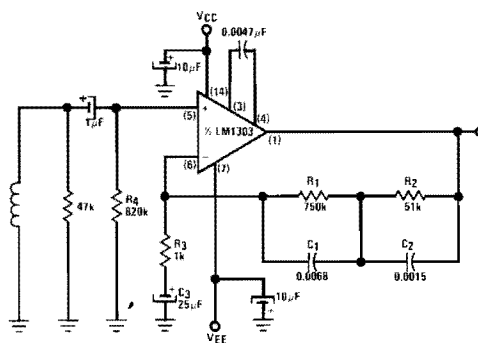


FIGURE 2.11.7 LM1303 Phono Preamp. (RIAA)

2.11.7 Inverse RIAA Response Generator

A useful test box to have handy while designing and building phono preamps is one which will yield the opposite of the playback characteristic, i.e., an inverse RIAA (or record) characteristic. The circuit (Figure 2.11.9) is achieved by adding a passive filter to the output of an LM387, used as a flat-response adjustable gain block. Gain is adjustable over a range of 24 dB to 60 dB and is set in accordance with the 0 dB reference gain (1 kHz) of the phono preamp under test. For example, assume the preamp being tested has +34 dB gain at 1 kHz. Connect a 1 kHz generator to the input of Figure 2.11.9. The passive filter has a loss of -40 dB at 1 kHz, which is corrected by the LM387 gain, so if a 1 kHz test output level of 1 V is desired from a generator input level of 10 mV, then the gain of the LM387 is set at +46 dB (+46 dB - 40 dB + 34 dB = 40 dB ($\times 100$); 10 mV $\times 100 = 1$ V). Break frequencies of the filter are determined by Equations (2.11.9)-(2.11.11).

$$f_1 = 50 \text{ Hz} = \frac{1}{2 \pi R_9 C_4} \quad (2.11.9)$$

$$f_2 = 500 \text{ Hz} \approx \frac{1}{2 \pi R_{10} C_4} \quad (2.11.10)$$

$$f_3 = 2120 \text{ Hz} = \frac{1}{2 \pi R_{10} C_5} \quad (2.11.11)$$

The R7-C3 network is necessary to reduce the amount of feedback for AC and is effective for all frequencies beyond 20 Hz. With the values shown the inverse RIAA curve falls within 0.75 dB of Table 2.11.1.

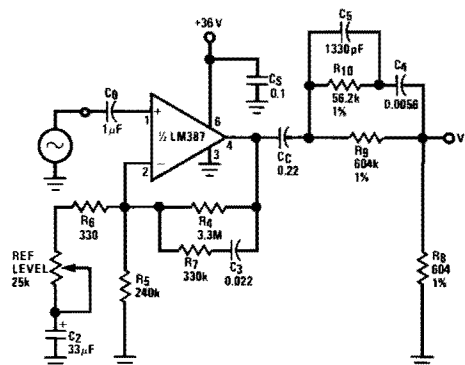


FIGURE 2.11.9 Inverse RIAA Response Generator

2.12 TAPE PREAMPLIFIERS

2.12.1 Introduction

A simplified diagram of a tape recording system is shown in Figure 2.12.1. The tape itself consists of a plastic backing coated with a ferromagnetic material. Both the record and erase heads are essentially inductors with circular metal cores having a narrow gap at the point of contact with the tape. The tape coating then forms a low reluctance path to complete the magnetic circuit. As the tape moves across the record head gap, the magnetic field at the trailing edge of the gap leaves the tape coating permanently magnetized with a remanent flux level (Φ_R) proportional to the signal current in the record head windings.

The bias and erase currents (I_B and I_E) are constant amplitude and frequency waveforms (between 50 kHz and 200 kHz) generated by the bias oscillator. In the erase head, the amplitude of the waveform (from 30 Volts to 150 Volts typically) will determine the degree to which previously recorded signals are "erased" from the tape — in a good machine this will be from 60 dB to 75 dB below the normal recording level. This same waveform, reduced in amplitude to between 5 to 25 times the maximum recording signal level, is used in the record head to determine the "operating point" of the magnetic recording process. Distortion, maximum output level and sensitivity are strong functions of the bias level. To gain some insight into the need for bias, let us take a closer look at the recording process.

Figure 2.12.2(a) shows the permanent magnetization B_r (or remanent flux) of a short section of magnetic tape, obtained by applying a magnetizing field H produced by a dc current in the record head winding. This curve is clearly non-linear and if an ac signal current was used in the head winding a highly distorted recording would be made. One solution would be to apply a steady dc bias to the record head along with the ac signal so that the tape was always magnetized in a linear region of the curve (between points A and B for example). This method, called *dc bias*, uses only one part of the curve and reduces the distortion but has a very poor S/N ratio. An improvement may be obtained by pre-magnetizing the tape to saturation and using a dc bias on the record head to bring the magnetization back to zero. Even so, S/N ratios above 30 dB are not easy to achieve.

For high S/N ratios and low distortion, another method called *ac bias* is used.

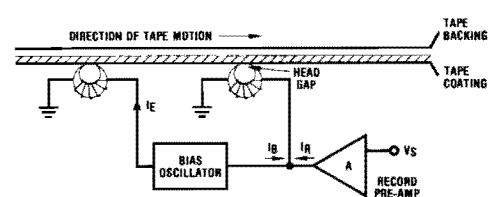


FIGURE 2.12.1 Simplified Recording System