DESIGNING THE TAPR TNC AUDIO INPUT FILTER

Margaret Morrison, KV7D and Dan Morrison, KV7B
4301 E. Holmes Street
Tucson, Arizona 85711
602-327-4775

Standard modulation for present terminal node controllers is Bell 202 compatible 1200 Hz / 2200 Hz phase-continuous FSK. Fig. 1 shows the typical spectral characteristics of such data. Notice that frequencies ranging from about 500 Hz to 2900 Hz are present in random NRZI data. One might guess that frequencies outside the central region from 1000 Hz to 2400 Hz could be eliminated with no degradation of demodulator performance. This turns out to be incorrect: the demodulator PLL needs this information to ensure timely response to data transitions. Thus, the ideal audio response over the link should be flat from below 500 Hz to over 2900 Hz.

In fact, the audio response of a typical FM 2-meter link looks like Fig. 2. This response curve shows dramatic rolloff over the frequency range of interest. Most of the rolloff is due to receiver audio characteristics and, in fact, some transmitted signals actually seem more like phase-modulation rather than FM, with high frequency emphasis. In many cases, the demodulator on the TAPR TNC cannot handle this rolloff, and successful demodulation of these signals requires reduced baud rates. We decided to put in a filter ahead of the demodulator to alleviate this problem.

The filter needed to take up as little space as possible, to be simple to modify, and most importantly, to be effective. In view of the fact that a variety of clock signals were already available on the TAPR TX, we opted for a switched-capacitor filter design, since ease of modification and parts-count make this a clear winner in the design competition. We quickly realized that in order t0 produce frequency compensation over a very wide range at least two sections would be required. Lyle Johnson, our hardware guru, thus selected the National Semiconductor MF-10 dual section filter as the part of choice, leaving the design details up to us.

Filter Configuration

This marvelous part may be configured in a number of ways as first or second order filter sections depending on external components, and each section may be configured independently of the other. After some experimentation we chose two second order sections -- one set up as a highpass filter, the other as a lowpass filter. This choice optimizes both amplitude and phase response over the wide range of frequencies required.

The manufacturer's literature lists nine different recommended configurations, each yielding a variety of output options. The choice is quickly narrowed to one, mode 3, which provides both highpass and lowpass outputs, and which provides maximum flexibility in selecting filter parameters. In particular, the frequency parameter, f0, is adjustable via resistor choices rather than by clock frequency alone. This, as well as dynamic range considerations, directed us to select mode 3 for both sections. In addition, we chose the option which scales f0 to fclk/100 rather than to fclk/50 in order to have as high a clock rate as possible. This allows the switching noise, which occurs at the clock frequency, to be most easily removed by post-filtering.

Filter Response

The input section, configured as a highpass filter, is shown in Fig. 3a; the second section is configured as a lowpass filter, shown in Fig. 3b. The frequency response for these two sections may be expressed as functions of frequency, f, and the parameters f0, Q, and either highpass gain parameter GLP or lowpass gain parameter GHP. Let $\omega = jf/f0$, where $j^2 = -1$. Then, the highpass filter's (complex) frequency response is

$$h(f) = GHP \frac{\omega^2}{\omega^2 + \omega/Q + 1} \quad (1)$$

and the lowpass response is

$$h(f) = GLP \frac{1}{\omega^2 + \omega/Q + 1} \quad (2)$$

For either, the amplitude response ma; be determined as $|h|$, and the phase response as

$$\phi_{out} - \phi_{in} = \arctan\left[Im(h)/Re(h)\right] \quad (3)$$

The parameters f0H, QH, and f0L, QL are related to the clock frequency, and the programming resistors, R2, R3, and R4, by the following equations:

$$f0 = \frac{fclk}{100}\sqrt{\frac{R2}{R4}} \quad (4)$$

$$Q = \frac{R3}{R2}\sqrt{\frac{R2}{R4}} \quad (5)$$

where F0 is f0H or f0L and Q is QH or QL, depending on the section.

For the highpass section, the gain is GHP = -R2/R1, while for the lowpass section the gain is GLP = -R4/R1. Notice that for either section the input resistor, R1, only affects the overall gain and does not influence the shape of the response.

Optimization Procedure

The design task begins by choosing f0H, QH, f0L, and QL so as to produce as flat an overall filtered response as possible given the unfiltered response. These four parameters are determined by successive interactive computer optimization of the filtered response. A systematic search is made for the optimum parameters for each section alternately.

The optimization of either section consists of a nested iteration which determines its Q in an inner loop and its f0 in an outer loop. The other section's parameters remain fixed during this iteration. The inner loop calculates a least-squares straight line fit to the filtered audio amplitude response (expressed in dB), over a specified range of frequencies, and varies Q to produce a flat response., that is, best-fit slope of zero.

This procedure is written as a function whose value is the RMS deviation of the filtered response from the fit, and which also returns to the outer loop as arguments the value of Q, the slope, and the intercept of the fit. The outer procedure consists of a non-linear optimization (minimization} of the RNS error between the filtered response and the best-fitting frequency-indendent response, as a function of the f0 parameter.

The specified frequency range for the calculation can be varied to achieve best results. Typically, optimizing over the range 1000 Hz to 2400 Hz produces very good results, as rolloff begins well outside the range of optimization. It is helpful to be able to graphically display intermediate response curves during the optimization and to allow only inner-loop iterations on occasion.

Programming the Filter

The remaining work is to determine the eight resistor values (four in each section) which best implement the optimized response functions. Several considerations allow a unique choice to be made. First, for best dynamic range in mode 3, fclk/100 should be as near to f0 as possible. Since both sections run off the same clock, the optimum compromise clock frequency is the geometric mean,

$$\frac{fclk}{100} = \sqrt{f0H \cdot f0L} \qquad (6)$$

In practice, on the TAPR TNC this meant picking the nearest available frequency to this (fclk = 115.2 kHz), since the main crystal clock frequency is a power of two times any of the available values. Having selected the proper clock frequency we are able to determine the resistance values. This is done in the following steps:

1. Notice that in either section the input resistor, R1, does not determine response shape -- only overall gain of the stage. Using Eqs. (4) and (5) determine the ratios R2 : R3 : R4 which affect the shape of the response,

2. Temporarily design each section so as to have unity amplitude response at 1200 Hz. Having thus determined the overall gain parameters GLP and GHP, determine the ratio of R1 to R2 for the highpass section and the ratio of RL to R4 for the lowpass section based on the gain-resistor relationships.

3. Arbitrarily set the lowest resistance value in each section to 10 kilohms. This allows at least two of the eight resistors to be of garden variety. In addition, the resistances will be as low as possible while staying well above the minimum value recommended by the manufacturer (5 kil-ohms). Finish the design by using the ratios Ri : R2 : R3 : R4 to determine the remaining resistors.

4. (Optional) To further ease the task of obtaining resistor values, set both input resistors to 10 kilohms as well. Our experience is that for the filters we are designing, this has little effect on either the dynamic range or the overall gain.

The method outlined here efficiently produces broadband designs. The result of applying this procedure to the unfiltered response shown in Fig. 2 is shown in Fig. 4. The amplitude and phase response of this filter is shown in Figs. 5a and 5b. Using optional step 4 above, the overall gain of the highpass section is 6.73 and that of the lowpass section is 3.73. The resistance values, in kilohms, are:

Highpass	Lowpass
R1 = 10.0	R1 = 10.0
R2 = 63.7	R2 = 16.3
R3 = 58.3	R3 = 10.0
R4 = 10.0	R4 = 37.3

A single FORTRAN program was written to perform the design steps, menu driven and producing graphical display of the response as the design proceeds. A listing is available for the cost of photocopying, and the source is available provided a 9-track tape is sent to the authors..

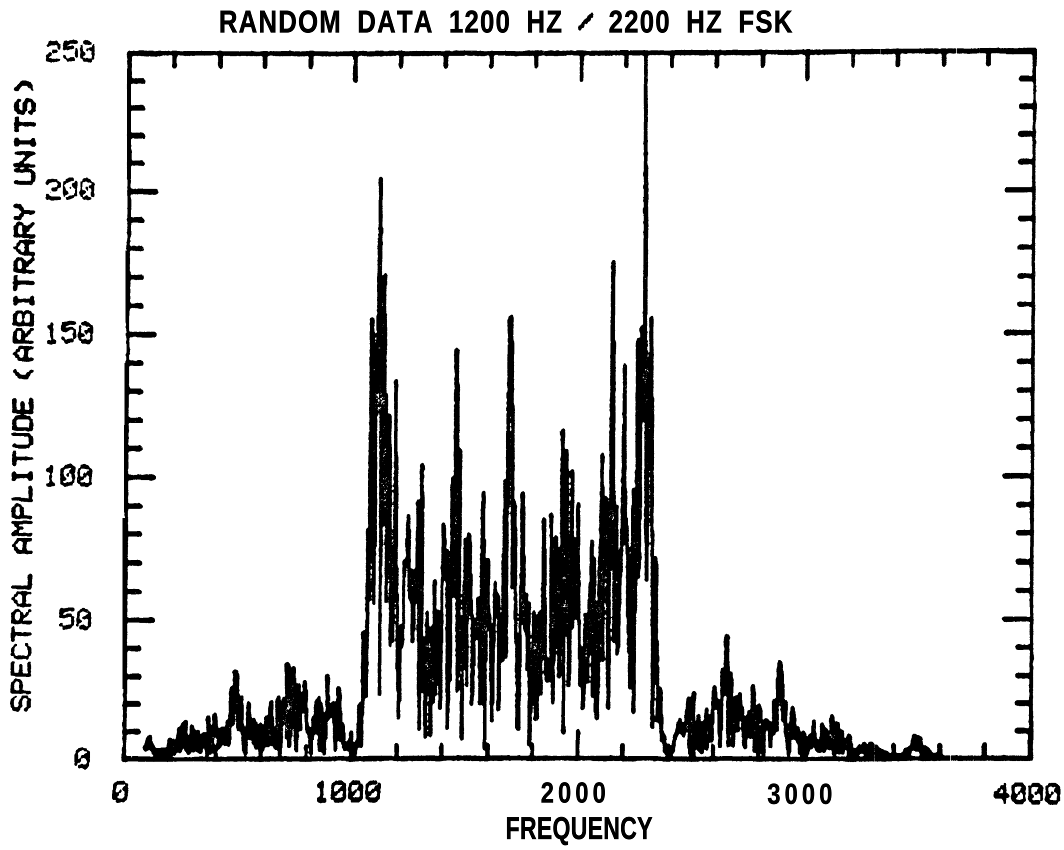

Figure 1. Typical spectral characteristics of random Bell 202 compatible phase-continuous FSK data.

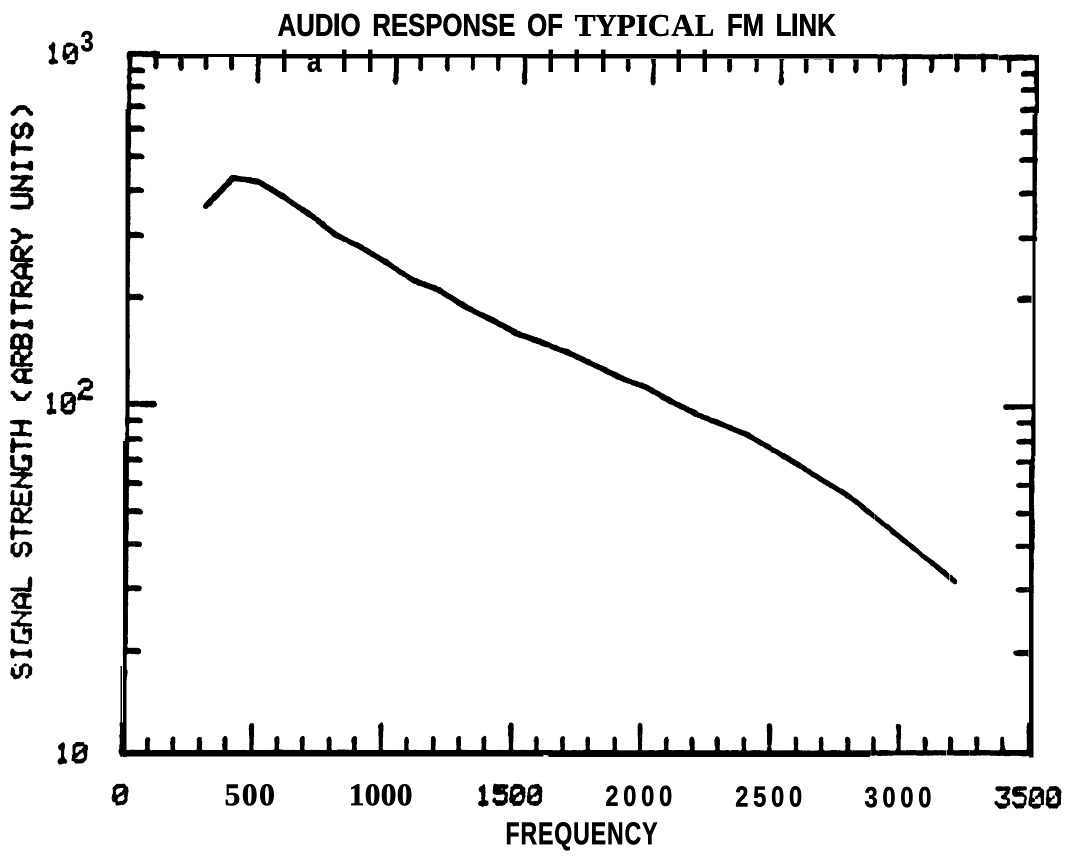

Figure 2. Audio response of a typical 2-meter FM link.

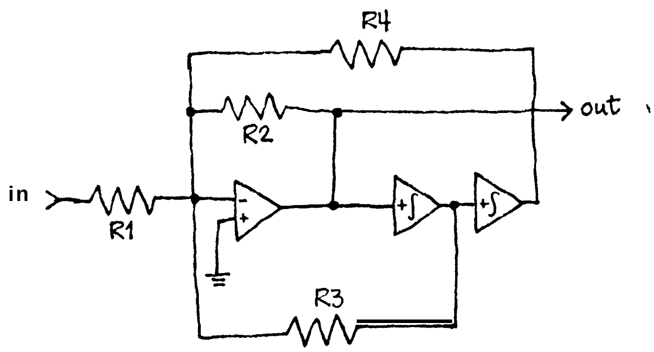

Figure 3a. Equivalent circuit for the input highpass filter section.

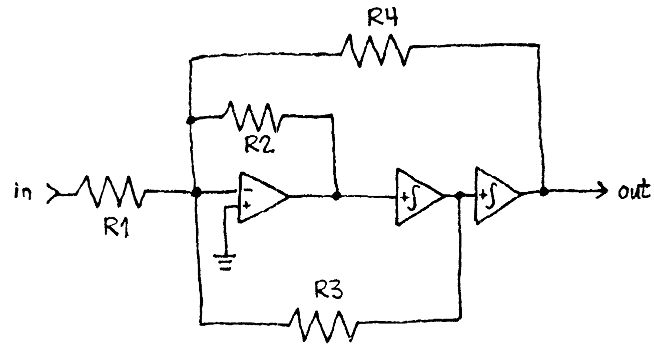

Figure 3b. Equivalent circuit for the output lowpass filter section.

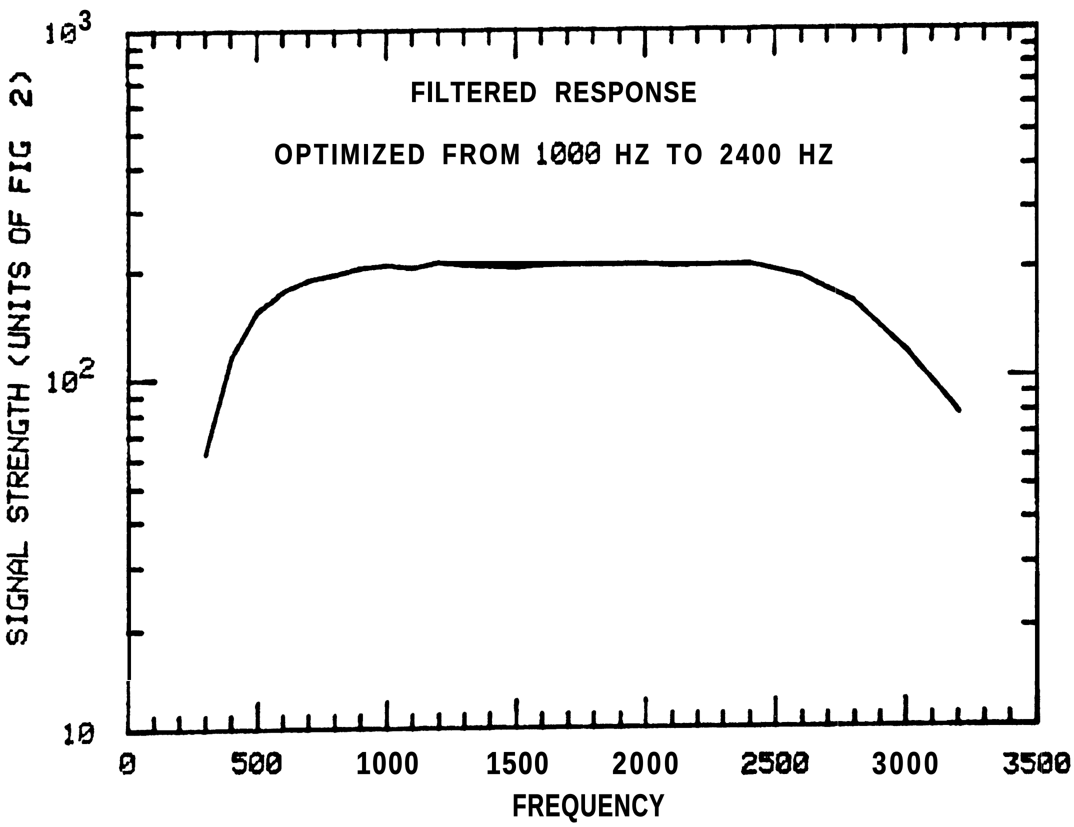

Figure 4. Filtered audio response of Fig. 2

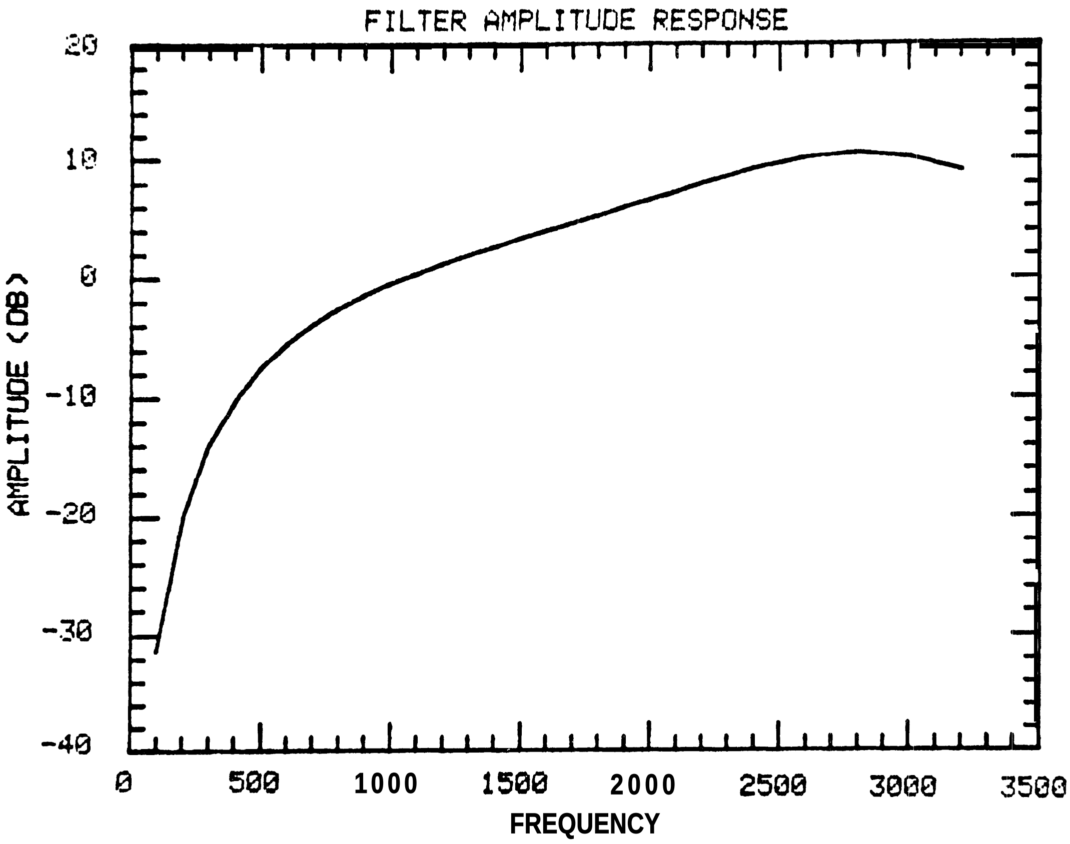

Figure 5a. Filter amplitude response.

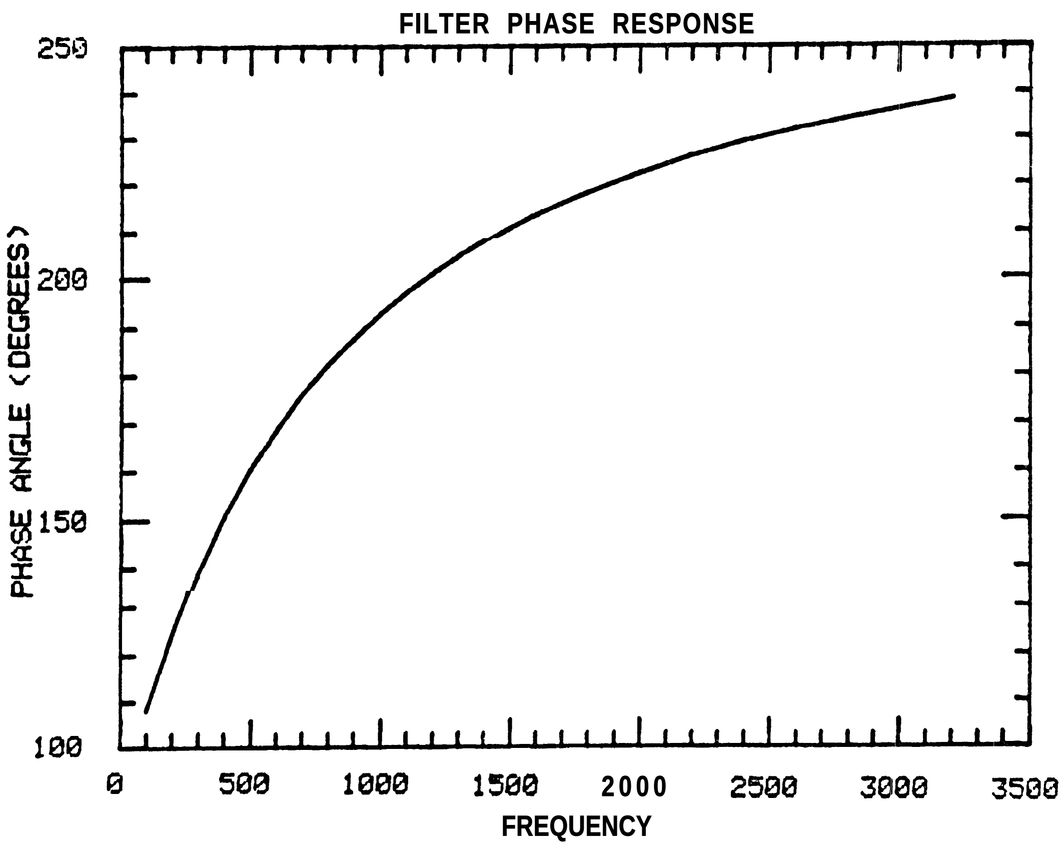

Figure 5b. Filter phase response.