SUPPLEMENT
to
USER'S MANUAL

SECTION 1 Synopsis of Added Features

Several additions and modifications have been incorporated into this release of DISPRO®.

- Main menu additions
  1. Under *Time-Response* the first item on the pull-down menu is *Compute Impulse/Step/Sinusoidal Response*. This indicates that you can now compute the time response of a filter to a sinusoidal input. In addition, in the TIMERESP module you may also plot the behavior of IIR filters, as well as display numeric values. This added feature simplifies the tone testing of filters, by not requiring that you generate a test signal, store it in a file, compute and save the time response, and then use the time plot module. The plotting feature is available only for simulation of the filter using floating-point arithmetic; the idea is to be able to see the degree to which any variable overflows. In fixed-point arithmetic simulations the values of variables are either clipped at the maximum values, or allowed to wrap around as happens in 2's-complement arithmetic. Consequently, the only way to see the extent of overflows is to use floating-point arithmetic simulations. (Note that this sinusoidal response feature is probably not too exciting for FIR filters, but it is provided anyway—although plotting is not provided.).
  2. There is a new item, *Config*, which allows you to change the directory path for filter data files, and also to specify the printer (primarily needed for dumping graphics from the screen). Any changes you make are saved to the DISPRO20.CFG file, and will be loaded as the defaults in the next session of DISPRO. If you find that a graphics screen dump is not coming out properly on your printer it may be due to having the wrong type of printer selected as the default. Remember that DISPRO supports graphics output only to dot-matrix printers which are compatible with the IBM/Epson codes, and to HP laser and inkjet printers (HP LaserJet & DeskJet), or any PCL-compatible printer.

- Improvements to IIR Coefficient Scaling

  In DISPRO there is implemented a coefficient scaling technique for IIR filters which attempts to establish values for the A, or gain, coefficients for each biquad and first-order section so that the frequency response from the input of the filter to the output of each section is bounded by 0 dB. When the filter is implemented with the merged-biquad topology then there will be no overflows, in steady-state, for a unit-amplitude sinusoidal input. Actually, this condition of 0-dB-scaling is achievable for almost all designs but there are situations in which it may not be attained. As an aid there has been added the option of scaling to a level of other than 0 dB. For example, if 0-dB-scaling cannot be achieved for any choice of section sequencing (you can arrange the sections with largest magnitude poles first, smallest magnitude first, or in any arbitrary order) you could elect to scale, for example, to a peak of 6 or 12 dB and precede the filter with a gain of 0.5 or 0.25.

The procedure used to determine the peaks of the frequency responses has been significantly refined. The location of peaks is now done with a resolution of better than 1 part in 50,000, but without actually computing that many samples of the frequency response. Great advantage is taken of the properties of biquad transfer functions.

- Additions to Frequency Response Computation and Plotting

  In support of the scaling of IIR filters there has been added the capability of computing and plotting the frequency response of any consecutive set of sections in a filter. Thus, you can investigate the response of any single biquad or first-order section, or the cascade of two or more sections. The only restriction is that the sections must be taken in sequence. I.e., you must analyze sections 3 through 6, for example, as a group; you cannot analyze just section 3 cascaded with just section 6, but must consider sections 3, 4, 5, and 6 as a cascaded group.

- Miscellaneous

  When values are requested for various parameters and options there is usually an entry region defined by a background color of red. The default value, or the last input value, appears in this input region in reverse video—black characters on a white background. If you press any valid key, such as a number or letter, the value in reverse video will be deleted, to be replaced by your entry. If you wish to keep most of the characters in the reverse video field, intending to modify only one or a few, then you should strike either the right arrow, —►, or left arrow, ◄—, cursor key. The reverse video field will revert to the normal entry field colors of yellow on red and you can use the standard editing keys to modify the value.

SECTION 2 Illustration of the New Features in Operation

This example shows the use of the additional features in the Frequency and Time Response modules. The filter being analyzed is a lowpass, seventh-order Elliptic, with the specifications:

FSAMP = 10 KHz, AMAX = -0.5, AMIN = 60, FP = 200 Hz, FS = 243.8 Hz

2.1 Coefficient Scaling

This is the popup box that requests a value for the peak to be scaled to, with the default being 0 dB.

```
03-05-2005                        DISPRO(R)                          14:20:08
 ─────────────────────| ELLIPTIC DIGITAL FILTER DESIGN |─────────────────────
 1        AUTOMATIC - Sequence, Scale, and Store in Filter Data File (FDF)
 2          SPECIFY - Biquad Sequence & Scaling (before storage in FDF)
 3    PRINT/DISPLAY - Specifications and Coefficients (floating-point)
 4       The peak of the response of each partial cascade can be
                  scaled to any value, usually <= 0 dB.
 5   COMP
         >> Desired value for scaled peaks (dB) ? 0
 6                                                          re in FDF)
 7           CREATE - Another Filter Data File (FDF)
 8           MODIFY - Change specifications
 ───────────────────────────────────────(Copyright(C) John O'Donnell 2005)
  [Type Esc to EXIT].....Choice? ....
 Use cursor keys or type menu item number. [Strike ENTER key to make selection]
```

If scaling to a level of 0 dB cannot be accomplished for any sequencing of the sections of the IIR filter then you should choose a value greater than 0 dB. Alternatively, you can choose a value less than 0 dB if you expect that overflow will occur, even with a merged-biquad implementation. (Remember, the combination of scaling for 0 dB peaks and using the merged-biquad structure guarantees freedom from overflow only in steady state for a single tone input. Most signals will be more complicated, and the nonlinear phase of the IIR filter can cause a "pileup" which results in signal peaks greater than 0 dB.)

The frequencies of the peaks, and the transmission values are displayed for your evaluations. If satisfactory then you can proceed to store the coefficient set in a filter data file, or try another scaling by *Esc*aping back to the initial menu screen. Note that the peak frequencies are not for the individual sections, but are for the cascade of all of the sections to that point in the filter.

```
These are the maximum transmissions from the input of
the filter to the output of each section in the cascade.
It is desirable that each of these transmissions be <= 0 dB,
to reduce overflow problems in fixed-point arithmetic.

 Section#   Frequency at Peak(Hz)   Peak Transmission (dB)
    1            200.2000                  -0.0000
    2            198.6000                   0.0000
    3            197.6000                   0.0000
    4            0.000000                  -0.0001
```

2.2 Frequency Response of Selected Sections—Partial Responses

```
03-05-2005                     DISPRO(R)                        14:20:53

            FREQUENCY RESPONSE COMPUTATION & PLOTTING
 1      AUTOMATIC PLOTTING - using floating-point coefficients & dB scale
 2   QUANTIZE COEFFICIENTS - for response computation
 3       COMPUTE RESPONSE - for plotting (optional partial response for IIR)
 4  PLOT COMPUTED RESPONSE - for any frequency and amplitude range
 5     PRINT/DISPLAY/STORE - numerical frequency response values
 6      SAVE RESPONSE DATA - to filter data file
 7           STATUS REPORT - filter specs & current computed response data
                                        (Copyright(C) John O'Donnell 2005)
 [Type Esc to EXIT].....Choice? .... _

Use cursor keys or type menu item number. [Strike ENTER key to make selection]
```

This screen shows the addition to menu item 3. In order to compute a partial response you will have to specify all of the details, and consequently cannot use the AUTOMATIC choice which plots the full filter response.

```
                  FREQUENCY RESPONSE COMPUTATION
              Coefficient wordlength: floating point

              Lowest frequency for plot (Hz) 0.00000
             Highest frequency for plot (Hz) 500.000
      Number of frequency values in plot (1000 max) 500.000

                    COMPUTE PARTIAL RESPONSE ?
                               NO
                               YES
                             ↑ ↓ ↵

      There are 4 sections.  The partial response can be computed for
      any set of consecutive sections, or for a single section.

            Begin at section 1
              End at section 1

Type new values. ↓ or ENTER for no change. ESC to exit. ↑ to redo parameters.
```

After you have specified the frequency range and the number of points, you elect to compute the full response or a partial response. The partial response can be for any single section in the cascade, or for any sequential group of filters in the cascade. Here is illustrated the dialog associated with computing the response of only section 1 of the filter.
A similar dialog can be conducted for each of the four sections of the filter. The individual frequency responses are shown on page S-7.

```
                    FREQUENCY RESPONSE COMPUTATION

               Coefficient wordlength: floating point

               Lowest frequency for plot (Hz) 0.00000
              Highest frequency for plot (Hz) 500.000
     Number of frequency values in plot (1000 max) 500.000

                   COMPUTE PARTIAL RESPONSE ?
                              NO
                              YES
                            ↑ ↓ ←┘

    There are 4 sections.  The partial response can be computed for
    any set of consecutive sections, or for a single section.

          Begin at section 1
            End at section 2

 Type new values. ↓ or ENTER for no change. ESC to exit. ↑ to redo parameters.
```

This shows the dialog relating to computing and plotting the response of the first two sections of the filter in cascade. Similar dialogs produce the response for any set of sections in cascade.

The responses of the first two and the first three cascaded sections are shown on page S-7.

2.3 Computation and Plotting of Time-Domain Sinusoidal Response

The plotting capability that has been added to the TIMERESP module can be used for an impulse response, a step response, a sinusoidal response, and for a forced response (test signal data in an .EXC file, usually created by the *Generate Test Signal* menu item). In all cases the plotting is available only for floating-point arithmetic simulations; for fixed-point arithmetic simulation the only plotting capability is for the filter output using the TIMEPLOT module (data values generated in TIMERESP must be saved to the .FDF file in order for TIMEPLOT to plot them.)

```
Help  Design  Coefficients  Frequency-Response  Time-Response  Spectrum  Config
                                  Compute Impulse/Step/Sinusoidal Response
                                  Compute Forced Time Response
                                  Generate Test Signal
                                  Plot Time Samples

       Version 2.0 Copyright(C) Dr. John O'Donnell 1983 to 2005
            BRT71EFR.EXE Copyright(C) Microsoft Corp. 1990
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ESC to exit ...use 1st character or ←↑↓- keys for menu choice... ENTER to select
```

This illustration is of the sinusoidal response capabilities so you should select the first item on the **Time-Response** pull-down menu.

2.3.1 *Merged-Biquad Topology*

```
                IMPULSE/STEP/SINUSOIDAL RESPONSE OF FILTER
                  FILE C:\FILTER\DATA\SOURCE\ELLIP07A.FDF

▸ Status of parameters
        Topology: Merged biquad
    Coefficients: (none read from file yet).
      Arithmetic: floating point
Overflow action: Ignore
▸  Breakpoint settings   (none)

                          SELECT ACTION
      COEFFICIENTS -- COEFFICIENT WORDLENGTH
        ARITHMETIC --   Floating Point       arameters
          OVERFLOW --   Fixed Point
       BREAKPOINTS --                         breakpoints
               NEW --       ↑ ↓ ←┘ Esc        filter response
            RESUME -- Continue current response computation
 SAVE/PRINT/DISPLAY -- Response sample values in buffer
    IMPLEMENTATION -- View / print scaled coefficients
                          ↑ ↓ ←┘ Esc

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```

The coefficient scaling procedure is designed to prevent overflow in the merged-biquad form of filter implementation. This is the first case that will be investigated.
The graphical outputs are on page S-8.

The first step is simply to read in the coefficients as floating-point values.

```
            IMPULSE/STEP/SINUSOIDAL RESPONSE OF FILTER
              FILE C:\FILTER\DATA\SOURCE\ELLIP07A.FDF

▸ Status of parameters
     Topology: Merged biquad
   Coefficients: floating point...
     Arithmetic: floating point
Overflow action: Ignore
▸  Breakpoint settings  (none)

                        SELECT ACTION
       COEFFICIENTS -- Specify wordlength
          ARITHME ┌SELECT COMPUTATIONAL WORDLENGTH┐meters
             OVERF│   Floating Point (32 bits)    │
          BREAKPOI│ Fixed Point (2 to 23 Bits+Sign)│eakpoints
                  │                                │ter response
              RES └──────CHOOSE TOPOLOGY───────────┘mputation
     SAVE/PRINT/DISPLAY - │Merged Biquads    │ in buffer
         IMPLEMENTATION - │Canonic Form Biquads│efficients
                                ↑ ↓ ◄┘

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```

Select **ARITHMETIC** and then select **Merged-Biquads**.

```
                     CHOOSE RESPONSE TYPE
      IMPULSE -- Input is single non-zero sample at time 0
         STEP -- Input is a constant value at each sample time
   SINUSOIDAL -- Input is a sampled sine wave
                          ↑ ↓ ◄┘ Esc

                  Sinusoidal Response Computation

   Samples of a sine wave with peak=1.0 will be input at each sample time.
        It may be necessary to reduce this value to avoid overflow.
      For fixed-point arithmetic the decimal value specified will be
 converted to an integer value consistent with the computational wordlength.

   ** Value for input level (1.0 is default and maximum) 1.0

   ** Frequency of sine wave (Hz) < 5000   150

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```

Select the **NEW** item on the main menu, and then select **SINUSOIDAL** on the pop-up menu. After that you will have to set the amplitude and frequency of the sine wave. Here an amplitude of 1.0 and a frequency of 150 Hz have been specified (remember, this example filter is lowpass with a passband edge of 200 Hz.)

```
                     CHOOSE RESPONSE TYPE
      IMPULSE -- Input is single non-zero sample at time 0
         STEP -- Input is a constant value at each sample time
   SINUSOIDAL -- Input is a sampled sine wave
                          ↑ ↓ ◄┘ Esc

                  Sinusoidal Response Computation

   Samples of a sine wave with peak=1.0 will be input at each sample time.
        It may be necessary to reduce this value to avoid overflow.
      For fixed-point arithmetic the decimal value specified will be
 converted to an integer value consistent with the computational wordlength.

                        SELECT DISPLAY TYPE
    NUMERIC -- Display values of input and output
 GRAPHICAL -- Plot input, output, and state variables (EGA/VGA only)
                          ↑ ↓ ◄┘ Esc

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```

Next, elect the **GRAPHICAL** display option.

```
                     CHOOSE RESPONSE TYPE
      IMPULSE -- Input is single non-zero sample at time 0
         STEP -- Input is a constant value at each sample time
   SINUSOIDAL -- Input is a sampled sine wave
                          ↑ ↓ ◄┘ Esc

                  Sinusoidal Response Computation

   Samples of a sine wave with peak=1.0 will be input at each sample time.
        It may be necessary to reduce this value to avoid overflow.
      For fixed-point arithmetic the decimal value specified will be
 converted to an integer value consistent with the computational wordlength.

                        Plotting Parameters
          Data will be plotted as successive blocks of samples.
   You specify the block size (number of samples on screen at a time) and
    the number of blocks.  The total number of samples is limited to 8E6.

   ** Number of samples in each block 100
   ** Number of blocks 10
   ** Top of scale (use 1.1 for Merged-Biquads, 20 to 100 for Canonic 1.1

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```

You may have to experiment with the plotting parameters in order to obtain a result that best shows the filter behavior. Here it is specified that each screenful will hold 100 samples (at 10 Khz sample rate this is 1.0 millisecond of data), and there will be 10 such screensful. Note that you should set this last parameter to a reasonably high value in order not to have to repeat the plotting process from the beginning. You can stop the plotting process any time by striking the X key. You may, however, have to try more than one value for the top of the amplitude scale.

2.3.2 *Canonic Form Topology*

```
                IMPULSE/STEP/SINUSOIDAL RESPONSE OF FILTER
                  FILE C:\FILTER\DATA\SOURCE\ELLIP07A.FDF


▸ Status of parameters
       Topology: Merged biquad
   Coefficients: floating point...
     Arithmetic: floating point
Overflow action: Ignore
▸  Breakpoint settings  (none)
                          SELECT ACTION
        COEFFICIENTS -- Specify wordlength
          ARITHME ─SELECT COMPUTATIONAL WORDLENGTH─ meters
            OVERF      Floating Point (32 bits)
         BREAKPOI   Fixed Point (2 to 23 Bits+Sign)   eakpoints
                                                      ter response
              RES  ───────CHOOSE TOPOLOGY───────      mputation
   SAVE/PRINT/DISPLAY -      Merged Biquads       in buffer
       IMPLEMENTATION -  ::Canonic Form Biquads::  efficients
                                 ↑ ↓ ◄─┘

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```

The graphical outputs, one each for the linear and logarithmic amplitude scale choices, are on page S-8.

```
         IMPLEMENTATION OF THE 'A' COEFFICIENT FOR CANONIC FORM BIQUADS

.....FIRST FORM:
-  Input to biquad scaled by A1 < 1 ... provides some control of overflow.
-  Factor A as A = A1 * A2, with A2 = ( 1.677721E+07 / 1.677722E+07 )
   (This keeps |A2*D| < 2, and |A2*E| < 1)

          A2 + A2*D z^-1 + A2*E z^-2
      A1 ---------------------------
              1 + B z^-1 + C z^-2

.....SECOND FORM:
-  No scaling of input to biquad...faster execution, but no overflow control.
   ( A1 = 1, A2 = A )

          A + A*D z^-1 + A*E z^-2                ─'A' COEFFICIENT CHOICE─
          -----------------------                ::::::::::First Form::::::::::
            1 + B z^-1 + C z^-2                         Second Form

                                                      ↑ ↓ ◄─┘ Esc
```

This is the standard screen for choosing how to handle the A coefficient. The choice makes a small difference in execution time on most DSP microprocessors.

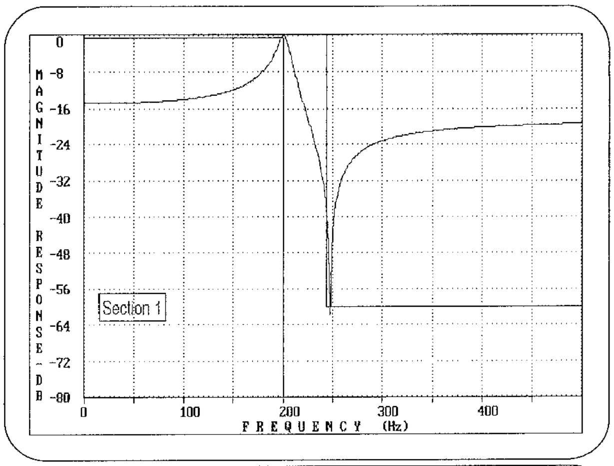
```
         IMPLEMENTATION OF THE 'A' COEFFICIENT FOR CANONIC FORM BIQUADS

.....FIRST FORM:
-  Input to biquad scaled by A1 < 1 ... provides some control of overflow.
-  Factor A as A = A1 * A2, with A2 = ( 1.677721E+07 / 1.677722E+07 )
   (This keeps |A2*D| < 2, and |A2*E| < 1)

          A2 + A2*D z^-1 + A2*E z^-2
      A1 ---------------------------
              1 + B z^-1 + C z^-2

                                 Plotting Parameters
           Data will b ── VERTICAL SCALE CHOICE ──  of samples.
     You specify the bloc ::::::::::::Linear::::::::::::  creen at a time) and
      the number of block  Log of absolute value (dB)  s is limited to 8E6.

     ** Number of samples ─────── ↑ ↓ ◄─┘ Esc ───────
     ** Number of blocks 10
```

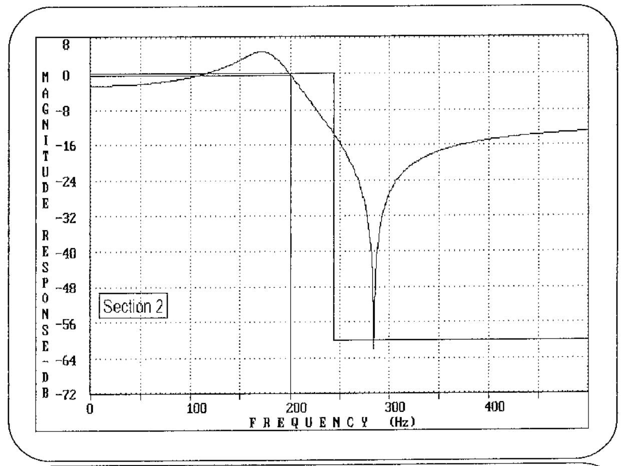
The option of a logarithmic amplitude scale is made available only for the canonic form because of the (usually) large swings of the internal state variables. When a linear scale is chosen, solid horizontal lines are drawn at ±1 (in Brown). In the plot on page S-8 you can see that the input signal and the outputs of the biquads are practically indecipherable, but the state variables—the dashed lines—cover almost the full vertical range.

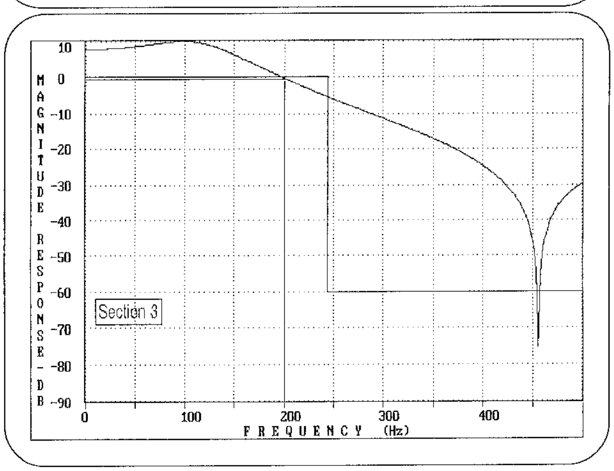
```
         IMPLEMENTATION OF THE 'A' COEFFICIENT FOR CANONIC FORM BIQUADS

.....FIRST FORM:
-  Input to biquad scaled by A1 < 1 ... provides some control of overflow.
-  Factor A as A = A1 * A2, with A2 = ( 1.677721E+07 / 1.677722E+07 )
   (This keeps |A2*D| < 2, and |A2*E| < 1)

          A2 + A2*D z^-1 + A2*E z^-2
      A1 ---------------------------
              1 + B z^-1 + C z^-2

                                 Plotting Parameters
           Data will b ── VERTICAL SCALE CHOICE ──  of samples.
     You specify the bloc          Linear             creen at a time) and
      the number of block ::Log of absolute value (dB)::  s is limited to 8E6.

     ** Number of samples ─────── ↑ ↓ ◄─┘ Esc ───────
     ** Number of blocks 10
```

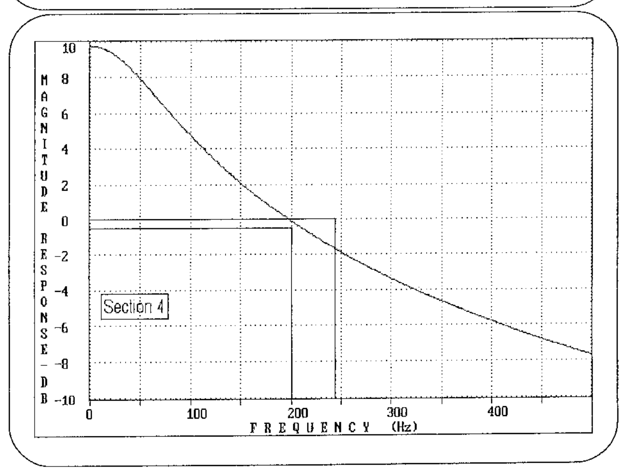
On the logarithmic scale (what is plotted is $20 * \log_{10}(|variable|)$ in dB) the solid line is at 0 dB. All of the section outputs, as well as the input, are seen to be below this 0 dB level. The state variables for the first three sections are well above 0 dB. The state variable for the last section is less than the 0 dB level because this last section is a first-order section. It's the complex-conjugate poles of the biquads which give rise to the large dynamic range.

These plots show the frequency responses of each of the individual sections of the seventh-order Elliptic filter. Each of the three biquad sections clearly has a pole and zero. The first-order section is seen to

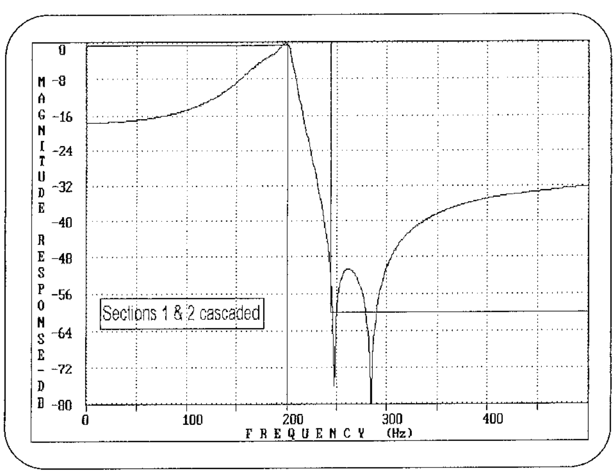


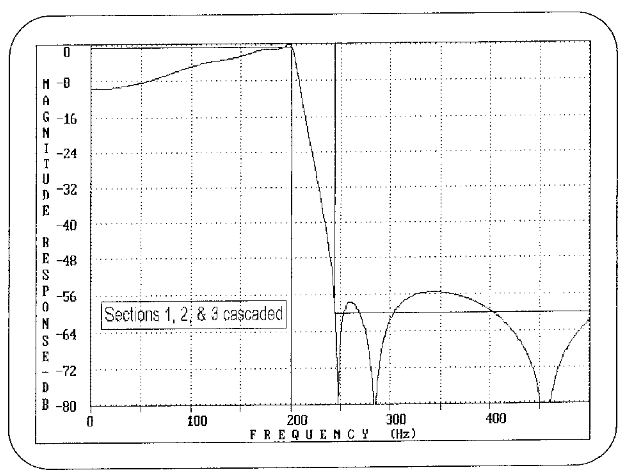


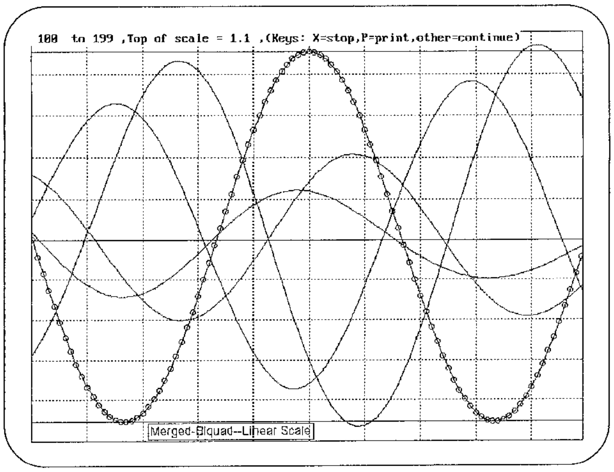




provide the final degree of passband and stopband shaping. Even though the frequency response of an individual section exhibits peaks greater than 0 dB, the cascade of sections will not. By computing the partial response for section 1 through 2, and 1 through 3, you can verify this fact. The response of sections 1 through 4 cascaded is just the filter frequency response, which is, of course, bounded by 0 dB.







The time-domain response for the 150 Hz sinusoidal input is plotted here for the merged-biquad topology. The input sinusoid samples are circled (remember, this is *not* a continuous-time waveform). This black-and-white reproduction is a little difficult to decipher; on the screen a simple color coding is employed which allows you to identify each of the waveforms. The color coding begins with bright white—color attribute 15—for the filter output, and proceeds down the color attribute scale to 0 , for black. In order to avoid repetitions of colors, the IIR filter order for which this plotting can be done is restricted to 30—a maximum of 15 sections. As with all of these plots the $\pm$ 1.0 levels are indicated by solid (brown) lines. Note that on this plot the outputs of all sections exhibit no overflow in steady state. There often are, however, several samples of overflow in the transient startup.

Color	Color Attribute Number	Filter Section Number
Bright White	15	N
Yellow (= Intense Brown)	14	N-1
Bright Magenta	13	N-2
Bright Red	12	N-3
Bright Cyan	11	N-4
Bright Green	10	N-5
Bright Blue	9	N-6
Gray	8	N-7
White	7	N-8
Brown	6	N-9
Magenta	5	N-10
Red	4	N-11
Cyan	3	N-12
Green	2	N-13
Blue	1	N-14
Black	0	N-15

This is the table of color assignments used in the plotting of the time-domain sinusoidal response. For the merged-biquad toploogy only solid-line curves are plotted. For the canonic form topology the solid curves are also the section outputs, but there are dashed-line curves for the internal, or state variables. For biquads these state variables are affected by the Q of the pole.

These next plots are for the canonic form, using a linear and a log-of-absolute-magnitude (dB) scale. As can be seen, there is serious overflow of the internal=state variables, due to the resonator dynamics.