SECTION 6

DISPRO OPERATION REFERENCE

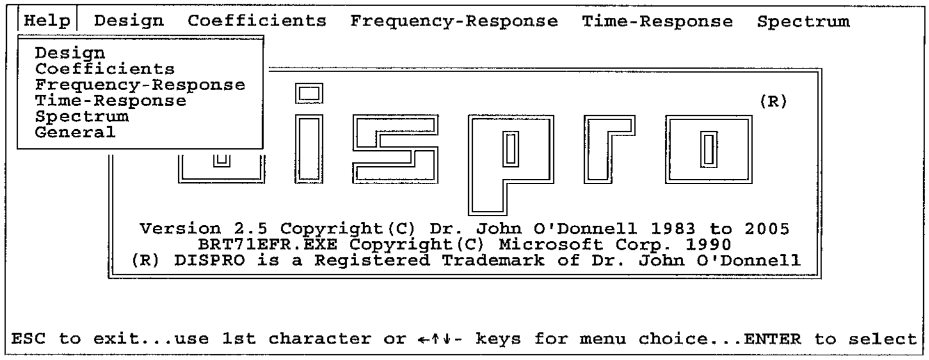
This section is organized in accordance with the top menu line selections which are accessed at the DISPRO command level. Each selection on the top menu line constitutes a logical category of DISPRO operations, and is associated with a "pull-down" menu containing all of the options in the selected category. You select a category and get the pull-down menu by using the left and right arrow keys on the cursor keypad to move along the main menu line.

Help	Provides access to text files with rudimentary information on each of the top menu line categories.
Design	Each of the six types of digital filters designed by DISPRO is accessed through this pull-down menu.
Coefficients	Possibilities are quantization of coefficients, export of coefficients to disk files in formats acceptable to DSP chip assemblers, and SHELLing to DOS or external program.
Frequency-Response	Only one selection: a unified module for computing the frequency response for any wordlength, plotting it, and printing the values.
Time-Response	Computation in a wide range of arithmetic characteristics of the impulse response or forced time response of a filter.
Spectrum	Fourier analysis *via* the FFT for periodic and non-periodic time functions.

The process of starting DISPRO so that you get to the main menu screen and are ready to make a selection is described in detail in Sect. 1.2. One point that should be made is that when you select a filter design option you will find that direct paths are available from the design module to the frequency-response and coefficient quantization modules. These operations are also made available from the main menu so that you can process a filter design that was done in a previous session of DISPRO.

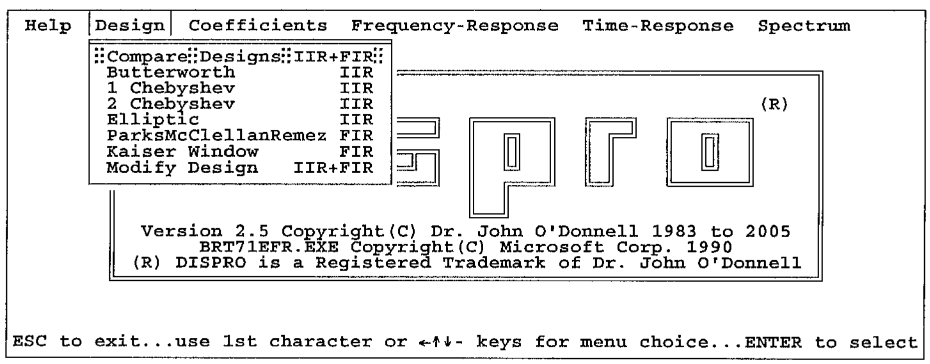
While you are at the main menu level and choose, for example, to compute the frequency response for a filter that was designed in a previous use of DISPRO, you may find that you are unsure of the filter data file name—you may have ELLIP06A.FDF and ELLIP06B.FDF and can't remember which one you want to work with. Simply select one of the files and, while in the frequency response module, use the *Status* selection to check on the filter characteristics. Alternatively, select *Coefficients* | *Quantize* and display the filter specifications and coefficients for any wordlength from 2 to 23 bits+sign — use 24 for floating-point coefficients.

6.1 Help



Each of the help selections provides a minimum description of the resources available under each of the top menu line headings. These help files are intended only as a readily available source of information and are not a substitute for reading the manual.

6.2 DESIGN



Under this main heading are found the eight filter design selections—the heart of DISPRO's operations. There is an identical graphical interface for the IIR and the PMR FIR designs—the Kaiser design has a different, non-graphical interface. The nature of the interface will be described with the aid of the screen images in Fig. 6-1. On an EGA/VGA display this screen will be in color; on a CGA screen it will be in black and white. [*NOTE* In order to get the solid block indicating the typing zone on a CGA screen you will have to run the DOS program GRAFTABL.COM before starting DISPRO.] All corrections/edits are made with the backspace key.

Some of the stages in the process of specifying a bandpass filter are shown in Fig. 6-1. The initial screen is in Fig. 6-1(a) where you choose LP, HP, BP or BS filter type. Having typed *BP* to design a bandpass filter the screen now shows the bandpass filter specification boundaries in a schematic form. You now choose the frequency units and specify the sampling frequency, in the chosen units. Fig. 6-1(a) shows a choice of Hz and

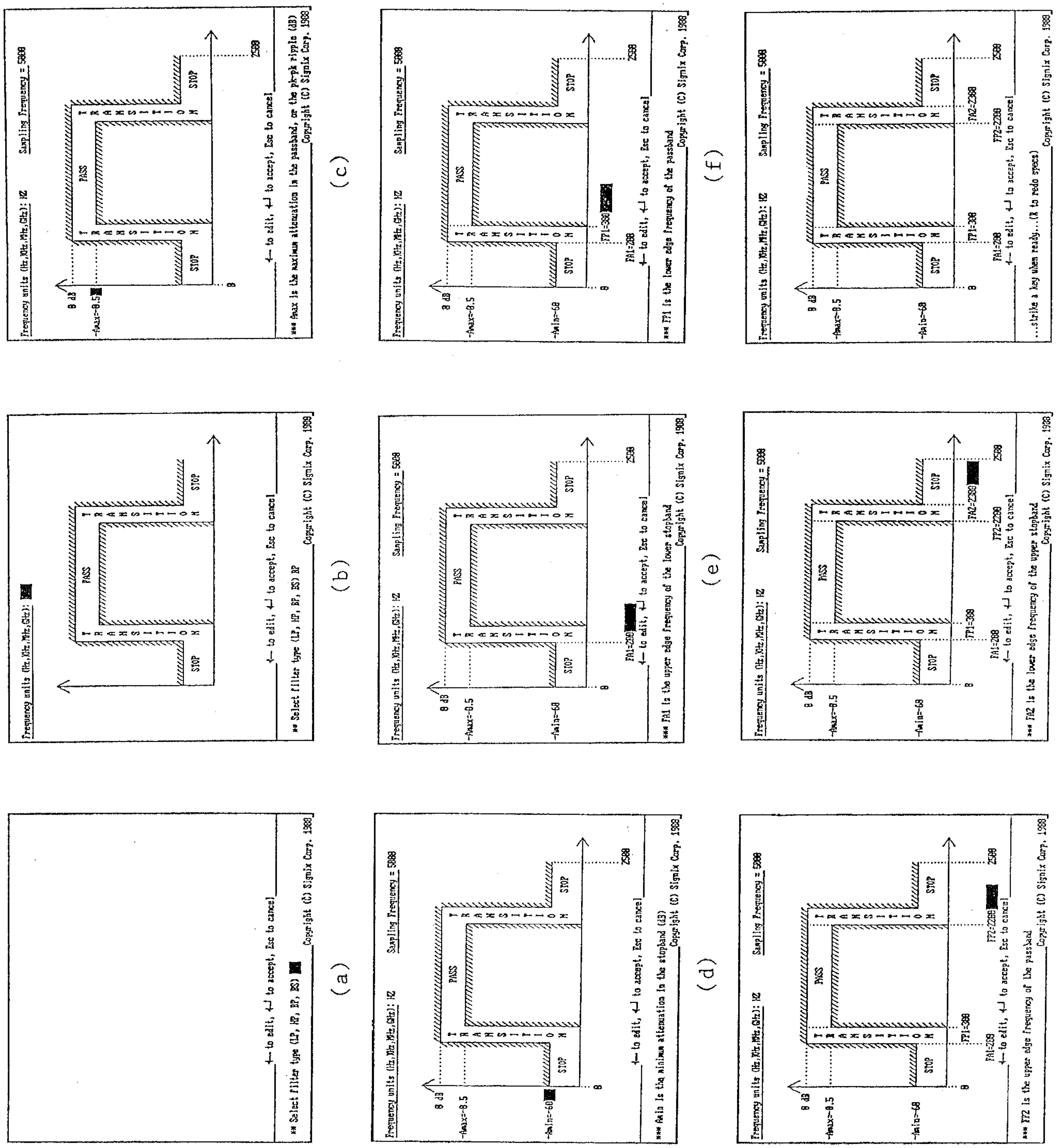


FIGURE 6-1 Procedure for Specifying a Filter Response: Bandpass Example

a sampling frequency of 8000 Hz. (When typing frequency values you may use power-of-10 notation anywhere in DISPRO; thus, in stead of typing 8000 you could have typed 8e3.)

The passband ripple—or passband edge attenuation for Butterworth and Chebyshev I filters—and stopband attenuation have default values of 0.5 dB and 60 dB respectively. To modify these values use the backspace key to clear the values and type new ones. Because the frequency response model on the screen is for the transmission characteristic of the filter, both values must be negative. Input values are checked for consistency; errors are flagged by the message **BAD**, which requires striking the *Enter* key to clear, and the typing of a valid value before the parameter is accepted. The largest value allowed for AMIN is 1000 dB—certainly not a major limitation! Fig. 6-1(c) shows that we have accepted the default values.

When the sampling frequency was specified, the limit for the frequency axis was set at one-half the sampling frequency; the value appears in screens (b) and (c). As you enter a value for the edge of a band it is checked to ensure that it is neither ≤ the previously entered band edge nor ≥ one-half the sampling frequency. After the last necessary band edge frequency value is input, and a complete set of specifications is available, the table of orders/length appears on the screen—see Fig. 6-1(d). You have the option to use the cursor keys to move to any specification value and change it, or to strike the *Enter* key to continue with the design process. There are additional points later in the design process at which you will be allowed to enter new specification values, although you can still always return to this graphical input screen.

When you strike the *Enter* key twice you are presented with the following menu

Select filter category, or strike Esc key to change specifications

The transition ratios (IIR) or transition band widths (FIR) are unequal. They can be easily adjusted in the selected design module. The filter order/length values shown below are for the smaller transition ratio, or band width, value.

----- FILTER CATEGORY -----	Order/Length:	Nominal	Min	Max
BUTTERWORTH	IIR	93.36	92	94
CHEBYSHEV 1	IIR	28.81	28	30
CHEBYSHEV 2	IIR	28.81	28	30
ELLIPTIC	IIR	14.58	14	16
PARKS-McCLELLAN-REMEZ	FIR (approximate)	176.35	176	177

↑ ↓ ↵ Esc

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which allows you to select from the IIR and PMR FIR filter categories. Let's now trace the steps after selecting the Elliptic IIR category—the process is the same for the Butterworth and Chebyshev, and is similar for the Parks-McClellan-Remez equiripple FIR.

6.2.1 IIR Filters—Completing the Design

After the initial specifications are entered you will be involved in an interactive process which has the goal of finding an integer-valued filter order which satisfies your specifications. You will have the opportunity to improve the filter performance if you choose a higher filter order, or you may decide to relax some performance specification in order to obtain a lower filter order. After developing an acceptable set of specifications and filter order you will be transferred to another module for the purpose of completing the design by establishing a sequence for the biquad sections, optionally scaling for 0 dB transmissions, and creating a filter data file which will hold the specifications and floating-point precision coefficients. The sequencing and scaling steps are intended to guard against overflow in fixed-point arithmetic and thus can be ignored for floating-point realizations.

We now transition to the IIRDES module where the actual filter length and the final filter specifications can be established. The next three screen images show our choices for the band edge adjustment to satisfy the geometric symmetry constraint, for the filter length, and again for adjustment of the stop band edges to achieve compatibility with the order we have chosen for the filter.

```
             PARAMETER SPECIFICATIONS FOR ELLIPTIC BANDPASS FILTER

                  Sampling Frequency (Hz)     8.00000E+03

               End of lower stopband (Hz)     200.000
               Beginning of passband (Hz)     300.000
                     End of passband (Hz)     3.30000E+03
         Beginning of upper stopband (Hz)     3.40000E+03
        Maximum passband attenuation (dB)     0.500000
        Minimum stopband attenuation (dB)     60.0000

       Stopband edges must be adjusted to obtain equal transition ratios

>  Choose one of the following adjustments ... (Esc to redo specifications)

    1.        End of lower stopband (Hz) = 255.702        (Order =  14.6)

    2.  Beginning of upper stopband (Hz) = 3.52793E+03    (Order =  11.1)

    3.        End of lower stopband (Hz) = 227.878        (Order =  12.6)
        Beginning of upper stopband (Hz) = 3.46362E+03

...................Choice (1,2,3, or Esc) ? 3
```

```
             PARAMETER SPECIFICATIONS FOR ELLIPTIC BANDPASS FILTER

                  Sampling Frequency (Hz)     8.00000E+03

               End of lower stopband (Hz)     227.878
               Beginning of passband (Hz)     300.000
                     End of passband (Hz)     3.30000E+03
         Beginning of upper stopband (Hz)     3.46362E+03
        Maximum passband attenuation (dB)     0.500000
        Minimum stopband attenuation (dB)     60.0000


                     The computed filter order is 12.552
         You must specify an EVEN integer value for the filter order.
          For a smaller order the filter performance must be reduced;
          for a greater order the filter performance can be improved.

[Max order is 99 ].... Choice for filter order ? 14

                         [Strike Esc to change specs]

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```

```
             PARAMETER SPECIFICATIONS FOR ELLIPTIC BANDPASS FILTER

                  Sampling Frequency (Hz)     8.00000E+03

               End of lower stopband (Hz)     227.878
               Beginning of passband (Hz)     300.000
                     End of passband (Hz)     3.30000E+03
         Beginning of upper stopband (Hz)     3.46362E+03
        Maximum passband attenuation (dB)     0.500000
        Minimum stopband attenuation (dB)     60.0000

                   > You have chosen a filter of order  14

         ------------SELECT PARAMETER ADJUSTMENT OPTION-------------
                    Increase stopband attenuation
                       Decrease passband ripple
          Adjust edge frequencies (beginning and end of stopbands)
         ----------------------- ↑ ↓ ↵ Esc------------------------

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```


After these decisions have been made, and we have a set of specifications that is consistent with the filter order, we can continue with the design or go back and redo/undo any of the choices we have made thus far. These options are in the menu below.

```
SPECIFICATIONS FOR A BANDPASS ELLIPTIC FILTER OF ORDER 14

          Sampling Frequency (Hz)     8.00000E+03

        End of lower stopband (Hz)     248.957
       Beginning of passband (Hz)     300.000
             End of passband (Hz)     3.30000E+03
  Beginning of upper stopband (Hz)     3.41537E+03
 Maximum passband attenuation (dB)     0.500000
 Minimum stopband attenuation (dB)     60.0000

              SELECT NEXT ACTION
           Proceed with filter design
     Select another parameter adjustment option
        Select another value for filter order
           Redo filter specifications
                  ↑ ↓ ←┘ Esc

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```

When we choose to *Proceed with filter design* then we enter the BANDTRAN module where the final coefficient values, in floating-point precision, are developed. The opening menu has the following options:

```
05-03-1995                    DISPRO(R)                         12:11:00

                   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       PRINT/PLOT - Poles and Zeros (floating-point)
5  COMPUTE & PLOT - Frequency Response (for any wordlength)
6         QUANTIZE - Coefficients (Hex values, poles & zeros, store in FDF)
7           CREATE - Another Filter Data File (FDF)
                                        (Copyright (C) John O'Donnell, 2005)
  [Type Esc to EXIT].....Choice? .... 2

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

We can discuss each of options briefly, in numerical order.

1 This is the simplest way to use this module: the biquad sections will be sequenced so that the largest magnitude poles are first, the *A* coefficients will be scaled to achieve 0 dB peak transmission from the filter input to the output of each biquad section (this may occasionally not be possible), and a filter data file will be named, created, and initialized with the filter specifications and coefficients.

2 You can set up a biquad sequence of your own choosing, and optionally scale for 0 dB transmissions.

3 The filter specifications and coefficients resulting from the last sequencing and scaling operation, if any, can be viewed on-screen or printed. If an filter data file was just created then these are the coefficients as stored in that filter data file. If you choose this selection first, after the menu appears, then the coefficients will be the "raw" values, before biquad sequencing, scaling, and storage in an filter data file. These values could serve as a reference in order to compare with the results of sequencing and scaling.

4 These are the pole and zero values corresponding to the coefficients that were viewed using selection *3*. Screen plotting of poles and zeros is an educational feature; more useful are the actual numerical values for poles and zeros which can later be compared with those for quantized coefficients (available by selecting option *6* and transferring to the *ROUNDCOFF* module).

5 & 6 These are simply "gateways" to the appropriate modules. Note that the frequency response module has local coefficient quantization capability so you can explore the effects of finite-precision coefficients on the frequency response without storing quantized coefficients in the filter data file.

7 More than one filter data file can be created if, for example, you wish to see the effect of biquad ordering on the filter performance in fixed-point arithmetic.

6.2.2 PMR FIR Filters—Completing the Design

The only modification which you might have to make to the original specifications is to get equal width transition bands for BP and BS filter types. Here we choose an adjustment which gives a transition band width that is the average of the two original transition band widths.

```
               > PARAMETER SPECIFICATIONS FOR BANDPASS FILTER <

                  Sampling Frequency (Hz)     8.00000E+03

               End of lower stopband (Hz)     220.0000
               Beginning of passband (Hz)     330.0000
                     End of passband (Hz)     3.00000E+03
         Beginning of upper stopband (Hz)     3.30000E+03
        Maximum passband attenuation (dB)     0.5000000
        Minimum stopband attenuation (dB)     60.00000
    * Stopband edges should be adjusted to get equal width transition bands *

    Choose one of the following adjustments ... (Type N for no adjustment)

         1.        End of lower stopband (Hz) = 30.00000

         2.  Beginning of upper stopband (Hz) = 3.11000E+03

         3.        End of lower stopband (Hz) = 125.0000
             Beginning of upper stopband (Hz) = 3.20500E+03

(Esc to redo specs)   Choice (1,2,3, or N) ? 3
```

The estimated filter length can be increased or decreased; this makes sense if you have done one design computation iteration and need to increase the length to get better performance, or if you have filter length constraints due to hardware characteristics. When you choose to continue the design (see Sect. 3.2.2.1.5 for arbitrary magnitude response designs) the PMR algorithm is executed and the design results displayed.

```
                  EQUIRIPPLE FIR BANDPASS FILTER OF LENGTH 86        05-07-1995
                                    FILE                             17:55:14

                         Sampling frequency  (Hz) 8.00000E+03

                       End of lower stopband (Hz) 125.0000
                       Beginning of passband (Hz) 330.0000
                             End of passband (Hz) 3.00000E+03
                 Beginning of upper stopband (Hz) 3.20500E+03
                Maximum passband attenuation (specified) (dB) .5
                Minimum stopband attenuation (specified) (dB) 60

            LOW                     HIGH        REJECTION
BAND        EDGE         TYPE       EDGE        OR RIPPLE     DEVIATION     WEIGHT

 1     0.000000          STOP   125.0000         58.53 dB     1.22E-03     2.797E+01
 2     330.0000          PASS   3.00000E+03       0.60 dB     3.42E-02     1.000E+00
 3     3.20500E+03       STOP   4.00000E+03      58.53 dB     1.22E-03     2.797E+01
                            ----SELECT NEXT ACTION----
    ACCEPT design results...create filter data file with coefficients
     REJECT design results...change specifications and do new design

                                   ↑ ↓ ←┘

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```
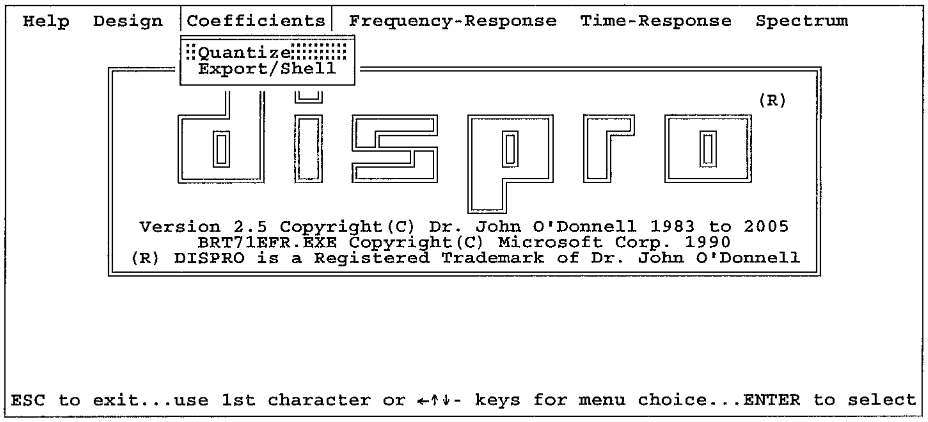
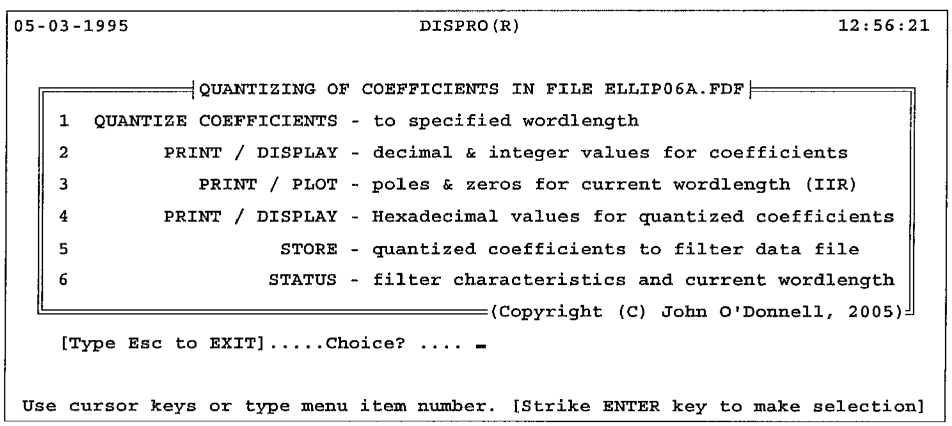

At this point you can return to the specification process if the results are unsatisfactory, but remember that the indicated ripple(s) and attenuation(s) are valid only for filters that were not specified with an arbitrary response characteristic.

6.3 Coefficients



Choosing either of the items on this menu will require specifying a filter data file name. Note that if you want to explore frequency-domain or time-domain properties of the filter using quantized coefficients it is not necessary to enter this menu. The time and frequency response modules will create temporary sets of quantized coefficients as needed.

Choosing the *Quantization* option brings up the following screen.



The major use for this *ROUNDCOF* module is to obtain quantized coefficient values in a couple of different formats, and to write quantized coefficients to the filter data file where they can be accessed by user-created software (and also by the utility programs supplied with DISPRO and discussed in Section 5). You may also obtain pole-zero values for quantized coefficients in order to compare with the pole-zero values for the unquantized coefficients.

Export capabilities are limited in this release of DISPRO to outputting FIR coefficients to a file in formats appropriate to the NEC μPD77C25 and the Analog Devices ADSP2100 programmable DSP processors. The *SHELL* capability allows you to execute DOS commands and external programs without leaving DISPRO.

6.4 Frequency Response

```
Help  Design  Coefficients  Frequency-Response  Time-Response  Spectrum
                            Compute, print, plot

                                                  (R)
   Version 2.5 Copyright(C) Dr. John O'Donnell 1983 to 2005
        BRT71EFR.EXE Copyright(C) Microsoft Corp. 1990
   (R) DISPRO is a Registered Trademark of Dr. John O'Donnell

ESC to exit...use 1st character or ←↑↓→ keys for menu choice...ENTER to select
```

There is only one choice here, and after you have selected a file name the following screen appears (note that you can get here directly from the design phase for IIR or FIR filters by choosing the appropriate menu option).

```
05-11-1995                       DISPRO(R)                        16:50:23

             FREQUENCY RESPONSE COMPUTATION & PLOTTING
  1  AUTOMATIC--Compute and Plot Response (dB Scale)
  2  Specify coefficient WORDLENGTH for response computation
  3  COMPUTE response data for plotting
  4  PLOT response for specified frequency and amplitude range
  5  PRINT/DISPLAY response values (single,multiple,or band of frequencies)
  6  SAVE computed plot response data to filter data file
  7  STATUS REPORT  (Current file and available computed response data)
                                     (Copyright (C) John O'Donnell, 2005)
  [Type X to EXIT].....Choice? .... _

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

The frequency response module is extremely flexible, allowing you to view, print, or plot any segment of the frequency response of an IIR or FIR filter. You have total control over the frequency and magnitude axes for plotting, you can obtain numeric values for any segment of the response, and you can get a hard copy of any graphical or numerical data presentation. We can discuss each of the choices briefly, in numerical order.

1 This is the simplest way to use this module. For IIR filters a set of 500 points is computed covering the range of 0 to one-half the sampling frequency; the dB magnitude and group delay data are plotted. For FIR filters an FFT computation gives 512 points over the same range; only the dB magnitude response is plotted because the phase is linear and the group delay, in samples,is always just one-half the length of the filter. The coefficient wordlength for this automatic mode is always floating point.

2 If you specify a wordlength different from floating point then you will have to use *3* to compute response data, and *4* to plot the response.

3 Under this option you can select any segment of the frequency axis and place up to 1000 points in that segment. For FIR filters you can use the FFT to put up to 4096 points in the range 0 to FS/2.

4 This choice allows you to plot whatever response data are currently available—you can check on the available response data with choice *7*. You will have the options of a linear or dB magnitude scale and, for IIR, the options of plotting phase and/or group delay. The option of drawing the band boundaries lets you see how the response is affected by coefficient quantization—the band boundaries are always for floating-point precision coefficients.

The values for the frequency and magnitude axis scales are automatically chosen so as to show the full range of the available data; you may, of course, override these values and select any frequency and magnitude range you wish to. If you change the default values for the low and high axes limits then you will be presented with two possibly different sets of values for the plotting limits—one labelled *Automatic* and the other labelled *User*. By striking either the *A* or the *U* key you will make a choice. The reason for the choice is that all automatically determined axis intervals in DISPRO are selected to give exact values for each axis interval—no approximate, rounded values—and to allow easy interpolation. To this end, axis intervals in the *Automatic* mode are restricted to the numbers 1, 1.5, 2, 2.5, 3, 4, 5, 6, 8, and 10 multiplied by a positive or negative power of 10. Each of the tick mark values will be printed as an integer or decimal value with up to 10 digits; large numbers which exceed 10 digits, or decimal places, will be printed in power-of-10 notation. The axis labels printed in automatic mode are guaranteed not to exhibit rounding error; in the *User* mode the axis labels are developed using ordinary arithmetic operations and thus may exhibit rounding errors—such as having 0.399999 appear instead of 0.4.

When a plot is on the screen you can get a hard copy by striking the *P* key. The printer must be on line as LPT1: or, after a device timeout period, DISPRO will terminate and you will be returned to the DOS level. To guard against this a check is made for a printer on line each time option *4* is selected; a warning message is displayed if an off line condition is detected. If this message appears it does *not* mean that you must have a printer on line to continue; you may view all plots and data on the screen—just don't try to print. (Note: you can save yourself some trouble by using a resident printer spooler; this will prevent early program termination due to a printer error or an off-line printer.)

5 Screen plots give the overall view and you can perform screen-based interpolation to a reasonably high resolution. But when you need precise values at exact frequencies this choice will give you on-screen as well as printed values and tables.

6 If you wish to perform further processing of the frequency response, such as integrating to obtain gain-bandwidth or noise-gain values, then you can save the response data to the filter data file and process it with your programs or with a modification of a DISPRO utility program (see Section 5).

7 This choice allows you to check on what's available for plotting: the filter parameters, the current coefficient wordlength, and the number and range for the currently computed frequency response.

6.5 Time-Response

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

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              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
```

The time response of an IIR or FIR filter can be computed in floating-point or simulated fixed-point arithmetic. Test signals can be generated for use in the forced response computation, and the results of the impulse/step/sinusoidal response or forced time response computation can be plotted. In fact, the input for a forced time response calculation can come from any properly prepared file—for example, experimental data—and any properly structured file of time samples can be plotted.

Compute Impulse/Step Response: When you select the impulse/step response option you must pick a Filter Data File (FDF) containing the filter coefficients. Although DISPRO provides the capability to compute the impulse and/or the step response of an FIR filter, the result is exactly what you would get from viewing the quantized filter coefficients and their sum. The only reason for using the time response module for the impulse/step response of an FIR filter is to spot any temporary accumulator overflows in computing the step response.

The first screen presented by the *TIMERESP* module is

```
                 IMPULSE/STEP/SINUSOIDAL RESPONSE OF IIR FILTER
                        FILE E:\FILTER\DATA\BUTTR07A.FDF

► Status of parameters
       Topology: (none selected yet)
   Coefficients: (none read from file yet).
     Arithmetic: floating point
Overflow action: Ignore
►  Breakpoint settings  (none)
                              SELECT ACTION
           COEFFICIENTS -- Specify wordlength
             ARITHMETIC -- Establish computational parameters
               OVERFLOW -- Set action for overflows
            BREAKPOINTS -- Set time index values for breakpoints
                    NEW -- Begin the computation of filter response
                 RESUME -- Continue current response computation
   SAVE/PRINT/DISPLAY -- Response sample values in buffer
         IMPLEMENTATION -- View/print equations and scaled coefficients
                              ↑ ↓ ←┘ Esc
                Strike Esc key to exit time response module
```

Your first step is to select a wordlength for the coefficients. If the coefficient wordlength is less than the computational wordlength then specify it now; if not, then when a computational wordlength is specified the coefficients will automatically be rounded to that precision. Basically, the coefficients in the FDF are floating-point precision; you may read them in and change to any desired coefficient wordlength when you wish.

Selecting the *ARITHMETIC* option brings up a screen on which you can establish the computational parameters. Floating point computation is the default startup precision; the screen below shows the selection, for an IIR filter, of 16-bit (i.e., 15 bits plus sign) fixed-point arithmetic with ordinary 2's-complement accumulator operation (which is unsaturated), a double-length accumulator, rounding of the double-length accumulator value when it is converted to single length and stored in memory, and the Merged-Biquads topology.

```
                    SELECTION OF ARITHMETIC CHARACTERISTICS

 COMPUTATIONAL WORDLENGTH:  (2 to 23 bits, not including sign)   15
TWO'S COMPLEMENT ADDITION:  Unsaturated or Saturated.....(U/S)   U
             ACCUMULATION:  Single or Double length......(S/D)   S
             QUANTIZATION:  Rounding or Truncation.....(RD/TR)   RD
        BIQUAD STRUCTURE:  Merged or Canonic form.......(M/C)   M

   Type new values...use normal editing and cursor keys... <Esc> = Cancel
```

There is a similar screen which appears for FIR filters, without, however, a filter topology selection. By default all FIR filters simulated in the time response module use the direct-form topology—*i.e.*, the FIR filter output is computed as a sum of products using the convolution sum formula.

```
                    SELECTION OF ARITHMETIC CHARACTERISTICS

 COMPUTATIONAL WORDLENGTH:  (2 to 23 bits, not including sign)   15
TWO'S COMPLEMENT ADDITION:  Unsaturated or Saturated.....(U/S)   U
             ACCUMULATION:  Single or Double length......(S/D)   S
             QUANTIZATION:  Rounding or Truncation.....(RD/TR)   RD

   Type new values...use normal editing and cursor keys... <Esc> = Cancel
```

The *NEW* option starts a new computation and displays a snapshot of the filter variables. (For the significance of the display, and the filter scaling choices that are made before computation begins, see Sect. 3.3.3.) Any time you want to stop the computation the *Esc* key will get you back to the action menu screen. Note the top legend on the screen which states the reason that the computation has stopped. The *RESUME* option will resume the computation, and you will be able to select or deselect single-step operation. When the last breakpoint has been reached, or 16384 samples have been computed, the computation stops. If the computation is now complete—in this case, if the impulse/step response has effectively reached steady state—then you can save the results in the FDF by using the *SAVE* option. In order to plot or spectrum analyze the time response data it is *necessary* to save the time response samples to the FDF.

Compute Forced Time Response : The procedure is much the same—and is carried out by the same module—as for the impulse/step response computation. The difference is that you must give a second file name—for the excitation file—and when the *NEW* option is selected for beginning a new computation in the time response module you may have to choose a scale factor for the excitation signal if the peak value of the samples is greater than unity in magnitude.

Generate Test Signal : The *WAVEGEN* module provides a very general means for creating a set of time sample data for testing the response of a filter. The menu items are self-explanatory. Generated data are normally stored in a disk file for use as an excitation source, or for plotting or spectrum analysis.

```
                     S I G N A L   G E N E R A T I O N

                          SELECT TYPE OF SIGNAL
   SUM OF SINE WAVES -- Arbitrary frequencies or Fourier Series
   SPECIAL WAVEFORMS -- Rect. pulse, square/sawtooth/triangle/chirp
      GAUSSIAN NOISE -- Zero-mean white, specify mean-square
      KEYBOARD INPUT -- Arbitrary set of time samples
        SAVE TO FILE -- Store waveform samples in disk file
                              ↑ ↓ ←┘ Esc

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```

Plot Time Samples : The time domain plotting procedure is similar to the frequency response plotting approach. You can plot impulse/step or forced time responses in IIR/FIR filter data files, the impulse response (using the coefficients) of an FIR filter, or time sample data in an excitation file—one produced by the Generate Test Signal option, or a user-created file (see Sect. 3.3.3.3.2 for procedure).

6.6 Spectrum

```
Help  Design  Coefficients  Frequency-Response  Time-Response  Spectrum
                                                              Fourier Spectrum
                                                              Fourier Series
                                                                   (R)

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ESC to exit...use 1st character or ←↑↓→ keys for menu choice...ENTER to select
```

Spectrum computation is done with a radix-2 FFT, which is embedded in a procedure that efficiently computes spectra of real-valued time sample sequences up to 8192 points long. The differences between the *Series* and *Spectrum* options are in the assumed nature of the samples and the scaling of the results.

When you select the *Fourier Series* option it is assumed that the set of time samples being processed consists of one or more periods of a periodic time function. You will also use this option when the set of time samples is a sum of tones which are not harmonically related, thus making the signal non-periodic; in this case the spectrum levels for each tone will not be exact, but the error will be small if a large number of cycles of each tone is included in the set of time samples processed by the FFT. When the spectrum is computed the spectral values are scaled by 1/N where N is the number of real-valued samples processed. Thus, to get the exact Fourier series coefficients and magnitude spectrum from an N-point FFT, the set of time samples processed must have exactly N samples in one or more periods of the periodic signal—where N is equal to a power of two. To be precise, you need only have the N samples in one period—additional periods provide no more information, of course. Because this condition is often difficult to meet in practice, you should anticipate processing more than one period, noting that there is less error in FFTing 100.5 periods than in FFTing 1.5 periods. Also, the time samples must not be windowed. For further information refer to Sect. 3.4.

The *Fourier Spectrum* option is the one you should select when, for example, obtaining the frequency response of an IIR filter from the impulse response computed in fixed-point arithmetic; in this case the FFT is just evaluating $H(z)$ for $z = e^{jw}$—windowing should not be used. If the Fourier spectrum option is used for a set of values which represent the sampled version of a continuous-time waveform—basically, a set of time samples from something other than a filter data file—then the correct spectral levels are obtained only if the spectrum is scaled by $T = 1/(\text{sampling frequency})$. A prompt for the sampling frequency appears after the FFT computation; this value for the sampling frequency is used both to scale the spectrum and to provide the scale for the frequency axis of the plot, although no response is necessary if you wish to see only the relative spectral levels in a spectrum plot which is normalized to the largest component.