

ALE STARTER GUIDE

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ABSTRACT

The automatic loudspeaker equalization (ALE) program supports the following Texas Instruments digital audio processors

- TAS3001
- TAS3002
- TAS3004
- TAS3103

This application note addresses the use of ALE in developing application-specific coefficients for the TAS3103 digital audio processor.

The TAS3103 digital audio processor is a three-channel audio processor. Many audio processing features are provided in the TAS3103, including

- Up to sixteen biquad filters per channel for speaker equalization
- Loudness compensation unique to each channel
- Dynamic range compression / expansion

The automatic loudspeaker equalization (ALE) program provides a user-friendly environment in which to fully use the capability of the TAS3103 in a given application. Specifically, the ALE provides the ability to

- Develop equalization filters by
 - Specifying filter type(s) and parameters
 - Specifying a desired speaker response given a measured speaker response
- Tailor loudness compensation to achieve a specified audio presentation
- Set dynamic range parameters

This application note consists of three major sections, with each section addressing one of the three previously listed capabilities.

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Initial Setup Steps

The ALE software package is provided with every TAS3103EVM, and is also available to users who do not wish to purchase a TAS3103EVM. Requests for the ALE package alone can also be sent to digitalaudio@ti.com, along with descriptions of the intended end use.

The first step to using the ALE software package is to unzip and install it. Upon opening the installed ALE package, the GUI shown in Figure 1 appears. This GUI only applies when using the ALE to create equalization (EQ) filter coefficients and is the default GUI. Two other GUIs can be accessed – a GUI for generating dynamic range compression coefficients and a GUI for generating loudness compensation filter coefficients.

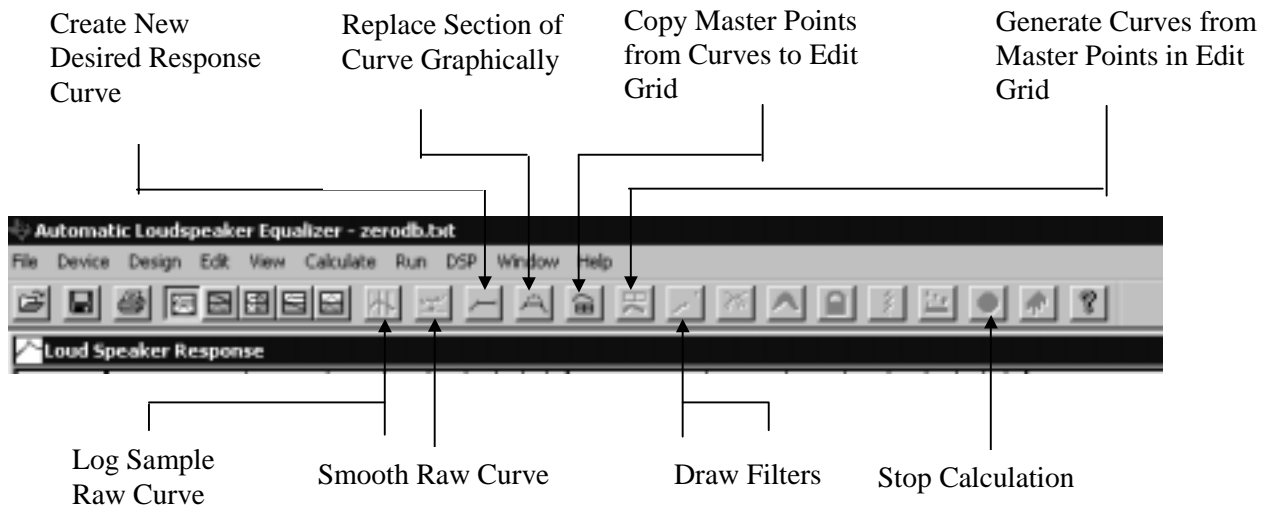


Figure 1: Automatic Loudspeaker Equalizer GUI

The first step that needs to be taken in using the ALE is to click on the *Device* button and select *TAS3100*. Next, the sample rate of the application being address by ALE needs to be set. This is done by clicking on the *View* button and selecting *Settings*. The *Frequency Limits* tab should come up, but if it does not, select it. In the *Frequency Limits* panel that comes up, the default setting for *Fs Sampling Frequency Hz* is seen to be 44100 (44.1 KHz). If this is not correct, enter the correct value and click on the *OK* button. Now the sample rate is properly set.

Filter Coefficient Generation

Manual Filter Generation

To generate a filter, click on the *Edit* button and select *Edit Filter Parameters*. The panel shown in Figure 2 results.

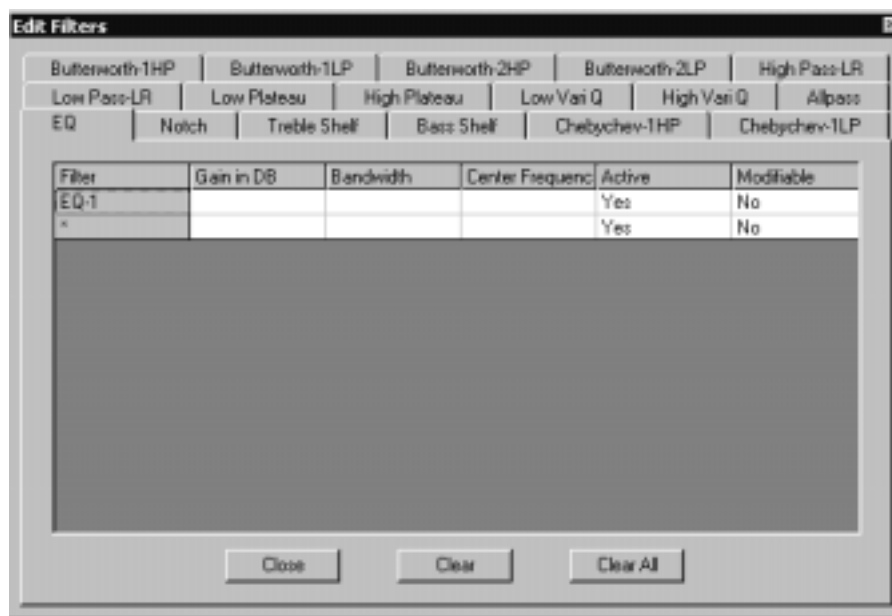


Figure 2: Edit Filter Parameters Panel

Butterworth-1HP and Butterworth-1LP are single-pole Butterworth high-pass and low-pass filters. Butterworth-2HP and Butterworth-2LP are two-pole (2nd order) Butterworth high-pass and low-pass filters. High Pass-LR and Low Pass-LR are Linkwitz-Riley high-pass and low-pass filters commonly used in building audio cross-over filters. The programmed cut frequency for these filters is the 6 dB cut frequency rather than the standard 3 dB cut frequency. The low plateau filter has 0 dB gain at frequencies above the upper band edge, peaks at the selected gain between the upper and lower band edges, and exhibits a single-pole roll off below the lower band edge. The high plateau filter has 0 dB gain at frequencies below the lower band edge, peaks at the selected gain between the upper and lower band edges, and exhibits a single-pole roll off above the upper band edge. The low VARI Q and high Vari Q filters are 2nd order low-pass and high-pass filters with selectable peaking at the selected 3-dB frequency. The allpass filter is a second-order unity-gain filter with a monotonically decreasing phase of 2π over the frequency band specified. The EQ filter is a bell-shaped filter with 0 db gain outside the programmed bandwidth, and a peak gain as specified within the specified bandwidth. The notch filter is a 2nd order notch filter. The treble shelf and bass shelf are standard 2nd order shelf filters. The cut frequency for treble shelf filters is the 3 dB cut frequency with respect to the response at frequencies above the cut frequency. The cut frequency for bass shelf filters is the 3 dB cut frequency with respect to the response at frequencies below the cut frequency. The Chebyshev-1HP and Chebyshev-1LP filters are 2nd order Type-I Chebyshev high-pass and low-pass filters (the 1 in this case references the type Chebyshev filter rather than the order of the filter).

Once a filter type is selected and specified (by clicking on the appropriate tab and entering the required parameters), clicking on the *Draw Filters* button generates the filter's response. If the response meets requirements, go to the *Edit* button and select *Generate TAS3xxx Data* to generate the biquad coefficients required to implement the specified response. Going to the *Edit* button and selecting

Edit TAS3xxx Data displays the values of the generated biquad coefficients, and provides a means of manually editing the coefficient file if needed.

Additional filters can be specified and cascaded with filters previously selected and specified by returning to the *Edit Filters* panel, selecting another filter type, and entering its parameters. Clicking on the *Draw Filters* button places the specified filter in cascade with the filters defining the response currently displayed, and generate a new composite response. To save the composite response, go to the *Edit* button and select *Generate TAS3xxx Data*. To add more filters, repeat the *Edit Filters* and *Draw Filters* sequence until the response required is obtained. If a given filter addition does not produce the response desired, it can be eliminated by going to the *Edit Filters* panel, highlighting the filter to be eliminated, clicking the right button on the mouse, and clicking on *OK* in the *Confirm Filter Deletion* panel that appears. To see the results, having eliminated the filter, click again on the *Draw Filters* button.

If the displayed response is the response desired, the TAS3103 coefficients implementing this response need to be generated and then saved. Clicking on the *Edit* button and selecting *Generate TAS3xxx Data* generates the coefficients. The generated data can then be saved to a file by clicking on the *File* button and selecting *Save As TAS3xxx Program Data*. The file name and directory can be selected, but the file extension must remain EQ310X. Retaining this extension makes the file compatible with the TAS3103 DAS-DCT 2.0 GUI, and from this GUI, the filter can be downloaded into the TAS3103 chip.

There can be confusion over the differences in the roles of *Save As TAS3xxx Program Data* and *Generate TAS3xxx Data*. When the ALE is first opened and a filter suite is created, clicking on *Save As TAS3xxx Program Data* both generates and saves the filter coefficient set. Ever after, however, the *Generate TAS3xxx Data* must be selected before the *Save As TAS3xxx Program Data* is selected in order to save the latest changes to a previously generated filter suite or to save a newly generated filter set.

Graphic Filter Generation

ALE also provides the ability to draw a required response, and then derive the filter coefficients required to attain the drawn response. (EQ or bell-shaped filters are used exclusively when deriving filters to implement drawn responses). To use this feature, it is first necessary to go to the *File* button and select *Open Raw Response File*. Files with both .txt extensions and .raw extensions can be used to specify the raw response, as long as they have the structure shown below. The *Data* column is the magnitude response in dB at the frequency specified.

Hz	Data
12.20703,	-39.43244
24.41406,	-39.74691
36.62109,	-40.19416
48.82813,	-40.70994
61.03516,	-41.2616
●	●
●	●

One use for these files is to input speaker response curves. A previously recorded speaker response is input into ALE using *Open Raw Response File* and selecting the desired raw response. A user then draws a desired speaker. Having drawn the desired speaker response, the ALE can then be commanded to generate a filter that implements the drawn filter response with respect to the raw response reference. (The desired response can also be input as a file. This option is discussed in the next section).

To begin the construction of a graphically-specified filter response, a raw response must first be input. When filters are created to implement the specified drawn response, they are created to provide the specified response given the selected raw response as the starting point. To illustrate this process, an example is presented. To start, go to the *File* button, select *Open Raw Response File*, and then select *zerodb.txt*. This selection results in a 0 db reference response (shown as a straight blue line). Next it is necessary to draw the filter response desired. To draw a filter response, click on the *Create New Desired Response Curve* button. Place the mouse at the desired start point and draw straight-line segments to create the filter response desired. When done, click on the *Create New Desired Response Curve* button

again. This causes a set of magenta boundary lines to be drawn. These lines define the excursions allowable in generating the filter response from the desired response drawn. An example of a response graphically created is shown in Figure 3, and includes the blue 0 dB raw response curve.

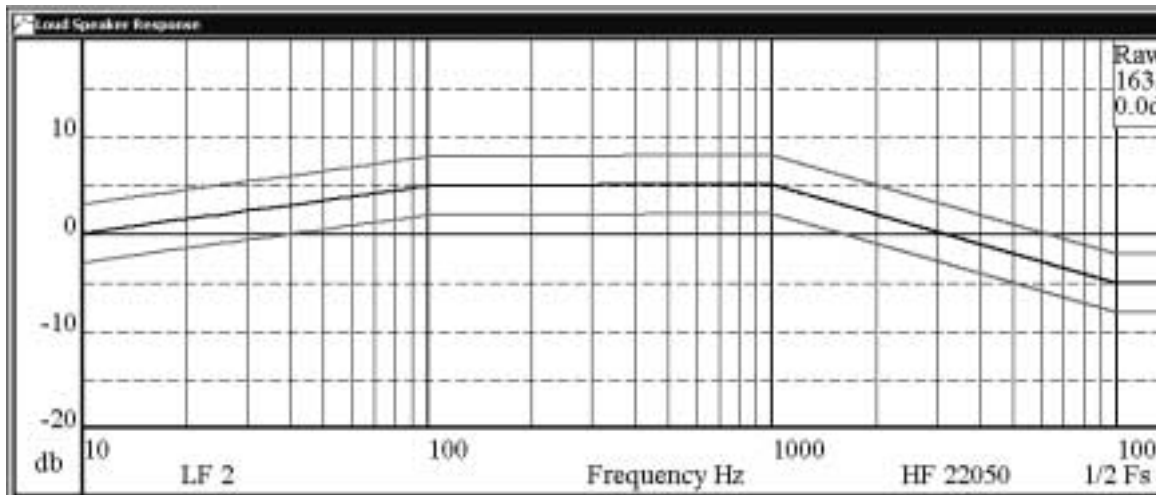


Figure 3: Graphically-Created Desired Response Curve

If a mistake is made in drawing the desired response, clicking on the *Create New Desired Response Curve* button a third time will bring up the window shown in Figure 4. Selecting Yes deletes the drawn desired response curve.

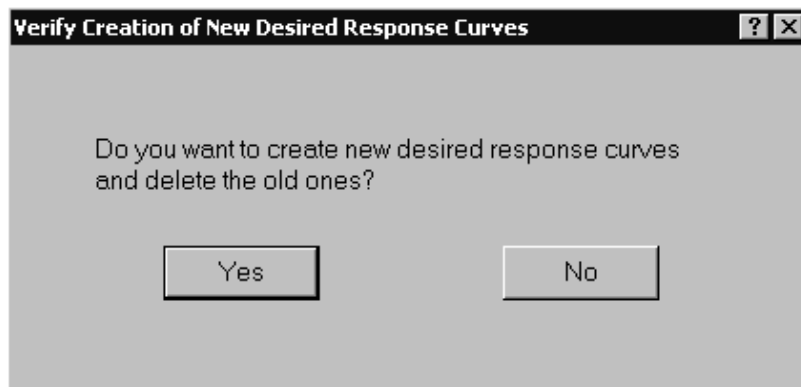


Figure 4: Create New Response Curve Panel

The graphically created response can also be saved as a desired response file by clicking on the *File* button and selecting *Save as Desired Response*. The file name and directory in which to store the file can be specified, but it is important to maintain the extension .crv. The use of .crv files is illustrated in the next section.

Once drawn, transferring the points describing the desired response into the computation grid enables manual refinements to be made to the drawn desired response. This transfer is accomplished by clicking on the *Copy Master Points From Curves To Edit Grid* button. The panel shown in Figure 5 comes up.

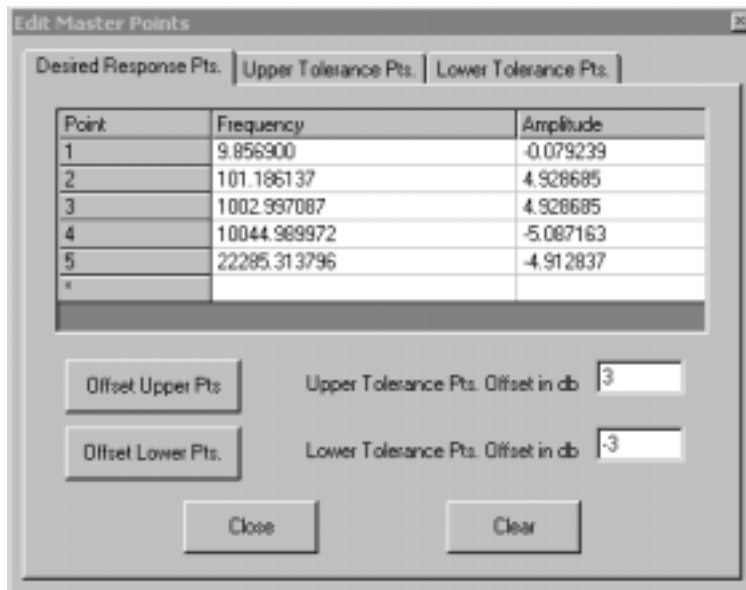


Figure 5: Edit Master Points Panel

This panel shows the frequency and amplitude of the Master or inflection points of the drawn response curve. These Master points can be edited manually. For example, the amplitude of point 3 can be changed from 4.928685 to 9.0. Master points can also be deleted or added. To delete a Master point, simply right-mouse click on the Master point to be deleted and then click on the *OK* button in the *Confirm Master Point Deletion* panel. Master points can be added by going to the empty row in the *Edit Master Points* panel (the row with the * in the Point column) and typing in the frequency and amplitude of the point to be added.

Any time a change is manually made to the Master points, the *Offset Upper Pts* button and the *Offset Lower Pts* button must be clicked to apply the upper and lower tolerances to the revised set of Master points. The *Edit Master Points* panel can also be used to manually change the upper and lower range of the tolerance curves. Changing tolerance range values manually is accomplished by entering the revised tolerance range value and clicking on the appropriate *Upper Tolerance Pts* and *Lower Tolerance Pts* buttons. The *Upper Tolerance Pts* and The *Lower Tolerance Pts* tabs in the *Edit Master Points* panel, on the other hand, are used when it is desired to change the tolerance level setting at a given individual Master point.

To observe the effect of any manual adjustments on the drawn response, the *Generate Curves from Master Points in Edit Grid* button must be activated. To save the adjustments made on the drawn response, click on the *File* button and select *Save as Desired Response*.

The drawn desired response curve and tolerance curves can also be graphically edited. To graphically edit a drawn response curve or tolerance curve, click on the *Replace Section Of Curve Graphically* button. Then place the cursor on the section of the curve to be modified and drag the mouse to draw the modified shape. Note that this time the shape drawn follows the path of the mouse, and is not restricted to straight-line segments. Terminate the curve modification by releasing the mouse at a point on the response or tolerance curve. The start and stop points of the graphic modification must always be on the original curve.

Once the response curve and the tolerance curves are as required, clicking on the *Calculate* button and selecting *Modified Response* generates the filters. (Several EQ filters may be required to attain the desired response. The *Modified Response and Pause* selection stops the process after each filter is added, requiring the user to click on the *OK* button on the *Continue Calculation* panel that comes up between filter calculations, to continue the process). While the filters are being derived, the *Stop Calculation* button – the third button from the right – becomes red. This is an important feature as the computation process can

require minutes to complete, and the red light is the only indication that the program is computationally active. As the filters are being calculated the current modified response curve appears in red and the prior modified response curve appears in yellow. Clicking on the *Stop Calculation* stops the calculation but retains those filters completed up to the point the program was stopped.

The resultant graphic filter design for the drawn filter response example depicted previously is shown in Figure 6.

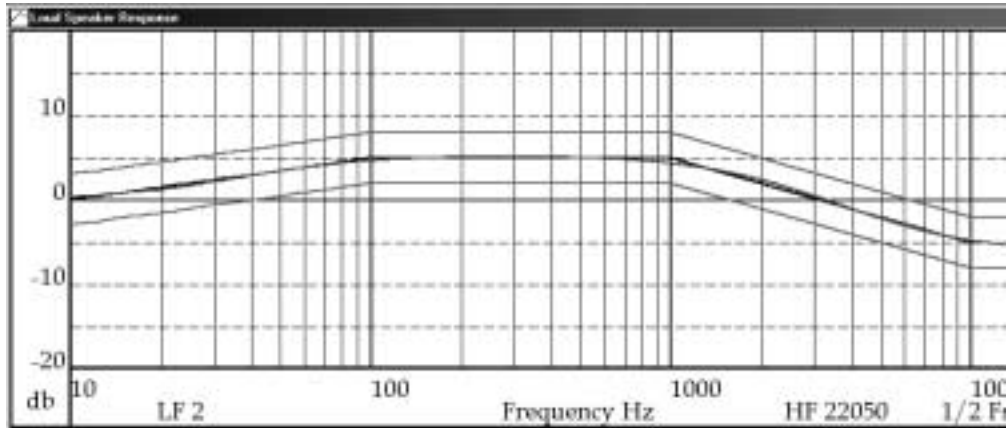


Figure 6: Resultant Response of Graphic Filter Design Example

Once a filter is calculated, to derive the biquad coefficients for the resultant response it is necessary to first click on the *Edit* button and select *Copy Generated Filters to Edit Grid*. This results in the automatically generated filters appearing in the *Edit Filters* panel, shown in Figure 2. The biquad coefficients for these filters cannot be generated until the filters reside in the *Edit Filters* panel. Once in the *Edit Filters* panel, if no manual modifications are to be made, clicking on the *Edit* button and selecting *Generate TAS3xxx Data* generates the biquad coefficients. The *Edit TAS3xxx Data* panel shown in Figure 7 appears. Application specific data, such as Speaker Model #, Speaker Serial #, etc. can be entered in this panel, and the biquad coefficients generated can be manually modified by typing in changes.

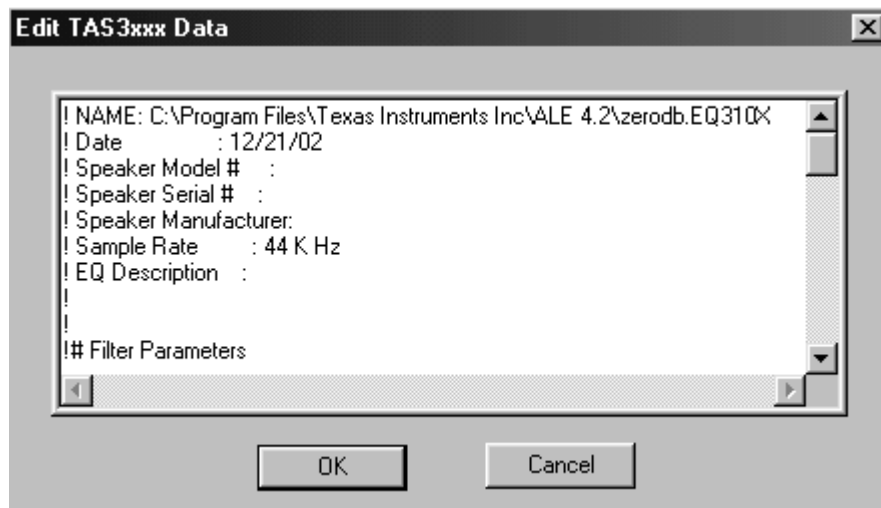


Figure 7: Edit TAS3xxx Data Panel

Clicking on the *File* button and selecting *Save as TAS3xxx Program Data* creates a file containing the filter coefficients, along with the application specific data. The directory to store the file and

the file name can be selected, but the file extension must remain .EQ310X. The content of the .EQ310X file for the example illustrated is presented in Figure 8. As seen, this example results in the generation of three biquad filters, and each filter is labeled by an Q## heading (Q1, Q2, and Q3). When the TAS3103 DCT GUI retrieves this file it discards the top information, replaces the Q## number by a sub-address (x##), and downloads the coefficients into the specified TAS3103 filter bank.

```

! NAME: A:\Starter_Example_01.EQ310X
! Date       : 12/27/02
! Speaker Model #   : Starter_Guide_Speaker
! Speaker Serial #  : Starter_Guide_Serial #
! Speaker Manufacturer: Starter_Guide_Manufacturer
! Sample Rate      : 44 K Hz
! EQ Description   : Starter_Guide_EQr
!
!
!# Filter Parameters
!# EQ-1, 5.149749, 2006.389480, 309.887964, No, Yes, 1
!# EQ-2, -5.204320, 14088.737454, 13022.071425, No, Yes, 2
!# EQ-3, -4.825576, 1575.456844, 22050.000000, No, Yes, 3
! Channel Number 1
Q1
00 DF 93 45
FF A0 34 E7
00 8D 07 E6
FF 20 6C BB
00 52 C3 33
Q2
FF ED 5F F9
00 3D A6 6F
00 55 42 5F
00 12 A0 07
FF ED 17 31
Q3
FF 2A 08 13
FF AA 08 13
00 77 0A BF
00 D5 F7 ED
00 5E ED 2D
    
```

Figure 8: ALE Filter File: Starter_Example_01.EQ310X

Equalization Filter Generation

This section addresses using ALE to create equalization filters. In doing so, many other features of ALE will be described – including logarithmic sampling, sample smoothing, frequency boundary manipulation, tolerance boundary manipulation,

In the previous example, the Raw Response file zerodb.txt was a 0 db reference response shown as a straight blue line. This choice allowed the drawn response to be the response of the filter generated. Another use of the Raw Response file is to provide a means of generating equalization filters directly from the raw response curves of the speakers. To do this, the Raw Response file is the raw response curves of the speakers, and the drawn response is the desired speaker response. In this case, ALE creates a suite of filters to transform the speaker response from the raw, unprocessed response to the drawn, or desired, response. And, in addition to being able to specify the desired speaker response by drawing the desired response, the desired speaker response can also be specified by inputting a Desired Response file - a .crv file.

To illustrate the process, the raw speaker response curve6.raw is imported into ALE by going to the *File* button, selecting *Open Raw Response File*, and clicking on curve6.raw. This curve represents the response of the speaker to be equalized, and is shown in Figure 9.

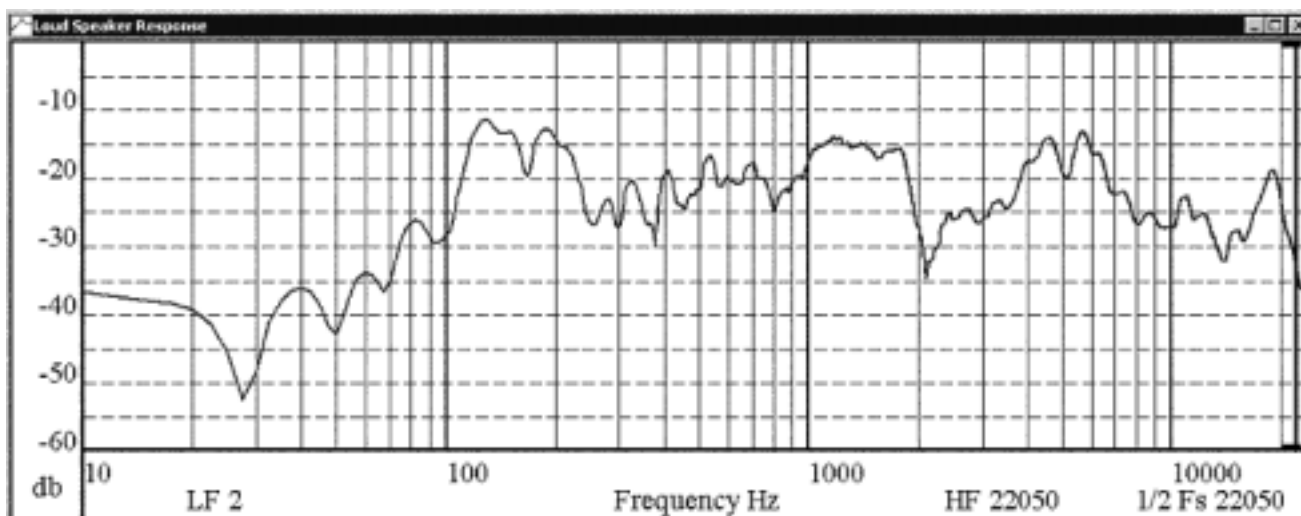


Figure 9: Raw Speaker Response Example – Curve6.raw

The points that comprise curve6.raw are linearly placed in frequency. The first task is to convert the point placement to a logarithmic grid to improve the efficiency of the ALE computation process. (In many applications the improvement in processing time can be significant, and is attained with no compromise in the quality of the equalization process). To convert the raw response curve to a logarithmic grid, click on the *Log Sample Raw Curve* button, enter the desired number of logarithmically sampled points (300 is typically used), and then click *OK*.

If ALE has to contend with sharp discontinuities in the Raw Response curve while generating the filter suite required to deliver the desired response, excessive computation time and an excessive number of generated filters results. To combat this problem, a smoothing function is provided to smooth the shape of the raw response curve prior to initiating the computation process. Eliminating the sharp discontinuities in the raw response curve notably improves calculation time and filter count, but has little effect on the overall effectiveness of the resulting suite of equalization filters. The smoothing process is applied by simply clicking on the *Smooth Raw Curve* button. Figure 10 shows the raw response curve that results after

clicking on the *Smooth Raw Curve* button four times. Note that the sharp discontinuities have been eliminated.

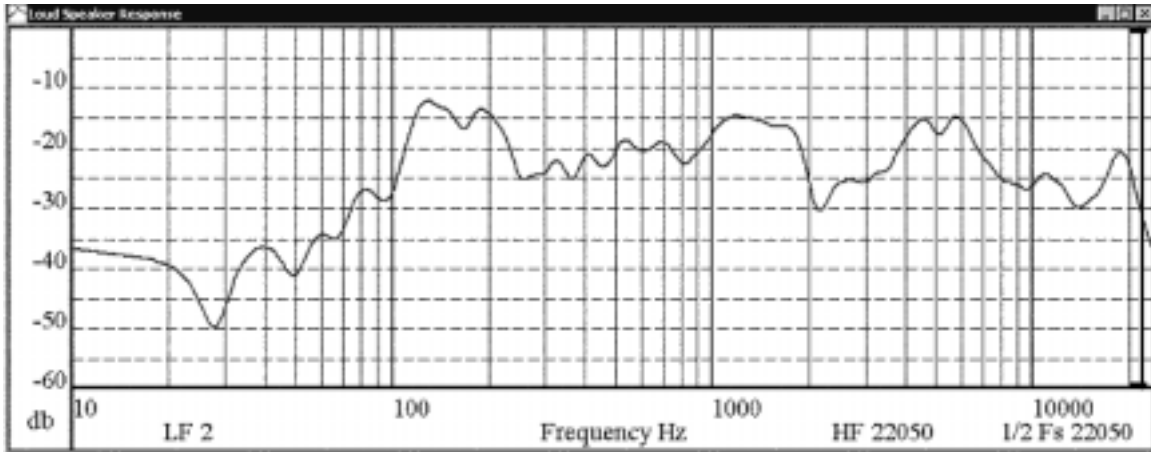


Figure 10: Raw Speaker Response Curve6.raw After Smoothing

. Next the desired filter response is imported by going to the *File* button, selecting *Open Desired Response File*, and, for this example, selecting *desired_app_note.crv*. [.crv files must be generated within the ALE program by drawing and editing (if required) a desired response, and then saving the desired response by going to the *File* button and selecting *Save as Desired Response*, being sure to maintain the .crv extension.]

The resulting graphic is shown in Figure 11. The blue graph is the raw response, the black graph the desired response, and the magenta graphs the tolerance bounds about the desired response.

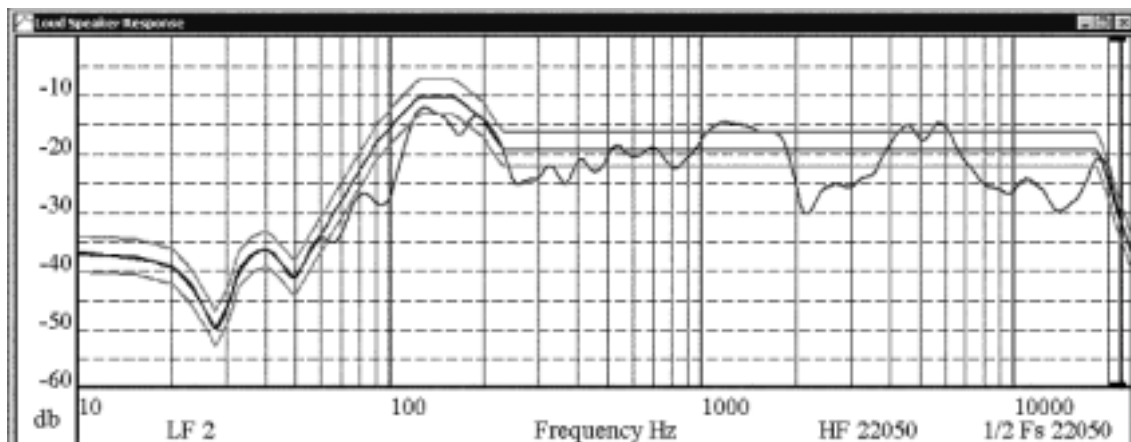


Figure 11: Desired Response Overlaid Onto Smoothed Raw Speaker Response

As seen, for this example the tolerance bound is ± 3 dB. The computational stress points in calculating filters to bring the raw response to the desired response within a tolerance specified by the tolerance band occur on the steep slopes of the desired response curve. If the tolerance bands can be relieved in these sections of the desired response, both calculation time and generated filter count are reduced. To modify the tolerance bands manually, first click on the *Replace Section of Curve Graphically* button. Then click on one of the ends of the section of the upper or lower tolerance band to be manually modified, and drag the mouse to shape the curve as desired, being sure to complete the dragging of the mouse on the tolerance band at the other end of the section being modified. Figure 12 shows the results of

manually modifying the tolerance bands bounding the steep slopes of the desired response curve – which are the frequency sections 30 to 130 Hz and 18 to 22 KHz.

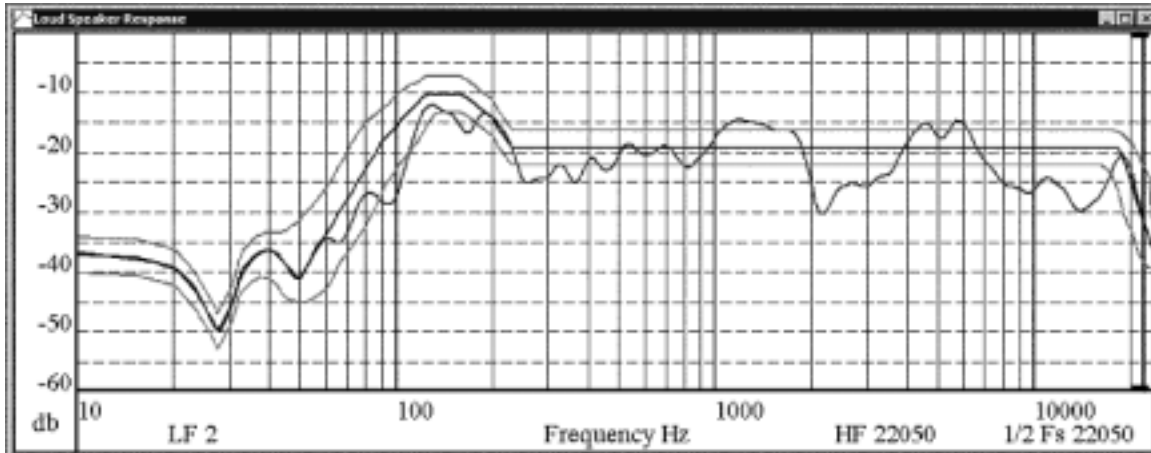


Figure 12: Desired Response Curve After Manual Alteration of Tolerance Bands

One more modification is made before generating the equalization filters. The equalization process need not extend down to the default lower frequency bound of 2 Hz, nor up to the upper frequency bound of $F_s / 2$ (Nyquist frequency). Adjusting the frequency bounds to more reasonable settings further reduces computation time and filter count. To adjust the frequency bounds, click on the *View* button and select *Settings*. This brings up the panel shown in Figure 13. As seen, the lower frequency bound has been set to 50 Hz, and the upper bound has been set to 20 KHz. (Also note that the sampling frequency has been set to 44.1 KHz.). When the desired frequency bounds have been set, the *Apply* button must be clicked.

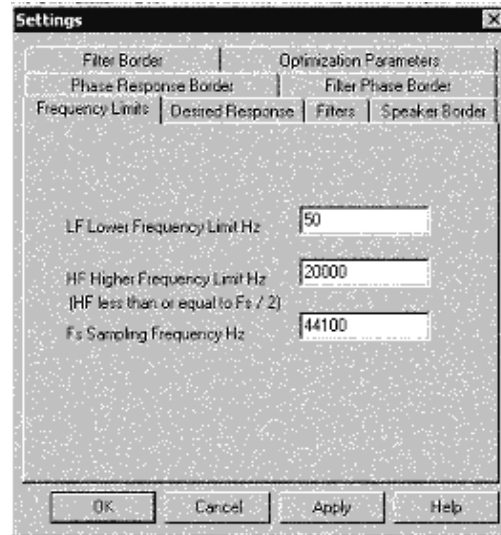


Figure 13: EQ Filter Design Settings Panel

Now the filters are ready to be calculated. Click on the *Calculate* button and select *Modified Response*. The computed filter response that results is shown in red in the plot of Figure 14. (The yellow plot is the modified response prior to the calculation of the final filter).

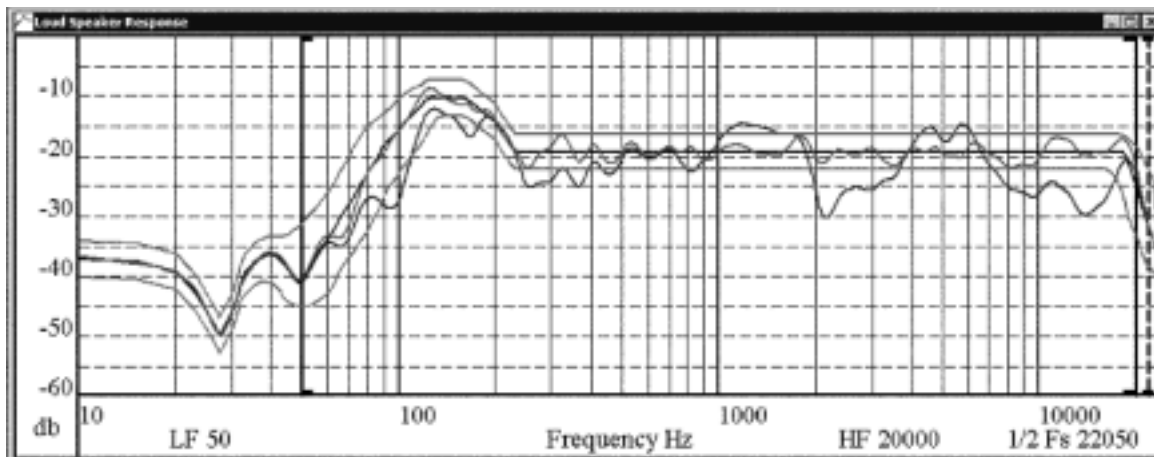


Figure 14: Computed EQ Filter Response

This particular example required 11 bell-shaped equalization filters, and these 11 filters are shown in Figure 15.

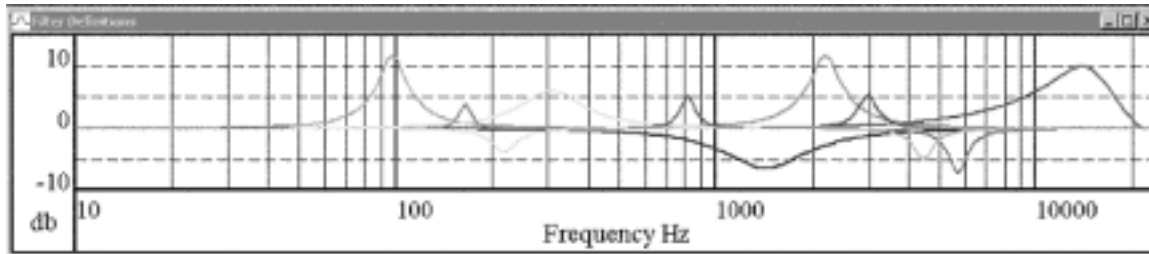


Figure 15: Filters Comprising Computed EQ Filter Response

A file containing these 11 sets of filter coefficients can be created by:

1. Clicking on the *Edit* button and selecting *Copy Generated Filters to Edit Grid* to transfer the generated filters to the Edit Grid domain.
2. Clicking on the *Edit* button and selecting *Generate TAS3xxx Data* generates the biquad coefficients
3. Clicking on the *File* button and selecting *Save as TAS3xxx Program Data* creates a .EQ310X file containing the filter coefficients, along with the added application specific data.

The content of the file is shown in Figure 16.

```

! NAME: C:\Program Files\Texas Instruments Inc\DAP Config Tool
2.0\Starter_Example_02.EQ310X
! Date       : 2/21/03
! Speaker Model #   :
! Speaker Serial #  :
! Speaker Manufacturer:
! Sample Rate   : 44 K Hz
! EQ Description  :
!
!
!# Filter Parameters
!# EQ-1, 9.997519, 5265.459281, 13930.606494, No, Yes, 1
!# EQ-2, 11.797349, 344.547785, 2194.327307, No, Yes, 2
!# EQ-3, -6.285571, 745.270301, 1432.655445, No, Yes, 3
!# EQ-4, 11.882147, 13.814435, 97.755808, No, Yes, 4
!# EQ-5, 6.096446, 124.661630, 307.354841, No, Yes, 5
!# EQ-6, -7.208300, 608.987920, 5701.820308, No, Yes, 6
!# EQ-7, 5.372781, 380.760577, 2985.929232, No, Yes, 7
!# EQ-8, -4.855725, 568.281489, 4456.470584, No, Yes, 8
!# EQ-9, 5.259423, 79.508570, 818.863435, No, Yes, 9
!# EQ-10, 3.996107, 13.110451, 165.030929, No, Yes, 10
!# EQ-11, -3.798699, 36.377875, 217.753747, No, Yes, 11
! Channel Number 1
Q1
FF B6 1D 52
FF C8 52 68
00 CE 28 47
00 49 E2 AE
FF E9 85 52
Q2
00 ED C0 F0
FF 86 22 59
00 88 DC 91
FF 12 3F 10
00 71 01 16
Q3
00 E1 ED E1
FF 99 47 F4
00 79 7D 66
FF 1E 12 1F
00 6D 3A A6
Q4
00 FF B9 37
FF 80 40 6F
00 80 5E 4F
FF 00 46 C9
00 7F 61 42

```

```

Q5
00 FD 80 D5
FF 82 40 E5
00 81 25 81
FF 02 7F 2B
00 7C 99 99
Q6
00 A0 21 A9
FF 97 2D 05
00 79 77 2E
FF 5F DE 57
00 6F 5B CD
Q7
00 E3 05 99
FF 86 C3 1C
00 82 E5 23
FF 1C FA 67
00 76 57 C1
Q8
00 C0 78 E1
FF 90 EF A2
00 7C 5F A7
FF 3F 87 1F
00 72 B0 B6
Q9
00 FC D3 E2
FF 81 71 1D
00 80 99 95
FF 03 2C 1E
00 7D F5 4D
QA
00 FF B0 C0
FF 80 3D 27
00 80 11 DC
FF 00 4F 40
00 7F B0 FC
QB
00 FE DA A2
FF 81 05 F5
00 7F D1 99
FF 01 25 5E
00 7F 28 71

```

Figure 16: ALE Filter File Starter_Example_02.EQ310X

Dynamic Range Compressor (DRC) Coefficient Generation

The EQ filter design GUI is the default GUI appearing when the ALE package is opened. To go to the dynamic range compressor (DRC) GUI, first click on the *Device* button and select *TAS3100*. Then click on the *Design* button and select *Dynamic Range Compression for TAS3100*. The GUI shown in Figure 17 appears.

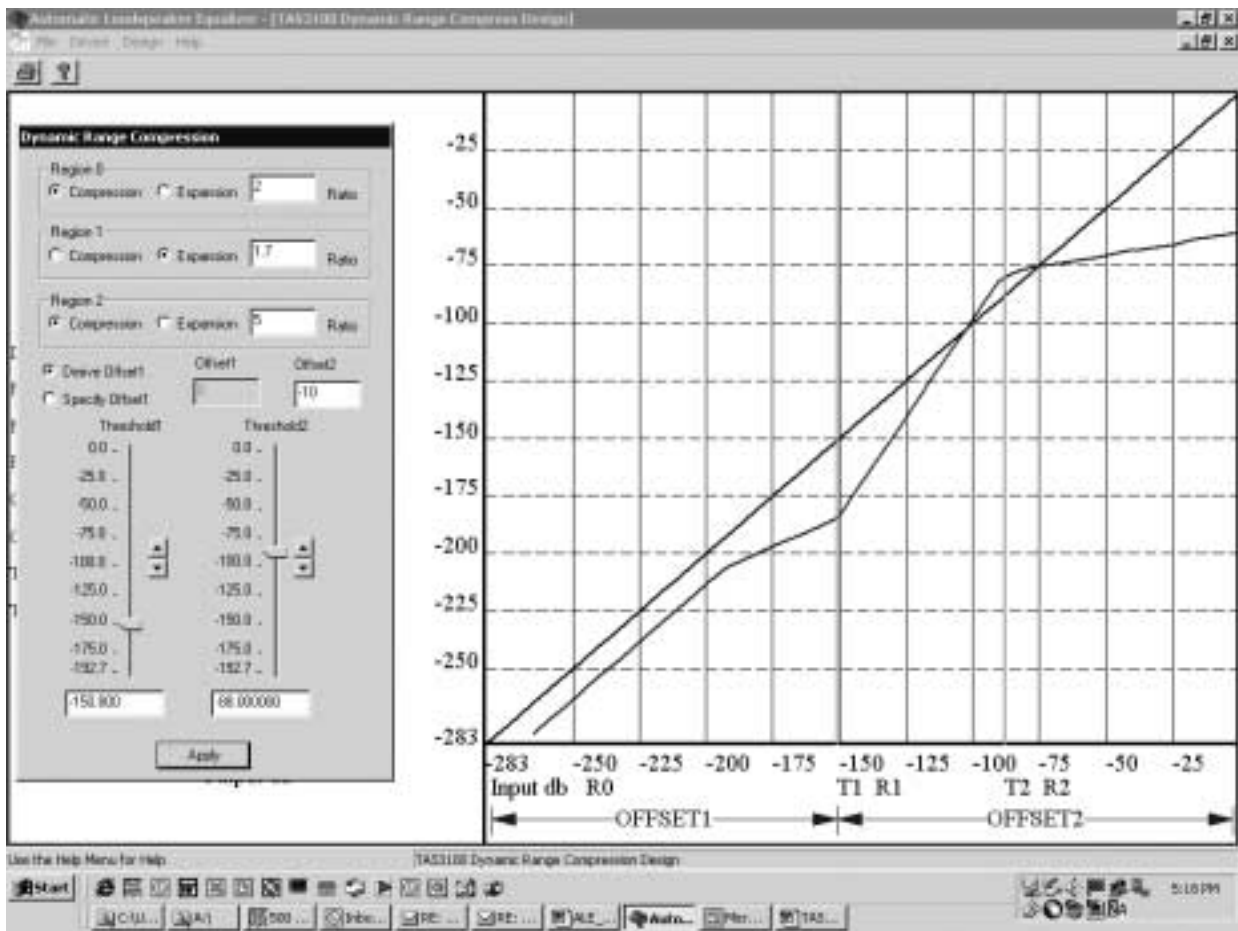


Figure 17: Dynamic Range Compression GUI

The black curve is a 1:1 transfer curve and represents the input level into the DRC. The blue line is the actual transfer curve of the DRC.

DRC Overview

The TAS3103 DRC provides three regions of operation, defined by threshold variables T1 and T2. The two green lines are vertical projections of these two threshold points. The three regions defined by T1 and T2 are referenced as Region 0, Region 1, and Region 2. Region 0 is the far left region, and applies to DRC input levels equal to or below T1. Region 1 is the central or middle region, and applies to DRC input levels equal to or below T2 and above T1. Region 2 is the far right region and applies to DRC input levels above T2.

Within each region, compression (output increases 1 dB for every n dB increase in input) or expansion (output increases n dB for every 1 dB increase in input) can be specified, and the ratio n is programmable. In addition, the offset of the DRC curve from the input curve can be specified at the two threshold points T1 and T2. The offset of the DRC curve from the input curve at threshold T2 is referred to as Offset2, and the offset of the DRC curve from the input curve at threshold T1 is referred to as Offset1.

DRC Parameter Specification / Computation

There are seven variables that define the operation of the DRC:

- Thresholds
 - T1
 - T2
- Compression / Expansion Gains
 - Region 0
 - Region 1
 - Region 2
- Offsets
 - Offset1
 - Offset2

The thresholds T1 and T2 define the three regions of DRC operation. Within each region, compression or expansion can be selected, and the degree of compression or expansion within each region can be set by inputting the appropriate compression / expansion ratio. Offset1 and Offset2 set the boost or cut of the DRC transfer function at the two threshold points T1 and T2, and serve to place or anchor the gain of the transfer function shape set by T1, T2, and the compression / expansion ratios.

DRC Threshold Settings

The threshold settings can be set either by using the slider associated with each threshold point, or by entering values into the two windows set under the two sliders.

However, the threshold points cannot be set independent of the processing gains programmed into the TAS3103. The thresholds T1 and T2 are not referenced to the signal level into the TAS3103, but rather the signal level into the DRC. To set a threshold point at the DRC input relative to the signal levels into the TAS3103, it is necessary to take the processing gain (or loss) between the TAS3103 input port and the DRC. Assume, as an example, the processing gain structure shown in Figure 18. When the audio signal is input into the TAS3103, an 8-bit headroom is established above the MSB of the incoming audio data. In addition, 8 bits of processing resolution are added below the 32 bits reserved for data input word sizes up to 32 bits. The result is a 48-bit word, and this is the word size used in the digital audio processor within the TAS3103. But when the audio data is routed to the DRC, only the upper 32 bits of this 48-bit word are used. This results in a 48 dB attenuation of the signal level into the DRC (8 bits x 6 dB / bit = 48 dB). Channel processing gain and use of the dedicated mixer into the DRC can change this apparent 48 dB attenuation in signal level into the DRC. For example, in the figure on the following page, the 2^4 mixer gain into the DRC, coupled with a net channel gain of 0 dB, changes the net 48 dB attenuation of the signal level into the DRC to a net attenuation of 24 dB. In summary, the ALE set the threshold points for the audio signal levels into the DRC. Signal levels into the TAS3103, the channel gain with the TAS3103, and the dedicated mixer that routes the channel output audio to the DRC must all be comprehended in selecting these threshold settings.

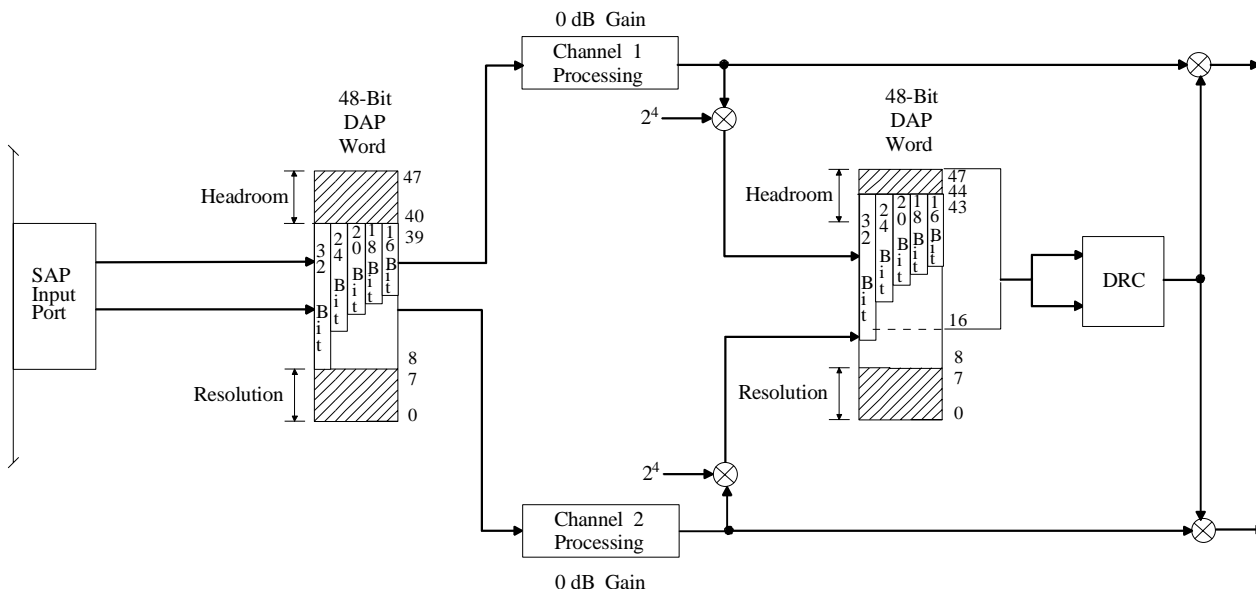


Figure 18: Example of Dependence of DRC Input Signal Level on Channel Gain

DRC Compression / Expansion Settings

Selecting, for each region, either the *Compression* button or the *Expansion* button in the *Dynamic Range Compression* panel, makes the choice of compression or expansion for each of the three regions of the DRC. The value of the ratio must be equal to or greater than 1.0, with a value of 1.0 giving the DRC transfer function a slope equal to the slope of the input curve.

For compression, a ratio of n means that the output level changes 1 dB for every n dB of change on the input. For expansion, a ratio of n means that the output level changes by n dB for every 1 dB the input changes.

DRC Offset Settings

Entering the value desired (in dB) in the *Offset 2* window provided in the *Dynamic Range Compression* panel sets *Offset2*. *Offset1* can be entered the same way, using the *Offset 1* window, or by commanding the tool to automatically compute *Offset1*. Selecting the *Derive Offset 1* button or the *Specify Offset 1* button in the *Dynamic Range Compression* panel makes this choice.

Threshold T_2 serves as the fulcrum or pivot point in the DRC transfer function. There can be no discontinuity in the DRC transfer function at T_2 . This means that the compression and/or expansion plots in regions 1 and 2 must join at threshold T_2 , and the point of intersection must be offset from the input curve by the value *Offset2*. If *Offset2* is positive, it is a cut, and the point of intersection is below the input curve by the value *Offset2*. If *Offset2* is negative, it is a boost, and the point of intersection is above the input curve by the value *Offset2*.

At input level T_1 , the compression and/or expansion plots in regions 0 and 1 are not required to intersect. There is a discontinuity in the DRC transfer function at threshold T_1 if the transfer curve in region 1, which starts at T_2 with a value *Offset2* relative to the input curve, is not offset from the input curve by a value *Offset1* when the plot intersects the vertical green line projection of T_1 . To avoid such a discontinuity at T_1 , *Offset1* must be computed knowing the values of T_1 , T_2 , *Offset2*, and the compression / expansion coefficient k_1 . To relieve users from having to perform this computation, the ALE GUI provides the option of not specifying *Offset1* and having the program automatically compute the *Offset1* value such that there is no

discontinuity at threshold T1. This option is selected by clicking on the *Derive Offset1* button on the *Dynamic Range Compression* panel.

Once all the parameters have been entered, clicking on the *Apply* button enters the data, and the resultant DRC transfer curve is plotted.

DRC Transfer Function Plot

The DRC transfer function for the DRC parameter settings selected can be viewed by clicking on the *Apply* button once all parameters have been set. For the ALE GUI window example previously shown, the settings are as follows:

- Thresholds
 - T1 = -88 dB
 - T2 = -150.8 dB
- Compression / Expansion Gains
 - Region 0 = 2:1 Compression
 - Region 1 = 1.7:1 Expansion
 - Region 2 = 5:1 Compression
- Offsets
 - Offset1 = - 10 dB
 - Offset2 = Automatically Computed

Unlike the 48-bit word size used in the rest of the digital audio processor, the DRC only uses 32-bit precision. The LSB then is at -192 dB (32 bits x -6 dB / bit = -192 dB). This means that for levels below -192 dB, the DRC sees a constant input level of 0, and thus the computed DRC gain coefficient remains fixed at the value computed when the input was at -192 dB. This effect is seen in the example DRC plot.

DRC Transfer Function File

Once the desired DRC transfer function is obtained, the result can be saved to a file by clicking on the *File* button and selecting *Save As DRC2 File*. The directory to store the file, and the file name can be selected, but the file extension must remain *.DR310X*. The content of the *.DR310X* file for the example illustrated is shown in Figure 19. The two threshold values and the two offset values are 48-bit numbers (25.23 format - see Section 3.1.2 in the TAS3103 data manual). The compression / expansion ratios *k* are 28-bit numbers (5.23 format - see Section 3.1.1 in the TAS3103 data manual).

```

T
00 00 00 00
0C 86 0F 2A    {T1}
00 00 00 00
07 4E E8 EF    {T2}
O
00 00 00 00
04 D2 00 4A    {O1}
00 00 00 00
01 2B 65 80    {O2}
k
0F C0 00 00    {k0}
00 59 99 99    {k1}
0F 99 99 9A    {k2}
    
```

Figure 19: ALE DRC File Starter_Example_02.DR310X

The .DR310X extension enables the TAS3103 Device Configuration Tool (DCT) - DAS DCT 2.0 GUI – to identify the file as a DRC file. The DCT DRC panel is shown in Figure 20.



Figure 20: Device Configuration Tool (DCT) Panel

As seen, the DRC file created in the ALE tool can be selected in the DCT tool and downloaded to the specified TAS3103 channel.

In addition to the DRC transfer function, complete specification of the DRC also requires specifying the time constants for the energy estimator and the attack and decay filters. The energy estimator derives an estimate of the rms value of the audio data stream into the DRC. Two programmable parameters, ae and $(1 - ae)$, set the effective time window over which the rms estimate is made in accordance with the equation

$$t_{window} = \frac{1}{F_s \ln(1 - Alpha_e)}$$

The equation only uses $(1 - Alpha_e)$ but the GUI requires that both $Alpha_e$ and $(1 - Alpha_e)$ be entered.

The output of the DRC is a gain coefficient that is used to adjust the level of the audio stream. The Attack & Decay parameters control the transition time of changes in the computed gain coefficient. Four parameters define the operation of the Attack & Decay element. Parameters $Alpha_d$ and $(1 - Alpha_d)$ set the decay or release time constant, and is used when the coefficient change is boosting the volume level. Parameters $Alpha_a$ and $(1 - Alpha_a)$ set the attack time constant, and is used when the coefficient change is cutting the volume level. The time constants are determined by the equations below, where t_a is the attack time constant and t_d is the decay time constant.

$$t_a = \frac{1}{F_s \ln(1 - aa)}$$

$$t_d = \frac{1}{F_s \ln(1 - ad)}$$

Loudness Compensation Filter Design

A listener's perception of the loudness of a given tone relative to other tones at different frequencies is not constant over a range of volume settings. Lower frequency tones are perceived as dropping faster in volume level relative to higher frequency tones as the volume is decreased. This same phenomenon also occurs, to a lesser degree, for higher frequency tones relative to mid-range tones. Typically, loudness compensation involves boosting the bass content, and sometimes the treble content, of the audio as the volume is decreased. The TAS3103 provides both a biquad loudness filter to define the portion of the frequency spectrum (bass or treble) to be adjusted in loudness, and four variables that implement the function $f(\text{volume})$ that controls the volume adjustment. The ALE TAS3100 loudness filter design GUI provides a graphic facility in which to define the parameters of the loudness filter and the four variables that define $f(\text{volume})$.

An excellent paper on loudness compensation is *Loudness Compensation: Use and Abuse* by Tomlinson Holman and Frank Kampmann. This article appeared in the Journal of the Audio Engineering Society, July / August 1978, Volume 26, Number 7 / 8

Loudness Compensation Transfer Function vs Loudness Biquad Filter Transfer Function

A block diagram of the TAS3100 loudness compensation process is shown in Figure 21. It is noted that because of the parallel paths that make up the loudness compensation block, the frequency response of the loudness biquad filter is not the overall frequency response of the loudness compensation block. Typically, a user wants to specify the frequency response of the loudness compensation block itself, and not worry about the topology of the loudness compensation block. The ALE provides this capability. The ALE also provides an option for programming the loudness biquad filter directly.

In Figure 21 the composite loudness compensation transfer function can be seen to be

$$\text{Loudness Compensation Composite Transfer Function} = H(z) = \left[\text{Plot Volume} + H'(z) \times \text{Plot Gain} \right]$$

where $H'(z)$ is the transfer function of the biquad loudness filter defining the portion of the spectrum to be adjusted in volume, Plot Volume is the commanded volume setting, and Plot Gain is the loudness adjustment coefficient as defined by $f(\text{volume})$. Plot Gain is the same as $f(\text{Plot Volume})$.

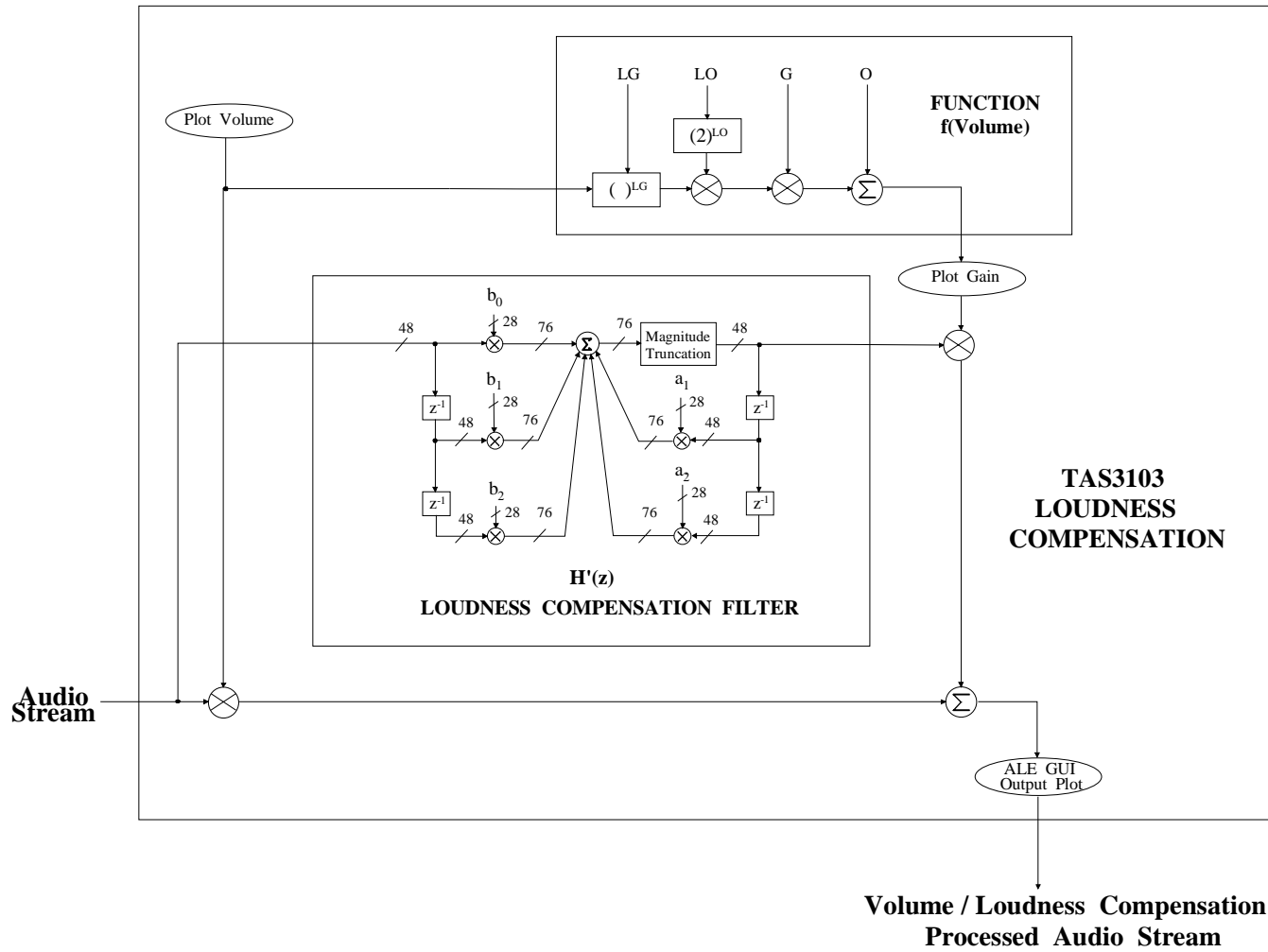


Figure 21: TAS3103 Loudness Processing Block

The transfer function of the biquad filter can be derived as shown in Figure 22.

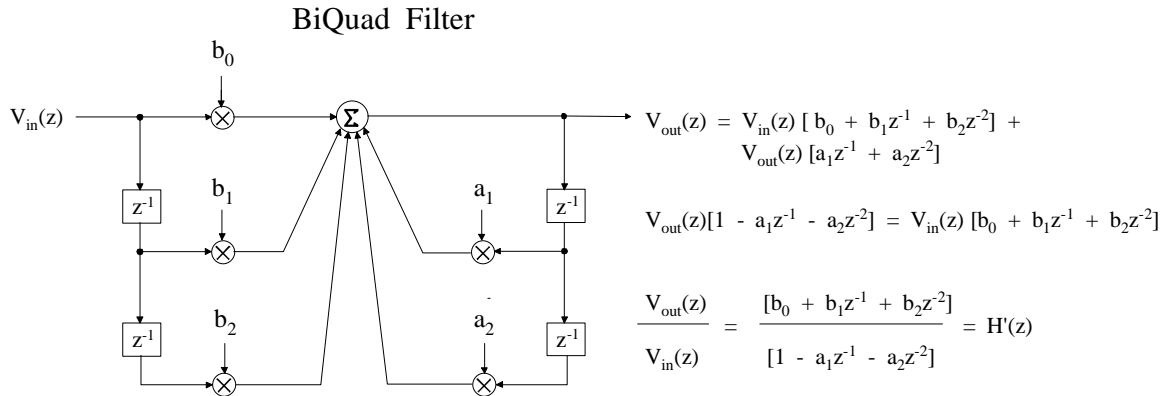


Figure 22: Biquad Filter Transfer Function Derivation

The transfer function $H'(z)$ for the biquad filter can be derived from the biquad equivalent transfer function of $H(z)$ as follows, assuming

Plot Volume) and Plot Gain

$$H(z) = H'(z) + 1$$

$$\therefore H'(z) = H(z) - 1$$

$$H'(z) = \frac{B(z)}{A(z)} - 1 = \frac{B(z) - A(z)}{A(z)}$$

$$H'(z) = \frac{b_0 - 1 + (b_1 + a_1)z^{-1} + (b_2 + a_2)z^{-2}}{1 - a_1 z^{-1} - a_2 z^{-2}}$$

To obtain the biquad coefficients for $H'(z)$, the denominator coefficients of the overall loudness biquad transfer function $H(z)$ are subtracted from the numerator coefficients of the overall loudness biquad transfer function $H(z)$ to form the numerator coefficients for the loudness biquad transfer function $H'(z)$. The denominator coefficients are the same for both transfer functions.

Using ALE To Specify / Design a Loudness Compensation Transfer Function

The EQ filter design GUI is the default GUI appearing when the ALE package is opened. To go to the TAS3100 loudness compensation filter design GUI, first click on the *Device* button and select *TAS3100*. Then click on the *Design* button and select *Design Loudness Filters For TAS3100*. (Note that if the Dynamic Range Compressor GUI is present, the *Design EQ Filters* GUI must first be selected and then the *Design Loudness Filters For TAS3100* GUI selected).

The Loudness Compensation GUI that comes up includes the Create Loudness Filter panel shown in Figure 23. Four options are provided for specifying the Loudness Compensation Transfer Function $H(z)$.

1. Enter user-specific biquad coefficients that directly implement $H(z)$
2. Create $H(z)$ using a EQ (bell-shaped) transfer characteristic
3. Create $H(z)$ using a bass shelf transfer characteristic
4. Create $H(z)$ using a treble shelf transfer characteristic (used if compensation addresses treble)

A fifth option of entering the biquad coefficients for the Loudness Biquad filter $H'(z)$ in the Loudness Compensation Block is also provided.

Specifying $H(z)$ Using Predetermined Biquad Coefficients

The tab Base Type Coefficients on the Create Loudness Filter panel is used when the equivalent biquad coefficients for the composite loudness transfer function $H(z)$ have been determined by independent means.

To illustrate the use of the Base Type Coefficients tab, click on the *Base Type Coefficients* tab to bring up the panel shown in Figure 24. For this example, the equivalent biquad coefficients for the composite loudness compensation transfer function $H(z)$ are assumed to have been independently calculated, assuming a 44.1-kHz sampling frequency, to be:

$$\begin{aligned} a_1 &= 00\ FE\ 2C\ 33 \Rightarrow 1.985723853 \\ a_2 &= FF\ 81\ CF\ 93 \Rightarrow -0.9858528375 \\ b_0 &= 00\ 80\ 5F\ 9E \Rightarrow 1.0029180049 \\ b_1 &= FF\ 01\ D3\ CD \Rightarrow -1.985723853 \\ b_2 &= 00\ 7D\ D0\ CF \Rightarrow 0.9829348325 \end{aligned}$$

These coefficients have been entered, in decimal, in the coefficient windows provided in the panel. Figure 24 indicates which window references which coefficient. **Note that it is a requirement that the negative values of coefficients a_1 and a_2 be entered.**

Caution: The panel in Figure 24 also implies that the coefficients can be entered as hexadecimal numbers. This option is not functional, and all work must remain in the decimal domain. However, the ALE correctly outputs the final coefficient values in the hexadecimal format when creating the output file xx.LD310X.

When the coefficients have been entered, clicking on the *OK* button in the *Create Loudness Filter* panel results in a plot of the overall loudness compensation transfer function. This plot is shown in Figure 25, and as can be seen, it is nothing more than a bell-shaped EQ filter with the following parameters:

$$\begin{aligned} \text{Gain} &= 3\ \text{dB} \\ \text{Bandwidth} &= 100\ \text{Hz} \\ \text{Center Frequency} &= 80\ \text{Hz} \end{aligned}$$

Should the full lower frequency range not appear on the plot of Figure 25, it is necessary to go back to the Design EQ Filters GUI by clicking on *Design* and selecting *Design EQ Filters*. Then click on *File*, select *Open Raw Response File* and open *zerodb.txt*. This step activates the Automatic Loudspeaker Equalizer GUI. Then click on *View*, select *Settings*, click on the *Frequency Limits* tab, and enter the lower frequency limit desired in the *LF Lower Frequency Limits Hz* window. Then clicking on *Design* and selecting *Loudness Filters for TAS3100I* brings back the Loudness filter Design GUI, with the plot now extended down to the lower frequency limit selected. Also, while still in the Automatic Loudspeaker Equalizer GUI, the sampling rate can also be changed by clicking on *View*, selecting *Settings*, clicking on the *Frequency Limits* tab, and entering the desired sampling frequency in Hz in the *Fs Sampling Frequency Hz* window.

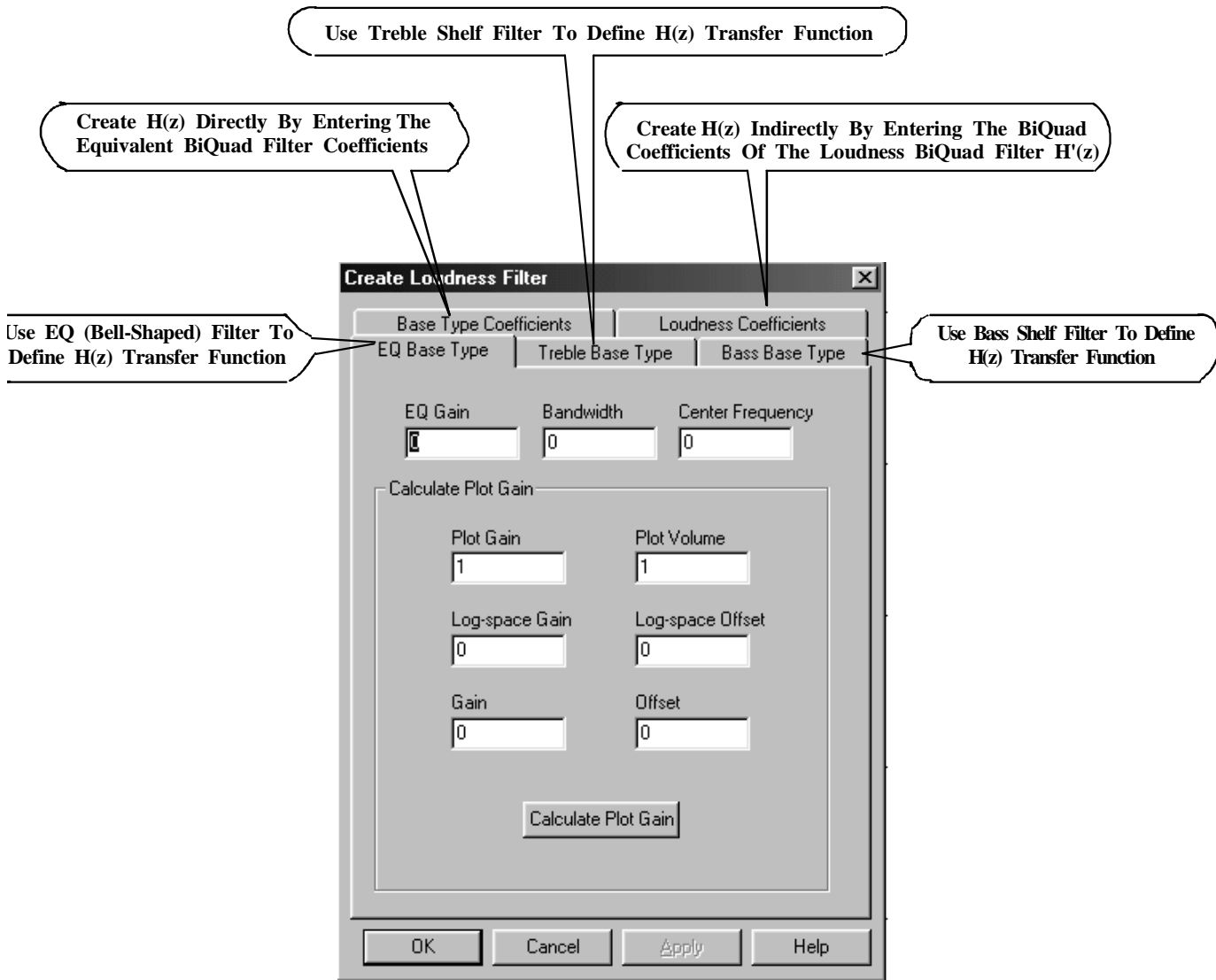


Figure 23: Create Loudness Filter Panel

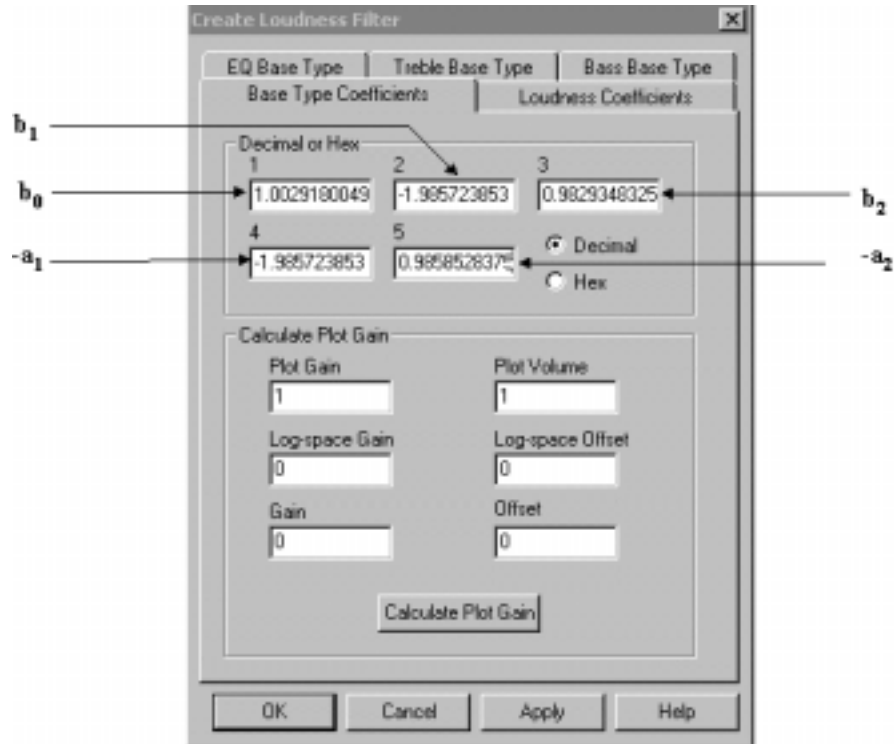


Figure 24: Base Type Coefficients Loudness Panel

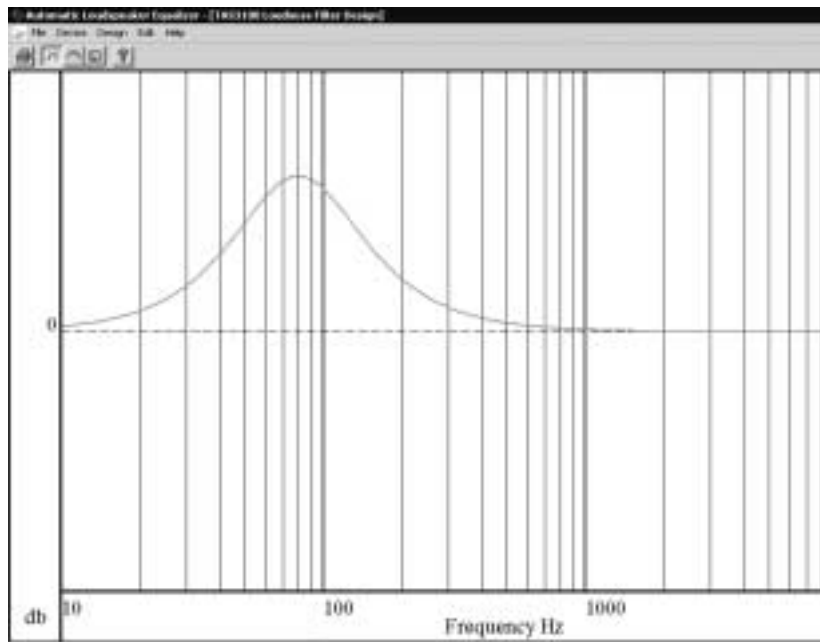


Figure 25: Loudness Compensation Transfer Function - H(z) Coefficients Entered Directly

If it is desired to edit the coefficients for $H(z)$, click on *Edit* and select *Select Filters To Edit Graphically*. Then click on the generated curve of the transfer function. The Create Loudness Filter panel reappears. Then click on the *Base Type Coefficients* tab. The entered coefficients reappear and can be altered as required. Clicking on the *OK* button results in a second curve being generated. If this second curve is the desired response, clicking on *Edit*, selecting *Delete Filters Graphically*, and then clicking on the first curve generated eliminates the first curve, leaving only the desired response graphed.

When the response desired has been successfully plotted, TAS3103 biquad filter coefficients must next be generated. Clicking on *Edit*, selecting *Select Filter to Add to TAS3XXX Data File Graphically*, and clicking on the desired response curve accomplishes this. The filter coefficients are now generated. Clicking on *File* and selecting *Save as LD31 Data K...* brings up a Save As panel. The file name and folder can be selected, but the extension LD310X must be retained. This extension allows the TAS3103 GUI to recognize the file as a loudness compensation file. The content of the file generated is shown in Figure 26, with the coefficient identifiers off to the side added for clarity.

```
L
00 FE 2C 32 => a1 = 1.985723733
FF 81 CF 94 => a2 = -0.9858527183
00 00 5F 9D => b0 = 0.0029178857
00 00 00 00 => b1 = 0.0
FF FF A0 62 => b2 = -0.0029178857
```

Figure 26: Loudness Compensation File For Predetermined $H(z)$ Coefficients

It is noted that the coefficients in Figure 26 are not the same biquad coefficients that were entered for $H(z)$. In fact, the coefficients are the biquad coefficients for $H'(z)$; ALE automatically computes these biquad coefficients so that they can be directly downloaded into the TAS3103 without requiring any further mapping of coefficients.

Specifying $H(z)$ Using EQ (Bell-Shaped) Filters

$H(z)$ can also be specified by selecting the type filter desired for the composite loudness compensation function. Using EQ or bell-shaped filters is one of the options. To use EQ filters to specify $H(z)$, select the *EQ Base Type* tab in the Create Loudness Filter panel shown in Figure 23. The panel shown in Figure 27 appears.

To illustrate the use of the panel, enter the filter parameters implemented by the $H(z)$ biquad coefficients used in the previous example, which are:

```
Gain in dB = 3 dB
Bandwidth = 100 Hz
Center Frequency = 80 Hz
```

These parameters are entered by using the appropriate windows of the panel associated with the EQ Base Type selection. Having entered the parameters, clicking on *OK* plots the $H(z)$ transfer function resulting from these gain, bandwidth, and center frequency selections. The plot obtained using the previous parameters is identical to the plot shown in Figure 25.

If the response desired is the response plotted, TAS3103 biquad filter coefficients can be generated by clicking on *Edit*, selecting *Select Filter to Add to TAS3XXX Data File Graphically*, and clicking on the desired response curve. Clicking on *File* and selecting *Save as LD31 Data K...* saves the data, once a file name and directory are entered.



Figure 27: EQ Base Type Loudness Panel

Specifying $H(z)$ Using Bass Shelf Filters

To use Bass Shelf filters to specify $H(z)$, select the *Bass Base Type* tab in the Create Loudness Filter panel shown in Figure 23. The panel shown in Figure 28 appears.

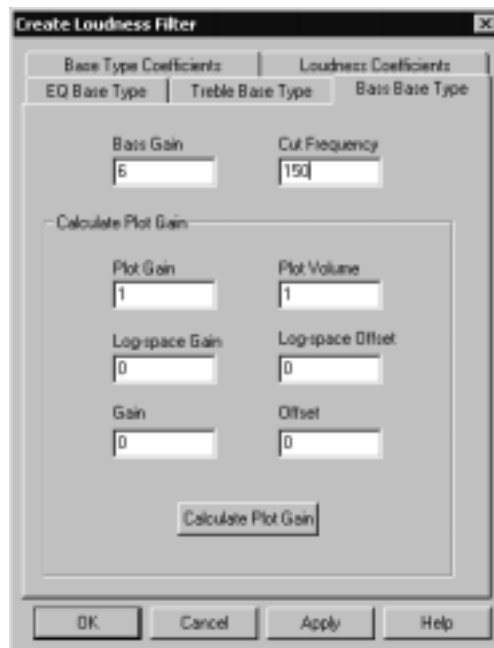


Figure 28: Bass Base Type Loudness Panel

The response to the parameters entered in Figure 28 – Bass Gain = 6 dB and Bass Cut (or Boost in this example) Frequency = 150 Hz – is shown in Figure 29.

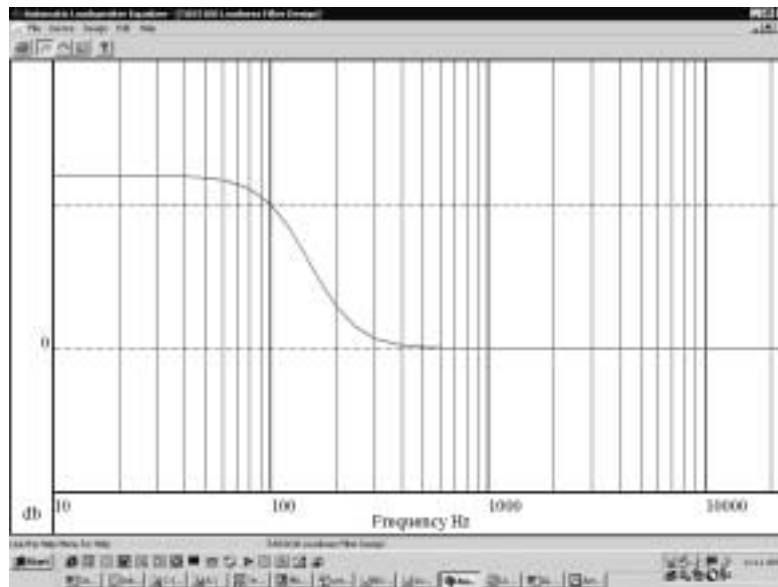


Figure 29: Loudness Compensation Transfer Function - H(z) Created Using Bass Filter

Clicking on *Edit*, selecting *Select Filter to Add to TAS3XXX Data File Graphically*, and clicking on the desired response curve generates TAS3103 biquad filter coefficients for the Bass Filter response. Clicking on *File* and selecting *Save as LD31 Data K...* saves the data, once a file name and directory are entered.

Specifying H(z) Using Treble Shelf Filters

To use Treble Shelf filters to specify H(z), select the *Treble Base Type* tab in the Create Loudness Filter panel shown in Figure 23. The panel shown in Figure 30 appears. The response to the parameters entered in Figure 30 – Treble Gain = -4 dB and Bass Cut Frequency = 2000 Hz – is shown in Figure 31.

Clicking on *Edit*, selecting *Select Filter to Add to TAS3XXX Data File Graphically*, and clicking on the desired response curve generates TAS3103 biquad filter coefficients for the Treble Filter response. Clicking on *File* and selecting *Save as LD31 Data K...* saves the data, once a file name and directory are entered.



Figure 30: Treble Base Type Loudness Panel

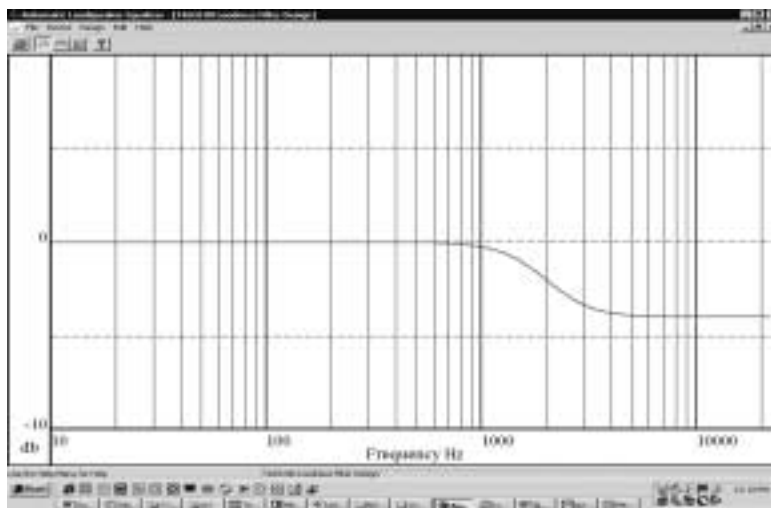
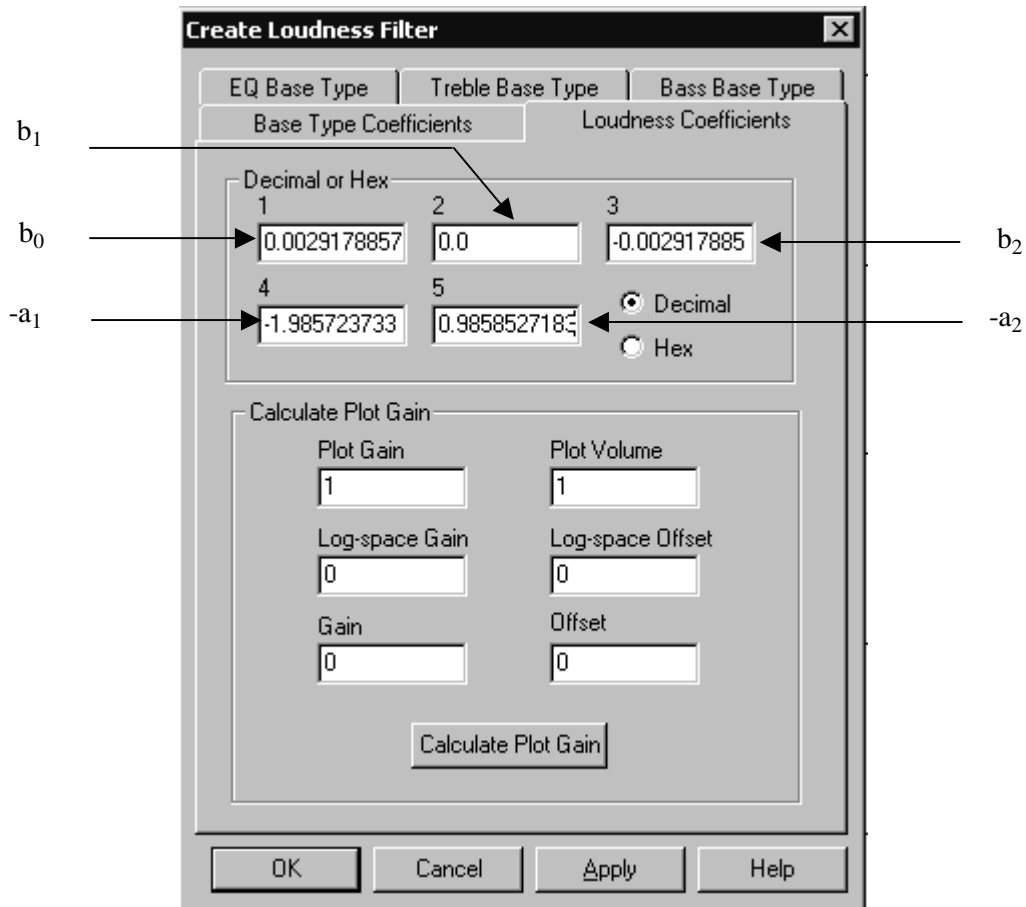


Figure 31: Loudness Compensation Transfer Function - H(z) Created Using Treble Filter

Specifying $H'(z)$ (The Loudness Biquad Filter) Using Predetermined Biquad Coefficients

There may be cases when it is desired to specify the composite loudness compensation transfer function by using predetermined coefficients for the loudness biquad filter. Selecting the *Loudness Coefficients* tab in the *Create Loudness Filter* panel shown in Figure 23 provides this capability. Upon selecting this tab, the panel shown in Figure 32 appears.



Create Loudness Filter

EQ Base Type | Treble Base Type | Bass Base Type

Base Type Coefficients | Loudness Coefficients

Decimal or Hex

1: 0.0029178857 (labeled b_1)

2: 0.0

3: -0.002917885 (labeled b_2)

4: -1.985723733 (labeled $-a_1$)

5: 0.985852718 (labeled $-a_2$)

Decimal
 Hex

Calculate Plot Gain

Plot Gain: 1

Plot Volume: 1

Log-space Gain: 0

Log-space Offset: 0

Gain: 0

Offset: 0

Calculate Plot Gain

OK | Cancel | Apply | Help

Figure 32: Loudness Coefficients Loudness Panel

Figure 26 lists the loudness biquad coefficients obtained by entering predetermined coefficients for the composite loudness compensation transfer function $H(z)$. Figure 32 shows these same coefficients entered in the Loudness Coefficients tab panel. Figure 33 shows the resultant response of entering these coefficients. The red curve is the overall loudness compensation response $H(z)$ resulting from entering the coefficients shown in Figure 32 and listed in Figure 26, and is the expected bell-shaped response. The green curve is the response of $H'(z)$, the loudness biquad filter itself, and is shown to emphasize that $H(z)$ and $H'(z)$ are not the same.

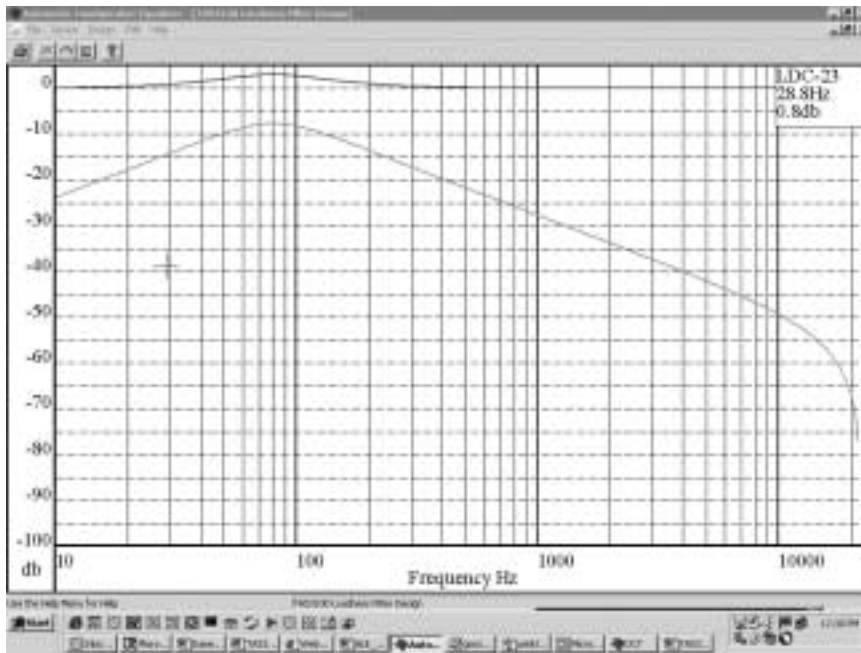


Figure 33: Loudness Compensation Transfer Function - H(z) Created Using Loudness Coefficients

Loudness Compensation vs Volume Command

In addition to specifying the composite loudness transfer function $H(z)$, the programmable loudness function block $f(\text{volume_setting})$ must also be specified to complete a loudness compensation design. The files generated by ALE do not include any of the parameters that govern $f(\text{volume_setting})$, but ALE does provide the ability to specify these parameters for display purposes. With this ability, a family of curves for the filter design chosen can be plotted as a function of volume setting, allowing a user to directly visualize the loudness compensation achieved for a given loudness filter design.

The output of the function block $f(\text{volume_setting})$ can be expressed in terms of the four I²C programmable loudness compensation coefficients as:

$$f(\text{volume_setting}) = \left[(\text{volume_setting})^{LG} * 2^{LO} * G \right] + O$$

where:

- LG = logarithmic gain
- LO = logarithmic offset
- G = gain
- O = offset

If LG = 0.5, LO = 0.0, G = 1.0, and O = 0.0, for example, $f(\text{volume_setting})$ becomes

$$f(\text{volume_setting}) = \sqrt{\text{volume_setting}}$$

As noted in Figure 21, the output of $f(\text{volume_setting})$, labeled **Plot Gain** in Figure 21, feeds a gain-control mixer. The output of the loudness biquad filter $H'(z)$ also feeds the same gain-control mixer. The output of this gain-control mixer is then summed with the volume-adjusted audio data stream. The net result is loudness-adjusted audio. Loudness compensation then allows a given spectral segment of the audio data stream (as determined by the loudness biquad filter $H'(z)$) to be given a delta adjustment in volume as determined by the programmable function $f(\text{volume_setting})$.

The output of the soft volume / loudness compensation block for the previous example of $LG = 0.5$, $LO = 0.0$, $G = 1.0$, and $O = 0.0$ is

$$\text{Audio}_{\text{Out}} = \text{Audio}_{\text{In}} * \left[\text{volume_setting} + H'(z) * \sqrt{\text{volume_setting}} \right]$$

Figure 34 shows the windows for entering LG , LO , G , and O in a given Create Loudness Filter panel.

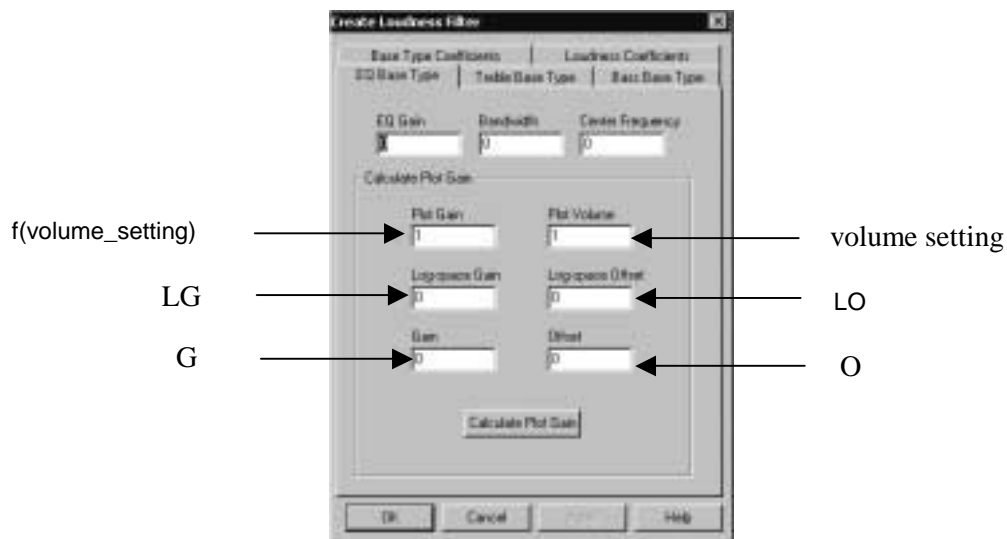
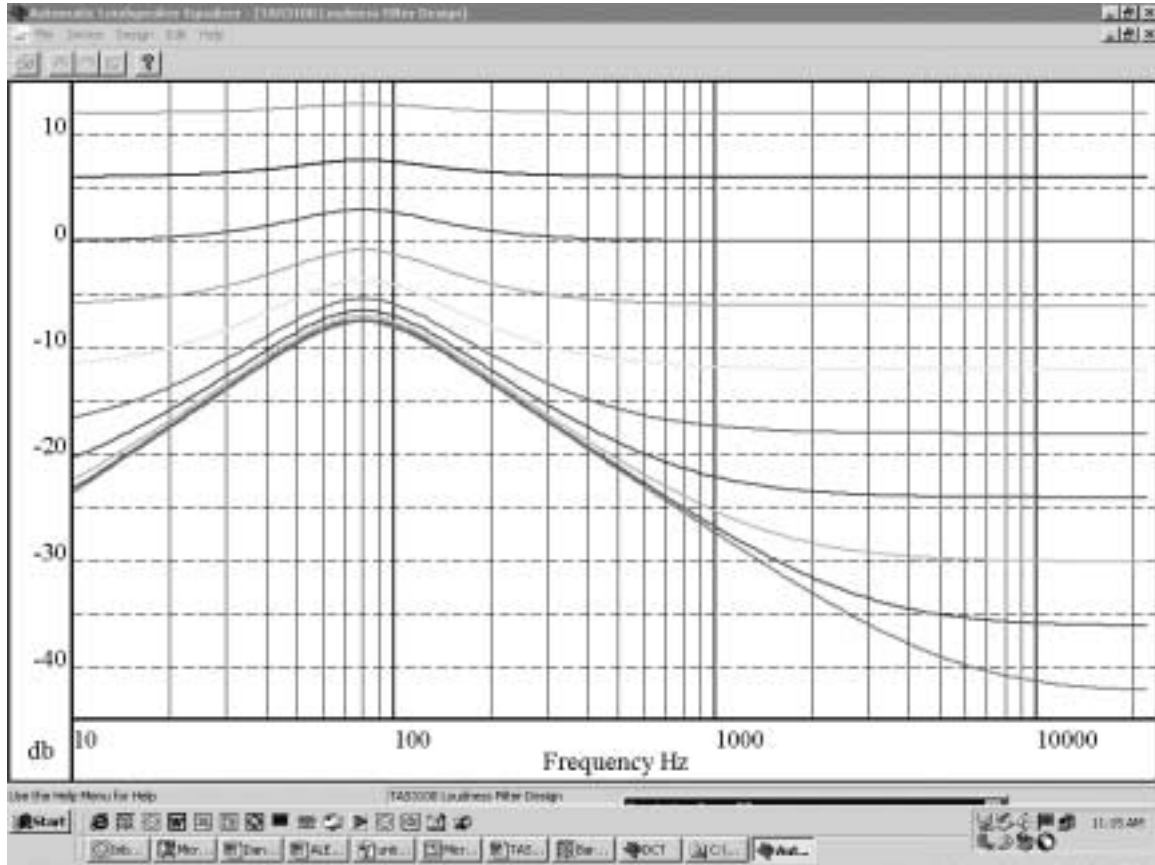


Figure 34: Programming Parameters For $f(\text{volume_setting})$

The *Plot Volume* window is used to enter the commanded volume setting. The *Plot Gain* window is the output of the function $f(\text{volume_setting})$. Performance plots are created by entering the commanded volume setting in the *Plot Volume* window and clicking on the *Apply* button. The computed value of $f(\text{volume_setting})$ can be seen in the *Plot Gain* window by clicking on the *Calculate Plot Gain* window.

Some loudness compensation schemes apply a fixed loudness adjustment coefficient to a range of volume settings. In this application, when the volume setting crosses a volume range boundary, a controller issues a new fixed loudness adjustment coefficient. The TAS3103 and the ALE both support this implementation of loudness compensation. In the TAS3103, the computation of $f(\text{volume_setting})$ can be turned off by entering a value of 0 for G . The loudness adjustment coefficient, for this case, is implemented by writing the value the coefficient to the I^2C -parameter O (refer to Figure 21). In the ALE, computation of $f(\text{volume_setting})$ can be turned off in the very same way by entering a value of 0 for G . Then entering the desired fixed loudness adjustment coefficient in O , the desired volume command in *Plot Volume*, clicking on the *Calculate Plot Gain* button, and then clicking on the *Apply* button yields a resultant plot for the volume command and loudness adjustment coefficient entered.

Figure 35 shows a suite of plots of volume_setting versus the $f(\text{volume_setting})$ function $\sqrt{\text{volume_setting}}$ for LG = 0.5, LO = 0.0, G = 1.0, and O = 0.0. It is noted that as the commanded volume setting decreases in value, the amount of bass boost relative to the rest of the audio spectrum increases. This phenomenon is the purpose of loudness compensation.



Volume Setting	Plot Color
4.0	Green
2.0	Blue
1.0	Red
0.5	Orange
0.25	Yellow

Volume Setting	Plot Color
0.125	Neon Pink
0.0625	Dark Green
0.03125	Light Blue
0.015625	Blue Green
0.0078125	Violet

Figure 35: Loudness Compensation Plots – Commanded Volume vs Output

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