General Digital Filters Library

User Reference Manual

56800E
Digital Signal Controller

56800E_GDFLIB
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The following revision history table summarizes changes contained in this document.

Table 0-1. Revision History

Date	Revision Label	Description
	0	Initial release
	1	Reformatted and updated revision
	2	FSLESL 2.0

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Chapter 2 INTRODUCTION

2.1 Overview

This reference manual describes General Digital Filters Library for Freescale 56800E family of Digital Signal Controllers. This library contains optimized functions for 56800E family of controllers. The library is supplied in a binary form, which is unique by its simplicity to integrate with user application.

2.2 Supported Compilers

General Digital Filters Library (GDFLIB) is written in assembly language with a C-callable interface. The library was built and tested using the following compiler:

- CodeWarrior™ Development Studio for Freescale™ DSC56800/E Digital Signal Controllers, version 8.3

The library is delivered in the *56800E_GDFLIB.lib* library module. The interfaces to the algorithms included in this library have been combined into a single public interface include file, the *gdflib.h*. This was done to reduce the number of files required for inclusion by the application programs. Refer to the specific algorithm sections of this document for details on the software application programming interface (API), defined and functionality provided for the individual algorithms.

2.3 Installation

If the user wants to fully use this library, the CodeWarrior tools should be installed prior to General Digital Filters Library. In case that General Digital Filters Library tool is installed while CodeWarrior is not present, users can only browse the installed software package, but will not be able to build, download, and run the code. The installation itself consists of copying the required files to the destination hard drive, checking the presence of CodeWarrior, and creating the shortcut under the Start->Programs menu.

Each General Digital Filters Library release is installed in its own new folder, named *56800E_GDFLIB_rX.X*, where *X.X* denotes the actual release number. This way of library installation allows the users to maintain older releases and projects and gives them a free choice to select the active library release.

To start the installation process, follow the following steps:

1. Execute the *56800E_FSLESL_rXX.exe* file.
2. Follow the FSLESL software installation instructions on your screen.

2.4 Library Integration

The General Digital Filters Library is added into a new CodeWarrior project by taking the following steps:

1. Create a new empty project.
2. Create *GDFLIB* group in your new open project. Note that this step is not mandatory, it is mentioned here just for the purpose of maintaining file consistency in the CodeWarrior project window. In the CodeWarrior menu, choose Project > Create Group..., type GDFLIB into the dialog window that pops up, and click <OK>.
3. Refer the *56800E_GDFLIB.lib* file in the project window. This can be achieved by dragging the library file from the proper library subfolder and dropping it into the *GDFLIB* group in the CodeWarrior project window. This step will automatically add the *GDFLIB* path into the project access paths, such as the user can take advantage of the library functions to achieve flawless project compilation and linking.
4. It is similar with the reference file *gdflib.h*. This file can be dragged from the proper library subfolder and dropped into the GDFLIB group in the CodeWarrior project window.
5. The following program line must be added into the user-application source code in order to use the library functions.

```
#include "gdflib.h"
```

2.5 API Definition

The description of each function described in this General Digital Filters Library user reference manual consists of a number of subsections:

Synopsis

This subsection gives the header files that should be included within a source file that references the function or macro. It also shows an appropriate declaration for the function or for a function that can be substituted by a macro. This declaration is not included in your program; only the header file(s) should be included.

Prototype

This subsection shows the original function prototype declaration with all its arguments.

Arguments

This optional subsection describes input arguments to a function or macro.

Description

This subsection is a description of the function or macro. It explains algorithms being used by functions or macros.

Return

This optional subsection describes the return value (if any) of the function or macro.

Range Issues

This optional subsection specifies the ranges of input variables.

Special Issues

This optional subsection specifies special assumptions that are mandatory for correct function calculation; for example saturation, rounding, and so on.

Implementation

This optional subsection specifies, whether a call of the function generates a library function call or a macro expansion.
This subsection also consists of one or more examples of the use of the function. The examples are often fragments of code (not completed programs) for illustration purposes.

See Also

This optional subsection provides a list of related functions or macros.

Performance

This section specifies the actual requirements of the function or macro in terms of required code memory, data memory, and number of clock cycles to execute. If the clock cycles have two numbers for instance 21/22, then the number 21 is measured on the MCF56F80xx core and the number 22 is measured on the MCF56F83xx core.

2.6 Data Types

The 16-bit DSC core supports four types of two's-complement data formats:

- Signed integer
- Unsigned integer
- Signed fractional
- Unsigned fractional

Signed and unsigned integer data types are useful for general-purpose computation; they are familiar with the microprocessor and microcontroller programmers. Fractional data types allow powerful numeric and digital-signal-processing algorithms to be implemented.

2.6.1 Signed Integer (SI)

This format is used for processing data as integers. In this format, the N-bit operand is represented using the N.0 format (N integer bits). The signed integer numbers lie in the following range:

$$-2^{[N-1]} \le SI \le [2^{[N-1]} - 1]$$ **Eqn. 2-1**

This data format is available for bytes, words, and longs. The most negative, signed word that can be represented is –32,768 ($8000), and the most negative, signed long word is –2,147,483,648 ($80000000).

The most positive, signed word is 32,767 ($7FFF), and the most positive signed long word is 2,147,483,647 ($7FFFFFFF).

2.6.2 Unsigned Integer (UI)

The unsigned integer numbers are positive only, and they have nearly twice the magnitude of a signed number of the same size. The unsigned integer numbers lie in the following range:

$$0 \le UI \le [2^{[N-1]} - 1]$$ **Eqn. 2-2**

The binary word is interpreted as having a binary point immediately to the right of the integer's least significant bit. This data format is available for bytes, words, and long words. The most positive, 16-bit, unsigned integer is 65,535 ($FFFF), and the most positive, 32-bit, unsigned integer is 4,294,967,295 ($FFFFFFFF). The smallest unsigned integer number is zero ($0000), regardless of size.

2.6.3 Signed Fractional (SF)

In this format, the N-bit operand is represented using the 1.[N–1] format (one sign bit, N–1 fractional bits). The signed fractional numbers lie in the following range:

$$-1,0 \le SF \le 1,0 - 2^{-[N-1]}$$ **Eqn. 2-3**

This data format is available for words and long words. For both word and long-word signed fractions, the most negative number that can be represented is –1.0; its internal representation is $8000 (word) or $80000000 (long word). The most positive word is $7FFF ($1.0 - 2^{-15}$); its most positive long word is $7FFFFFFF ($1.0 - 2^{-31}$).

2.6.4 Unsigned Fractional (UF)

The unsigned fractional numbers can be positive only, and they have nearly twice the magnitude of a signed number with the same number of bits. The unsigned fractional numbers lie in the following range:

$$0,0 \le UF \le 2,0 - 2^{-[N-1]}$$ **Eqn. 2-4**

The binary word is interpreted as having a binary point after the MSB. This data format is available for words and longs. The most positive, 16-bit, unsigned number is $FFFF, or $\{1.0 + (1.0 - 2^{-[N-1]})\} = 1.99997$. The smallest unsigned fractional number is zero ($0000).

2.7 User Common Types

Table 2-1. User-Defined Typedefs in *56800E_types.h*

Mnemonics	Size — bits	Description
Word8	8	To represent 8-bit signed variable/value.
UWord8	8	To represent 16-bit unsigned variable/value.
Word16	16	To represent 16-bit signed variable/value.
UWord16	16	To represent 16-bit unsigned variable/value.
Word32	32	To represent 32-bit signed variable/value.
UWord32	32	To represent 16-bit unsigned variable/value.
Int8	8	To represent 8-bit signed variable/value.
UInt8	8	To represent 16-bit unsigned variable/value.
Int16	16	To represent 16-bit signed variable/value.
UInt16	16	To represent 16-bit unsigned variable/value.
Int32	32	To represent 32-bit signed variable/value.
UInt32	32	To represent 16-bit unsigned variable/value.
Frac16	16	To represent 16-bit signed variable/value.
Frac32	32	To represent 32-bit signed variable/value.
NULL	constant	Represents NULL pointer.
bool	16	Boolean variable.
false	constant	Represents false value.
true	constant	Represents true value.
FRAC16()	macro	Transforms float value from <–1, 1) range into fractional representation <–32768, 32767>.
FRAC32()	macro	Transforms float value from <–1, 1) range into fractional representation <–2147483648, 2147483648>.

2.8 Special Issues

All functions in the General Digital Filters Library are implemented without storing any of the volatile registers (refer to the compiler manual) used by the respective routine. Only non-volatile registers (C10, D10, R5) are saved by pushing the registers on the stack. Therefore, if the particular registers initialized before the library function call are to be used after the function call, it is necessary to save them manually.

Chapter 3 FUNCTION API

3.1 API Summary

Table 3-1. API Functions Summary

Name	Arguments	Output	Description
GDFLIB_FilterIIR1Init	GDFLIB_FILTER_IIR1_T *pudtFilter	Void	The function initializes internal variables of a first order IIR filter.
GDFLIB_FilterIIR1	Frac16 f16In GDFLIB_FILTER_IIR1_T *pudtFilter	Frac16	The function calculates first order Direct Form 1 IIR filter.
GDFLIB_FilterIIR2Init	GDFLIB_FILTER_IIR2_T *pudtFilter	Void	The function initializes internal variables of a second order IIR filter.
GDFLIB_FilterIIR2	Frac16 f16In GDFLIB_FILTER_IIR2_T *pudtFilter	Frac16	The function calculates second order Direct Form 1 IIR filter.
GDFLIB_FilterMA32Init	GDFLIB_FILTER_MA32_T *pudtFilter	Void	The function initializes internal variables of a moving average filter.
GDFLIB_FilterMA32	Frac16 f16In GDFLIB_FILTER_MA32_T *pudtFilter	Frac16	The function calculates recursive form of an average filter.

3.2 GDFLIB_FilterIIR1Init

This function initializes the internal variables of a first order IIR filter.

3.2.1 Synopsis

```
#include "gdflib.h"
void GDFLIB_FilterIIR1Init(GDFLIB_FILTER_IIR1_T *pudtFilter)
```

3.2.2 Prototype

```
void GDFLIB_FilterIIR1InitFC(GDFLIB_FILTER_IIR1_T * const pudtFilter)
```

3.2.3 Arguments

Table 3-2. Function Arguments

Name	In/Out	Format	Range	Description
*pudtFilter	In/Out	N/A	N/A	Pointer to a filter structure, which contains filter coefficients and filter buffer; the GDFLIB_FILTER_IIR1_T data type is defined in header file GDFLIB_FilterIIRasm.h.

Table 3-3. User-Type Definitions

Typedef	Name	In/Out	Format	Range	Description
GDFLIB_FILTER_IIR1_T	udtFiltCoeff	In	N/A	N/A	Structure containing the filter coefficients
	f16FiltBufferX[2]	In/Out	SF16	$8000... $7FFF	Filter buffer storing input values
	f32FiltBufferY[2]	In/Out	SF32	$80000000... $7FFFFFFF	Filter buffer storing output values

Table 3-4. User-Type Definitions

Typedef	Name	In/Out	Format	Range	Description
GDFLIB_FILTER_IIR_COEFF1_T	f16B1	In	SF16	$8000... $7FFF	B1 coefficient of the filter
	f16B2	In	SF16	$8000... $7FFF	B2 coefficient of the filter
	f16A2	In	SF16	$8000... $7FFF	A2 coefficient of the filter

3.2.4 Availability

This library module is available in the ANSI C format.

This library module is targeted for the DSC 56F80xx platform.

3.2.5 Dependencies

List of all dependent files:

- GDFLIB_FilterIIRasm.h
- GDFLIB_types.h

3.2.6 Description

The **GDFLIB_FilterIIR1Init** function initializes the buffer and coefficients of the first order IIR filter. This function is called once, during the variable initialization, and since it clears the filter buffer, it must not be called together with the filter-calculation function.

3.2.7 Returns

This function initializes the filter structure pointed to by the pudtFilter pointer.

3.2.8 Range Issues

The filter coefficients must be defined prior to this function call. If the Matlab filter-design toolbox is used for the filter coefficients calculation, then all calculated coefficients must be divided by 2.0 in order to avoid saturation during filter calculation.

3.2.9 Special Issues

The function **GDFLIB_FilterIIR1Init** is the saturation mode independent.

3.2.10 Implementation

The **GDFLIB_FilterIIR1Init** function is implemented as a function call.

Example 3-1. Implementation Code

```
#include "gdflib.h"

static Frac16 mf16Value;
static Frac16 mf16FilteredValue;
static GDFLIB_FILTER_IIR1_T mudtFilterIIR1 = GDFLIB_FILTER_IIR1_DEFAULT;


void Isr(void);
```

```
void main(void)
{
         /* LPF 1st order butterworth 100Hz, Ts = 100us*/
         mudtFilterIIR1.udtFiltCoeff.f16B1 = FRAC16(0.0305 / (2.0));
         mudtFilterIIR1.udtFiltCoeff.f16B2 = FRAC16(0.0305 / (2.0));
         mudtFilterIIR1.udtFiltCoeff.f16A2 = FRAC16(-0.9391 / (2.0));

         /* Filter initialization */
         GDFLIB_FilterIIR1Init(&mudtFilterIIR1);
}

/* Periodical function or interrupt */
void Isr(void)
{
         /* Filter calculation */
         mf16FilteredValue = GDFLIB_FilterIIR1(mf16Value,
&mudtFilterIIR1);
}
```

3.2.11 Performance

Table 3-5. Performance of the GDFLIB_FilterIIR1Init Function

Code Size (bytes)	9	
Data Size (bytes)	0	
Execution Clock	Min	22
	Max	22

3.3 GDFLIB_FilterIIR1

This function calculates the first-order direct form one IIR filter.

3.3.1 Synopsis

```
#include "gdflib.h"
Frac16 GDFLIB_FilterIIR1(Frac16 f16In, GDFLIB_FILTER_IIR1_T *pudtFilter)
```

3.3.2 Prototype

```
asm Frac16 GDFLIB_FilterIIR1FAsm(Frac16 f16In, GDFLIB_FILTER_IIR1_T *
const pudtFilter)
```

3.3.3 Arguments

This subsection describes the input/output arguments to a function or a macro. It explains the algorithms being used by the functions or macro.

Table 3-6. Function Arguments

Name	In/Out	Format	Range	Description
f16In	In	SF16	0x8000... 0x7FFF	input signal to be filtered
*pudtFilter	In/Out	N/A	N/A	Pointer to a filter structure, which contains filter coefficients and filter buffer; the GDFLIB_FILTER_IIR_COEFF1_T data type is defined in header file GDFLIB_FilterIIRasm.h.

Table 3-7. User-Type Definitions

Typedef	Name	In/Out	Format	Range	Description
GDFLIB_FILTER_IIR1_T	udtFiltCoeff	In	N/A	N/A	structure containing filter coefficients
	f16FiltBufferX[2]	In/Out	SF16	0x8000... 0x7FFF	filter buffer storing input values
	f32FiltBufferY[2]	In/Out	SF32	0x80000000... 0x7FFFFFFF	filter buffer storing output values

Table 3-8. User-Type Definitions

Typedef	Name	In/ Out	Format	Range	Description
GDFLIB_FILTER_IIR_COEFF1_T	f16B1	In	SF16	0x8000... 0x7FFF	b1 coefficient of the filter
	f16B2	In	SF16	0x8000... 0x7FFF	b2 coefficient of the filter
	f16A2	In	SF16	0x8000... 0x7FFF	a2 coefficient of the filter

3.3.4 Availability

This library module is available in the C-callable interface assembly version format.

This library module is targeted for the DSC 56F80xx platforms.

3.3.5 Dependencies

The dependent files are:

- GDFLIB_FilterIIRasm.h
- GDFLIB_types.h

3.3.6 Description

The **GDFLIB_FilterIIR1Init** function calculates the first-order infinite impulse response (IIR) filter. The IIR filters are also called recursive filters, because both the input and the previously calculated output values are used for calculation. This form of feedback enables the transfer of energy from the output to the input, which theoretically leads to an infinitely long impulse response (IIR). A general form of the IIR filter, expressed as a transfer function in the Z-domain, is described as follows:

$$H(z) = \frac{B(z)}{A(z)} = \frac{b_1 + b_2 z^{-1} + b_3 z^{-2} + \ldots + b_{(N+1)} z^{-N}}{1 + a_2 z^{-1} + a_3 z^{-2} + \ldots + a_{(N+1)} z^{-N}} \qquad \text{Eqn. 3-1}$$

where N denotes the filter order. The first-order IIR filter in the Z-domain is therefore given as:

$$H(z) = \frac{B(z)}{A(z)} = \frac{b_1 + b_2 z^{-1}}{1 + a_2 z^{-1}} \qquad \text{Eqn. 3-2}$$

which is transformed into a time-domain difference equation as:

$$y(k) = b_1 x(k) + b_2 x(k-1) - a_2 y(k-1) \quad \textbf{Eqn. 3-3}$$

The filter difference equation is implemented in the digital signal controller directly, as written in Equation 3-3; this equation represents a direct-form one first-order IIR filter as depicted in Figure 3-1.

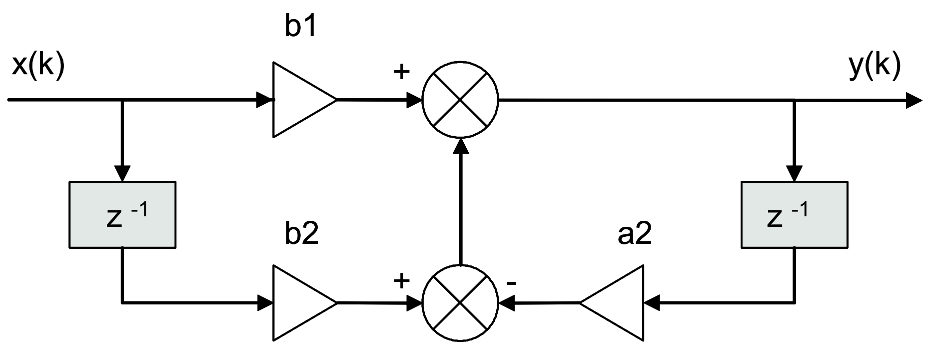

Figure 3-1. Direct-Form One First-Order IIR Filter

The coefficients of the filter depicted in Figure 3-1 can be designed to meet the requirements for the first-order low (LPF) or high-pass filter (HPF). The coefficient quantization error due to finite precision arithmetic can be neglected in the case of a first-order filter. A higher-order LPF or HPF can be obtained by connecting a number of the first-order filters in series. The number of connections gives the order of the resulting filter.

The filter coefficients are calculated using the Butterworth approximation. The Butterworth normalized transfer function in the s-plane is given as:

$$H_N(s) = \frac{1}{\prod_{k=1}^{n}(s - s_k)} \quad \textbf{Eqn. 0-1}$$

where

$$s_k = \begin{cases} e^{j(2k-1)\pi/2n} & \text{for even n} \\ e^{j(k-1)\pi/n} & \text{for odd n} \end{cases} \quad \textbf{Eqn. 3-4}$$

The normalized Butterworth first-order low-pass filter prototype is therefore given as:

$$H(s) = \frac{1}{s+1} \quad \textbf{Eqn. 3-5}$$

Transferring the prototype described in Equation 3-5 into a denormalized low-pass filter results in a transfer function:

$$H(s) = \frac{\omega_c}{s + \omega_c} \qquad \textbf{Eqn. 3-6}$$

This is a transfer function of Butterworth low-pass filter in the s-domain with the cutoff frequency given by the ω_c. Transformation of an analog filter described by Equation 3-6 into a discrete form is done using the bilinear transformation, resulting in the following transfer function:

$$H(z) = \frac{\frac{\omega_{cd}T_s}{2 + \omega_{cd}T_s} + \frac{\omega_{cd}T_s}{2 + \omega_{cd}T_s}z^{-1}}{1 + \frac{\omega_{cd}T_s - 2}{2 + \omega_{cd}T_s}z^{-1}} \qquad \textbf{Eqn. 3-7}$$

where ω_{cd} is the cutoff frequency of the filter in the digital domain and T_s is the sampling period. However, mapping of the analog system into a digital domain using the bilinear transformation makes the relation between ω_c and ω_{cd} non-linear. This introduces a distortion in the frequency scale of the digital filter relative to that of the analog filter. This is known as warping effect. The warping effect can be eliminated by pre-warping the analog filter, and then transforming it into the digital domain, resulting in this transfer function:

$$H(z) = \frac{\frac{\omega_{cd\_p}T_{s\_p}}{2 + \omega_{cd\_p}T_{s\_p}} + \frac{\omega_{cd\_p}T_{s\_p}}{2 + \omega_{cd\_p}T_{s\_p}}z^{-1}}{1 + \frac{\omega_{cd\_p}T_{s\_p} - 2}{2 + \omega_{cd\_p}T_{s\_p}}z^{-1}} \qquad \textbf{Eqn. 3-8}$$

where ω_{cd_p} is the pre-warped cutoff frequency of the filter in the digital domain, and T_{s_p} is the pre-warped sampling period. The pre-warped cutoff frequency is calculated as follows:

$$\omega_{cd\_p} = \frac{2}{T_{s\_p}} \tan\left(\frac{\omega_{cd}T_s}{2}\right) \qquad \textbf{Eqn. 3-9}$$

and the pre-warped sampling period is:

$$T_{s\_p} = 0.5 \quad \text{Eqn. 3-10}$$

Because the given filter equation is as described in Equation 3-3, the Butterworth low-pass filter coefficients are calculated as follows:

$$b_1 = \frac{\omega_{cd\_p} T_{s\_p}}{2 + \omega_{cd\_p} T_{s\_p}} \quad \text{Eqn. 3-11}$$

$$b_2 = \frac{\omega_{cd\_p} T_{s\_p}}{2 + \omega_{cd\_p} T_{s\_p}} \quad \text{Eqn. 3-12}$$

$$a_2 = \frac{\omega_{cd\_p} T_{s\_p} - 2}{2 + \omega_{cd\_p} T_{s\_p}} \quad \text{Eqn. 3-13}$$

A similar approach is adopted for a high-pass filter. Transferring the prototype described in Equation 3-5 into a denormalized high-pass filter results in this transfer function:

$$H(s) = \frac{s}{s + \omega_c} \quad \text{Eqn. 3-14}$$

Discretization of the analog filter given in Equation 3-14 by the bilinear transformation, with pre-warping the results is in the following transfer function:

$$H(z) = \frac{\dfrac{2}{2 + \omega_{cd\_p} T_{s\_p}} + \dfrac{-2}{2 + \omega_{cd\_p} T_{s\_p}} z^{-1}}{1 + \dfrac{\omega_{cd\_p} T_{s\_p} - 2}{2 + \omega_{cd\_p} T_{s\_p}} z^{-1}} \quad \text{Eqn. 3-15}$$

Because the given filter equation is as described in Equation 3-3, the Butterworth high-pass filter coefficients are calculated as follows:

$$b_1 = \frac{2}{2 + \omega_{cd\_p} T_{s\_p}} \qquad \textit{Eqn. 3-16}$$

$$b_2 = \frac{-2}{2 + \omega_{cd\_p} T_{s\_p}} \qquad \textit{Eqn. 3-17}$$

$$a_2 = \frac{\omega_{cd\_p} T_{s\_p} - 2}{2 + \omega_{cd\_p} T_{s\_p}} \qquad \textit{Eqn. 3-18}$$

3.3.7 Returns

The function returns the filtered value of the input f16In in the step k, and stores the input and the output values in the step k into the filter buffer.

3.3.8 Range Issues

The filter coefficients must be defined prior to this function call. All filter coefficients must be divided by 2.0 in order to avoid saturation during the filter calculation. Therefore in order to achieve the correct functionality, the filter output is multiplied by two. This is done automatically within the function.

3.3.9 Special Issues

The function **GDFLIB_FilterIIR1** requires the saturation mode to be set.

3.3.10 Implementation

The **GDFLIB_FilterIIR1** function is implemented as a function call.

Example 3-2. Implementation Code

```
#include <hidef.h> /* for EnableInterrupts macro */
#include "derivative.h" /* include peripheral declarations */

#include "gdflib.h"

static Frac16 mf16Value;
static Frac16 mf16FilteredValue;
static GDFLIB_FILTER_IIR1_T mudtFilterIIR1 = GDFLIB_FILTER_IIR1_DEFAULT;


void Isr(void);
```

```
void main(void)
{
        /* LPF 1st order butterworth 100Hz, Ts = 100us*/
        mudtFilterIIR1.udtFiltCoeff.f16B1 = FRAC16(0.0305 / (2.0));
        mudtFilterIIR1.udtFiltCoeff.f16B2 = FRAC16(0.0305 / (2.0));
        mudtFilterIIR1.udtFiltCoeff.f16A2 = FRAC16(-0.9391 / (2.0));

        /* Filter initialization */
        GDFLIB_FilterIIR1Init(&mudtFilterIIR1);
}

/* Periodical function or interrupt */
void Isr(void)
{
        /* Filter calculation */
        mf16FilteredValue = GDFLIB_FilterIIR1(mf16Value,
&mudtFilterIIR1);
}
```

3.3.11 Performance

This section specifies actual requirements of the function or macro in terms of required code memory, data memory, and number of clock cycles to execute.

Table 3-9. Performance of the GDFLIB_FilterIIR1 Function

Code Size (bytes)	24	
Data Size (bytes)	0	
Execution Clock	Min	43/41 cycles
	Max	43/41 cycles

3.4 GDFLIB_FilterIIR2Init

The function initializes internal variables of a second order IIR filter.

3.4.1 Synopsis

```
#include "gdflib.h"
void GDFLIB_FilterIIR2Init(GDFLIB_FILTER_IIR2_T *pudtFilter)
```

3.4.2 Prototype

```
void GDFLIB_FilterIIR2InitFC(GDFLIB_FILTER_IIR2_T * const pudtFilter)
```

3.4.3 Arguments

This subsection describes input/output arguments to a function or a macro. It explains algorithms being used by functions or macro.

Table 3-10. Function Arguments

Name	In/Out	Format	Range	Description
*pudtFilter	in/out	N/A	N/A	Pointer to a filter structure, which contains filter coefficients and filter buffer; the GDFLIB_FILTER_IIR2_T data type is defined in header file GDFLIB_FilterIIRasm.h

Table 3-11. User Type Definitions

Typedef	Name	In/Out	Format	Range	Description
GDFLIB_FILTER_IIR2_T	udtFiltCoeff	In	N/A	N/A	Structure containing filter coefficients
	f16FiltBufferX[3]	In/Out	SF16	0x8000... 0x7FFF	Filter buffer storing input values
	f32FiltBufferY[3]	In/Out	SF32	0x80000000... 0x7FFFFFFF	Filter buffer storing output values

Table 3-12. User Type Definitions

Typedef	Name	In/Out	Format	Range	Description
GDFLIB_FILTER_IIR_COEFF2_T	f16B1	in	SF16	0x8000... 0x7FFF	B1 coefficient of the filter
	f16B2	in	SF16	0x8000... 0x7FFF	B2 coefficient of the filter
	f16A2	in	SF16	0x8000... 0x7FFF	A2 coefficient of the filter
	f16B3	in	SF16	0x8000... 0x7FFF	B3 coefficient of the filter
	f16A3	in	SF16	0x8000... 0x7FFF	A3 coefficient of the filter

3.4.4 Availability

This library module is available in the ANSI C version format.

This library module is targeted for the DSC 56F80xx platform.

3.4.5 Dependencies

The dependent files are:

- GDFLIB_FilterIIRasm.h
- GDFLIB_types.h

3.4.6 Description

The **GDFLIB_FilterIIR2Init** function initializes the buffer and coefficients of a second order IIR filter. This function is called once, during variable initialization and since it clears the filter buffer it must not be called together with the filter calculation function.

3.4.7 Returns

The function initializes the filter structure pointed to by the pudtFilter pointer.

3.4.8 Range Issues

The filter coefficients must be defined prior to this function call. If Matlab filter design toolbox is used for the filter coefficients calculation then all calculated coefficients must be divided by 2.0 to avoid saturation during filter calculation.

3.4.9 Implementation

This optional subsection specifies whether call into function generates library function call or macro expansion. This subsection also consist of one or more examples of the use of the function. The examples are often fragments of code (not completed programs) for illustration purposes.

Example 3-3. Implementation Code

```
#include "gdflib.h"

static Frac16 mf16Value;
static Frac16 mf16FilteredValue;
static GDFLIB_FILTER_IIR2_T mudtFilterIIR2 = GDFLIB_FILTER_IIR2_DEFAULT;


void Isr(void);

void main(void)
{
         /* BPF Butterworth approximation fc=500Hz, bw=225Hz Ts = 100us
*/
         mudtFilterIIR2.udtFiltCoeff.f16B1= FRAC16(0.06612 / (2.0));
         mudtFilterIIR2.udtFiltCoeff.f16B2= FRAC16(0.0 / (2.0));
         mudtFilterIIR2.udtFiltCoeff.f16B3= FRAC16(-0.06612 / (2.0));
         mudtFilterIIR2.udtFiltCoeff.f16A2= FRAC16(-1.7762 / (2.0));
         mudtFilterIIR2.udtFiltCoeff.f16A3= FRAC16(0.8678 / (2.0));

         /* Filter initialization */
         GDFLIB_FilterIIR2Init(&mudtFilterIIR2);
}

/* Periodical function or interrupt at 100us*/
void Isr(void)
{
         /* Filter calculation */
         mf16FilteredValue = GDFLIB_FilterIIR2(mf16Value,
&mudtFilterIIR2);
}
```

3.4.10 Performance

This section specifies actual requirements of the function or macro in terms of required code memory, data memory and number of clock cycles to execute.

Table 3-13. Performance of GDFLIB_FilterIIR2Init Function

Code Size (words)	13 words	
Data Size (words)	0 words	
Execution Clock	Min	26 cycles
	Max	26 cycles

3.5 GDFLIB_FilterIIR2

The function calculates the second order Direct Form 1 IIR filter.

3.5.1 Synopsis

```
#include "gdflib.h"
Frac16 GDFLIB_FilterIIR2(Frac16 f16In, GDFLIB_FILTER_IIR2_T *pudtFilter)
```

3.5.2 Prototype

```
asm Frac16 GDFLIB_FilterIIR2FAsm(Frac16 f16In, GDFLIB_FILTER_IIR2_T *
const pudtFilter)
```

3.5.3 Arguments

This subsection describes input/output arguments to a function or a macro. It explains algorithms being used by functions or macro.

Table 3-14. Function Arguments

Name	In/Out	Format	Range	Description
f16In	In	SF16	0x8000... 0x7FFF	Input signal to be filtered
*pudtFilter	In/Out	N/A	N/A	Pointer to a filter structure, which contains filter coefficients and filter buffer; the GDFLIB_FILTER_IIR2_T data type is defined in header file GDFLIB_FilterIIRasm.h

Table 3-15. User Type Definitions

Typedef	Name	In/Out	Format	Range	Description
GDFLIB_FILTER_IIR2_T	udtFiltCoeff	In	N/A	N/A	Structure containing filter coefficients
	f16FiltBufferX[3]	In/Out	SF16	0x8000... 0x7FFF	Filter buffer storing input values
	f32FiltBufferY[3]	In/Out	SF32	0x80000000... 0x7FFFFFFF	Filter buffer storing output values

Table 3-16. User Type Definitions

Typedef	Name	In/Out	Format	Range	Description
GDFLIB_FILTER_IIR_COEFF2_T	f16B1	In	SF16	0x8000... 0x7FFF	b1 coefficient of the filter
	f16B2	In	SF16	0x8000... 0x7FFF	b2 coefficient of the filter
	f16A2	In	SF16	0x8000... 0x7FFF	a2 coefficient of the filter
	f16B3	In	SF16	0x8000... 0x7FFF	b3 coefficient of the filter
	f16A3	In	SF16	0x8000... 0x7FFF	a3 coefficient of the filter

3.5.4 Availability

This library module is available in the C-callable interface assembly version formats.

This library module is targeted for the DSC 56F80xx platform.

3.5.5 Dependencies

The dependent files are:

- GDFLIB_FilterIIRasm.h
- GDFLIB_types.h

3.5.6 Description

The **GDFLIB_FilterIIR2** function calculates the second order infinite impulse response (IIR) filter. IIR filters are also called recursive filters because the input and the previously calculated output values are used for calculation. This form of feedback enables transfer of the energy from the output to the input, which theoretically leads to an infinitely long impulse response (IIR).

General form of the IIR filter expressed as a transfer function in the Z-domain is described as follows:

$$H(z) = \frac{B(z)}{A(z)} = \frac{b_1 + b_2 z^{-1} + b_3 z^{-2} + \ldots + b_{(N+1)} z^{-N}}{1 + a_2 z^{-1} + a_3 z^{-2} + \ldots + a_{(N+1)} z^{-N}} \qquad \textbf{\textit{Eqn. 3-19}}$$

where N denotes the filter order. The second order IIR filter in the Z-domain is therefore given as:

$$H(z) = \frac{B(z)}{A(z)} = \frac{b_1 + b_2 z^{-1} + b_3 z^{-2}}{1 + a_2 z^{-1} + a_3 z^{-2}} \qquad \textbf{Eqn. 3-20}$$

which is transformed into the time domain difference equation as:

$$y(k) = b_1 x(k) + b_2 x(k-1) + b_3 x(k-2) - a_2 y(k-1) - a_3 y(k-2) \qquad \textbf{Eqn. 3-21}$$

The filter difference equation is implemented in Digital Signal Controller directly as written in Equation 3-21. This represents a Direct-Form 1 second order IIR filter as depicted in Figure 3-2.

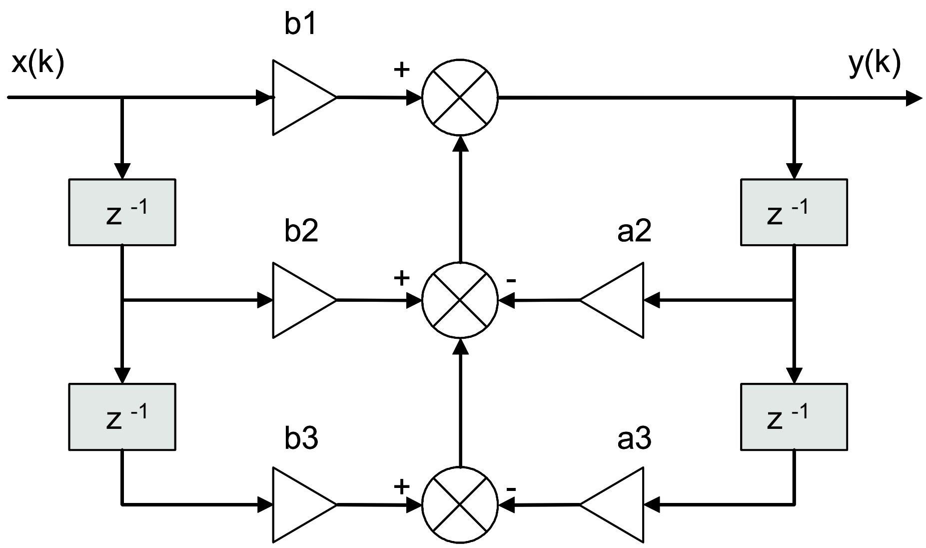

Figure 3-2. Direct Form 1 Second Order IIR filter

The coefficients of the filter depicted in Figure 3-2 can be designed to meet the requirements for the Band Pass (BPF) or the Band Stop filter (BSF). Although the filter is implemented as a second order filter, it is not recommended to use this implementation for the second order LPF or HPF due to the coefficient quantization error. This error arises because of the finite precision arithmetic used for the filter implementation. A higher order LPF or HPF can be obtained by connecting a number of the first order filters in series. The number of the connections gives the order of the resulting filter.

Filter coefficients are calculated using the Butterworth approximation. Butterworth normalized transfer function in ‘s’-plane is given as:

$$H_N(s) = \frac{1}{\prod_{k=1}^{n}(s - s_k)}$$ **Eqn. 3-22**

where

$$s_k = \begin{cases} e^{j(2k-1)\pi/2n} & \text{for even n} \\ e^{j(k-1)\pi/n} & \text{for odd n} \end{cases}$$ **Eqn. 3-23**

The normalized Butterworth second order low pass filter prototype is therefore given as:

$$H(s) = \frac{1}{s^2 + \sqrt{2}s + 1}$$ **Eqn. 3-24**

Transferring the prototype described in Equation 3-24 into a denormalized bandpass filter results in a transfer function:

$$H(s) = \frac{s\omega_{bw}}{s^2 + s\omega_{bw} + \omega_c^2}$$ **Eqn. 3-25**

which is a transfer function of Butterworth Band Pass Filter in ‘s’-domain with center frequency given by ω_c and bandwidth given by ω_{bw}. For the BPF center frequency and bandwidth relation, refer to the filter bode plot depicted on Figure 3-3.

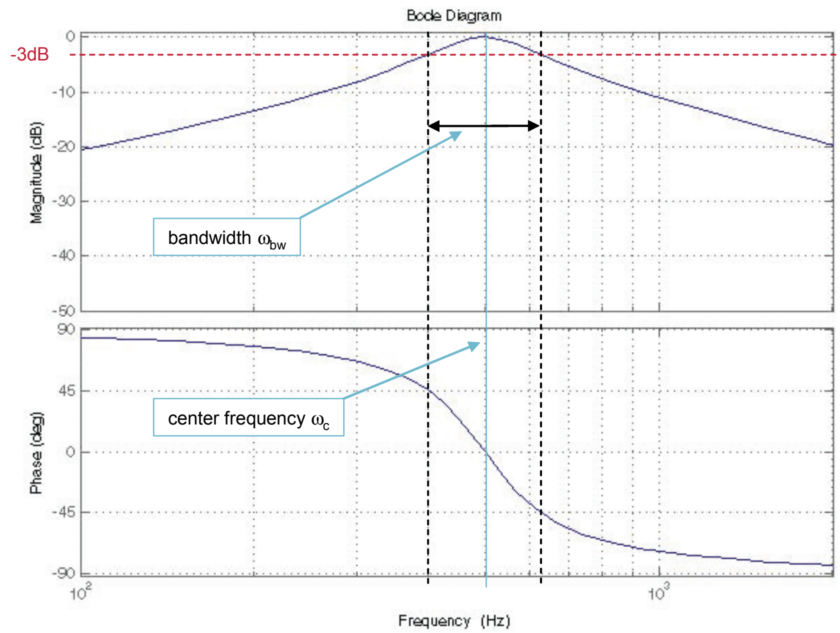

Figure 3-3. BPF Bode Plot (f_c=500Hz, f_{bw}=225Hz)

Transformation of an analog filter described by Equation 3-25 into a discrete form is done using Bilinear transformation, which results in the following transfer function:

$$H(z) = \frac{\frac{2T_s\omega_{bwd}}{C} + \frac{-2T_s\omega_{bwd}}{C}z^{-2}}{1 + \frac{2T_s^2\omega_{cd}^2 - 8}{C}z^{-1} + \frac{4 - 2T_s\omega_{bwd} + T_s^2\omega_{cd}^2}{C}z^{-2}} \qquad \text{Eqn. 3-26}$$

$$C = 4 + 2T_s\omega_{bwd} + T_s^2\omega_{cd}^2 \qquad \text{Eqn. 3-27}$$

where ω_{cd} is the center frequency, ω_{bw} is the bandwidth of the filter in the digital domain and T_s is the sampling period. However, mapping of the analog system into a digital domain using Bilinear transformation makes the relation between the analog and digital frequencies nonlinear. This introduces a distortion in the frequency scale of the digital filter relative to that of the analog filter, which is known as a warping effect. The warping effect can by eliminated by prewarping the analog filter and then transforming the prewarped transfer function into the digital domain. This results in the transfer function described as:

$$H(z) = \frac{\frac{2T_{s\_p}\omega_{bwd\_p}}{C} + \frac{-2T_{s\_p}\omega_{bwd\_p}}{C} z^{-2}}{1 + \frac{2T_{s\_p}^2\omega_{cd\_p}^2 - 8}{C} z^{-1} + \frac{4 - 2T_{s\_p}\omega_{bwd\_p} + T_{s\_p}^2\omega_{cd\_p}^2}{C} z^{-2}} \quad \text{Eqn. 3-28}$$

$$C = 4 + 2T_{s\_p}\omega_{bwd\_p} + T_{s\_p}^2\omega_{cd\_p}^2 \quad \text{Eqn. 3-29}$$

where ω_{cd-p} is the prewarped center frequency, ω_{bwd-p} is the prewarped bandwidth of the filter in digital domain and T_{s-p} is the prewarped sampling period. Prewarped center frequency is calculated as:

$$\omega_{cd\_p} = \frac{2}{T_{s\_p}} \tan\left(\frac{\omega_{cd}T_s}{2}\right) \quad \text{Eqn. 3-30}$$

prewarped bandwidth

$$\omega_{bwd\_p} = \frac{2}{T_{s\_p}} \tan\left(\frac{\omega_{bwd}T_s}{2}\right) \quad \text{Eqn. 3-31}$$

and prewarped sampling period:

$$T_{s\_p} = 0.5 \quad \text{Eqn. 3-32}$$

Therefore given the filter equation Equation 3-21, the Butterworth bandpass filter coefficients are calculated as follows:

$$b_1 = \frac{2T_{s\_p}\omega_{bwd\_p}}{4 + 2T_{s\_p}\omega_{bwd\_p} + T_{s\_p}^2\omega_{cd\_p}^2} \quad \text{Eqn. 3-33}$$

$$b_2 = 0 \quad \text{Eqn. 3-34}$$

$$b_3 = \frac{-2T_{s\_p}\omega_{bwd\_p}}{4 + 2T_{s\_p}\omega_{bwd\_p} + T_{s\_p}^2\omega_{cd\_p}^2} \quad \text{Eqn. 3-35}$$

$$a_2 = \frac{2T_{s\_p}^2 \omega_{cd\_p}^2 - 8}{4 + 2T_{s\_p}\omega_{bwd\_p} + T_{s\_p}^2 \omega_{cd\_p}^2} \qquad \text{Eqn. 3-36}$$

$$a_3 = \frac{4 - 2T_{s\_p}\omega_{bwd\_p} + T_{s\_p}^2 \omega_{cd\_p}^2}{4 + 2T_{s\_p}\omega_{bwd\_p} + T_{s\_p}^2 \omega_{cd\_p}^2} \qquad \text{Eqn. 3-37}$$

A similar approach is adopted for a bandstop filter. Transferring the normalized low pass filter prototype described in Equation 3-24 into a denormalized bandstop filter results in a transfer function:

$$H(s) = \frac{s^2 + \omega_c^2}{s^2 + \omega_{bw} s + \omega_c^2} \qquad \text{Eqn. 3-38}$$

Discretization of the analog filter given in Equation 3-38 by Bilinear transformation with prewarping results in a following transfer function:

$$H(z) = \frac{\frac{4 + \omega_{cd\_p}^2 T_{s\_p}^2}{C} + \frac{2\omega_{cd\_p}^2 T_{s\_p}^2 - 8}{C} z^{-1} + \frac{4 + \omega_{cd\_p}^2 T_{s\_p}^2}{C} z^{-2}}{1 + \frac{2\omega_{cd\_p}^2 T_{s\_p}^2 - 8}{C} z^{-1} + \frac{4 - 2\omega_{bwd\_p} T_{s\_p} + \omega_{cd\_p}^2 T_{s\_p}^2}{C} z^{-2}} \qquad \text{Eqn. 3-39}$$

where

$$C = 4 + 2T_{s\_p}\omega_{bwd\_p} + \omega_{cd\_p}^2 T_{s\_p}^2 \qquad \text{Eqn. 3-40}$$

For the BSF center frequency and bandwidth relation, refer to the filter bode plot depicted on Figure 3-4

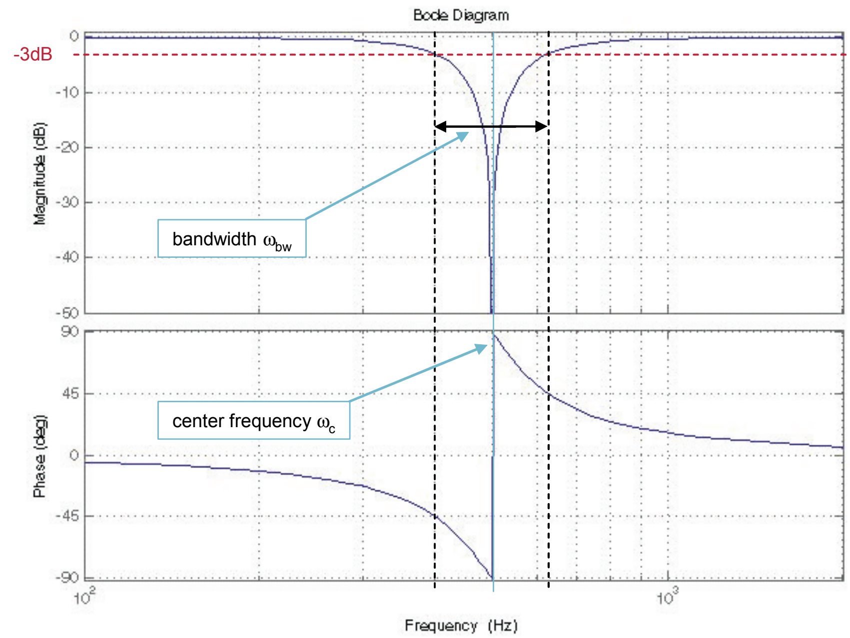

Figure 3-4. BSF Bode Plot(f_c=500Hz, f_{bw}=225Hz)

Therefore given the filter equation as described in Equation 3-21, the Butterworth bandstop filter coefficients are calculated as follows:

$$b_1 = \frac{4 + \omega_{cd\_p}^2 T_{s\_p}^2}{4 + 2T_{s\_p}\omega_{bwd\_p} + \omega_{cd\_p}^2 T_{s\_p}^2} \qquad \textbf{Eqn. 3-41}$$

$$b_2 = \frac{2\omega_{cd\_p}^2 T_{s\_p}^2 - 8}{4 + 2T_{s\_p}\omega_{bwd\_p} + \omega_{cd\_p}^2 T_{s\_p}^2} \qquad \textbf{Eqn. 3-42}$$

$$b_3 = \frac{4 + \omega_{cd\_p}^2 T_{s\_p}^2}{4 + 2T_{s\_p}\omega_{bwd\_p} + \omega_{cd\_p}^2 T_{s\_p}^2} \qquad \textbf{Eqn. 3-43}$$

$$a_2 = \frac{2\omega_{cd\_p}^2 T_{s\_p}^2 - 8}{4 + 2T_{s\_p}\omega_{bwd\_p} + \omega_{cd\_p}^2 T_{s\_p}^2} \qquad \textbf{Eqn. 3-44}$$

$$a_3 = \frac{4 - 2T_{s\_p}\omega_{bwd\_p} + \omega_{cd\_p}^2 T_{s\_p}^2}{4 + 2T_{s\_p}\omega_{bwd\_p} + \omega_{cd\_p}^2 T_{s\_p}^2} \quad \textit{Eqn. 3-45}$$

To avoid saturation, all the filter coefficients given in Equation 3-33 - Equation 3-37 and Equation 3-41 - Equation 3-45 must be divided by two to be able to implement it on the embedded side. Moreover the coefficients are implemented as 16-bit numbers, therefore the coefficients calculated as fractional numbers must be transformed into integer numbers (to be used on the target platform). The transformations are given as follows:

$$B_1 = \frac{b_1}{2} * 2^{15} \quad \textit{Eqn. 3-46}$$

$$B_2 = 0 \quad \textit{Eqn. 3-47}$$

$$B_3 = \frac{b_3}{2} * 2^{15} \quad \textit{Eqn. 3-48}$$

$$A_2 = \frac{a_2}{2} * 2^{15} \quad \textit{Eqn. 3-49}$$

$$A_3 = \frac{a_3}{2} * 2^{15} \quad \textit{Eqn. 3-50}$$

3.5.7 Returns

The function returns filtered value of input f16In in the step k and stores the input and output values in the step k into the filter buffer.

3.5.8 Range Issues

The filter coefficients must be defined prior to this function call. All filter coefficients must be divided by 2.0 to avoid saturation during filter calculation. Therefore, to achieve correct functionality, filter output is multiplied by two. This is done automatically within the function.

3.5.9 Special Issues

The function **GDFLIB_FilterIIR2** requires the saturation mode to be set.

3.5.10 Implementation

This optional subsection specifies whether call into function generates library function call or macro expansion. This subsection also consist of one or more examples of the use of the function.

Example 3-4. Implementation Code

```
#include "gdflib.h"

static Frac16 mf16Value;
static Frac16 mf16FilteredValue;
static GDFLIB_FILTER_IIR2_T mudtFilterIIR2 = GDFLIB_FILTER_IIR2_DEFAULT;


void Isr(void);

void main(void)
{
        /* BPF Butterworth approximation fc=500Hz, bw=225Hz Ts = 100us
*/
        mudtFilterIIR2.udtFiltCoeff.f16B1= FRAC16(0.06612 / (2.0));
        mudtFilterIIR2.udtFiltCoeff.f16B2= FRAC16(0.0 / (2.0));
        mudtFilterIIR2.udtFiltCoeff.f16B3= FRAC16(-0.06612 / (2.0));
        mudtFilterIIR2.udtFiltCoeff.f16A2= FRAC16(-1.7762 / (2.0));
        mudtFilterIIR2.udtFiltCoeff.f16A3= FRAC16(0.8678 / (2.0));

        /* Filter initialization */
        GDFLIB_FilterIIR2Init(&mudtFilterIIR2);
}

/* Periodical function or interrupt at 100us*/
void Isr(void)
{
        /* Filter calculation */
        mf16FilteredValue = GDFLIB_FilterIIR2(mf16Value,
&mudtFilterIIR2);
}
```

3.5.11 Performance

This section specifies actual requirements of the function or macro in terms of required code memory, data memory and number of clock cycles to execute.

Table 3-17. Performance of GDFLIB_FilterIIR2 function

Code Size (words)	39 words	
Data Size (words)	0 words	
Execution Clock	Min	61/56 cycles
	Max	61/56 cycles

3.6 GDFLIB_FilterMA32Init

This function initializes the internal variables of a moving average filter.

3.6.1 Synopsis

```
#include "gdflib.h"
void GDFLIB_FilterMA32Init(GDFLIB_FILTER_MA32_T *pudtFilter)
```

3.6.2 Prototype

```
void GDFLIB_FilterMA32InitFC(GDFLIB_FILTER_MA32_T * const pudtFilter)
```

3.6.3 Arguments

This subsection describes the input/output arguments to a function or a macro. It explains the algorithms being used by the functions or macro.

Table 3-18. Function Arguments

Name	In/ Out	Format	Range	Description
*pudtFilter	in/out	N/A	N/A	Pointer to a filter structure, which contains filter coefficients and filter buffer; the GDFLIB_FILTER_MA32_T data type is defined in header file GDFLIB_FilterMA32asm.h.

Table 3-19. User-Type Definitions

Typedef	Name	In/ Out	Form at	Range	Description
GDFLIB_FILTER_MA32_T	f32Acc	in/ out	SF32	$80000000... $7FFFFFFF	internal filter accumulator
	w16N	in	SI16	$8000... $7FFF	number of filtered points (filter window size)

3.6.4 Availability

This library module is available in the The ANSI C version formats.

This library module is targeted for the DSC 56F80xx platforms.

3.6.5 Dependencies

The dependent files are:

- GDFLIB_FilterMA32asm.h
- GDFLIB_types.h

3.6.6 Description

The **GDFLIB_FilterMA32Init** function initializes the accumulator of a moving average filter. This function is called once, during the variable initialization, and since it clears the filter buffer, it must not be called together with the filter calculation function. The size of the filter window (number of filtered points) shall be defined prior to this function call. The number of the filtered points is defined by assigning a value to the pFilter.w16N variable, stored within the filter structure. This number represents the number of filtered points as a power of two, as follows:

$$n_p = 2^{(\mathrm{pudtFilter.w16N})} \quad , \mathrm{pudtFilter.w16N} \geq 0 \qquad \textbf{\textit{Eqn. 3-51}}$$

where n_p is the actual number of filtered points (size of the filter window).

3.6.7 Returns

This function initializes the filter accumulator in the filter structure pointed to by the pudtFilter pointer.

3.6.8 Special Issues

The function **GDFLIB_FilterMA32Init** is the saturation mode independent.

3.6.9 Implementation

The **GDFLIB_FilterMA32Init** function is implemented as a function call.

Example 3-5. Implementation Code

```
#include "gdflib.h"

static GDFLIB_FILTER_MA32_T mudtFilterMA32 = GDFLIB_FILTER_MA32_DEFAULT;
static Frac16 mf16Value;
static Frac16 mf16FilteredValue;

void Isr(void);

void main(void)
{
        /* filter window size 2 ^ 2 = 4 points */
        mudtFilterMA32.w16N = 2;

        /* Filter initialization */
        GDFLIB_FilterMA32Init(&mudtFilterMA32);
}

/* Periodical function or interrupt */
void Isr(void)
{
        /* Filter calculation */
```

```
        mf16FilteredValue = GDFLIB_FilterMA32(mf16Value,
&mudtFilterMA32);
}
```

3.6.10 Performance

This section specifies the actual requirements of the function or macro in terms of required code memory, data memory, and number of clock cycles to execute.

Table 3-20. Performance of the GDFLIB_FilterMA32Init Function

Code Size (bytes)	2	
Data Size (bytes)	0	
Execution Clock	Min	16
	Max	16

3.7 GDFLIB_FilterMA32

This function calculates a recursive form of an average filter. It also has an inline version.

3.7.1 Synopsis

```
#include "gdflib.h"
Frac16 GDFLIB_FilterMA32(Frac16 f16In, GDFLIB_FILTER_MA32_T * const
pudtFilter)
Frac16 GDFLIB_FilterMA32i(Frac16 f16In, GDFLIB_FILTER_MA32_T * const
pudtFilter) - inline version
```

3.7.2 Prototype

```
asm Frac16 GDFLIB_FilterMA32FAsm(Frac16 f16In, GDFLIB_FILTER_MA32_T *
const pudtFilter)
inline Frac16 GDFLIB_FilterMA32FAsmi(register Frac16 f16In,
GDFLIB_FILTER_MA32_T *pudtFilter)
```

3.7.3 Arguments

This subsection describes the input/output arguments to a function or a macro. It explains the algorithms being used by functions or macro.

Table 3-21. Function Arguments

Name	In/Out	Format	Range	Description
f16In	in	SF16	0x8000... 0x7FFF	input signal to be filtered
*pudtFilter	in/out	N/A	N/A	Pointer to a filter structure, which contains filter coefficients and filter buffer; the GDFLIB_FILTER_MA32_T data type is defined in the header file GDFLIB_FilterMA32asm.h.

Table 3-22. User-Type Definitions

Typedef	Name	In/Out	Format	Range	Description
GDFLIB_FILTER_MA32_T	f32Acc	in/out	SF32	0x80000000... 0x7FFFFFFF	internal filter accumulator
	w16N	in	SF16	0x8000... 0x7FFF	number of filtered points (filter window size)

3.7.4 Availability

This library module is available in the C-callable interface assembly formats.

This library module is targeted for the DSC 56F80xx platforms.

3.7.5 Dependencies

The dependent files are:

- GDFLIB_FilterMA32asm.h
- GDFLIB_types.h

3.7.6 Description

The **GDFLIB_FilterMA32** function calculates a recursive form of an average filter. The filter calculation consists of the following equations:

$$acc(k) = acc(k-1) + x(k)$$ **Eqn. 3-52**

$$y(k) = \frac{acc(k)}{n_p}$$ **Eqn. 3-53**

$$acc(k) \leftarrow acc(k) - y(k)$$ **Eqn. 3-54**

where x(k) is the actual value of the input signal, acc(k) is the internal filter accumulator, y(k) is the actual filter output, and n_p is the number of points in the filtered window. The size of the filter window (number of filtered points) shall be defined prior to this function call. The number of filtered points is defined by assigning a value to the pFilter.w16N variable, stored within the filter structure. This number represents the number of filtered points as a power of 2, as follows:

$$n_p = 2^{(\mathrm{pudtFilter.w16N})} \quad , \mathrm{pudtFilter.w16N} \geq 0$$ **Eqn. 3-55**

where n_p is the actual number of filtered points (size of filter window).

3.7.7 Returns

The function returns the filtered value of the input f16In in the step k, and stores the difference between the filter accumulator and the output in the step k into the filter accumulator.

3.7.8 Range Issues

The internal filter accumulator acc(k) is implemented as a 32-bit variable.

3.7.9 Special Issues

The size of the filter window (number of filtered points) must be defined prior to this function call and must be equal to or greater than zero.

The **GDFLIB_FilterMA32** function is the saturation mode independent.

3.7.10 Implementation

The **GDFLIB_FilterMA32** function is implemented as a function call

Example 3-6. Implementation Code

```
#include "gdflib.h"

static GDFLIB_FILTER_MA32_T mudtFilterMA32 = GDFLIB_FILTER_MA32_DEFAULT;
static Frac16 mf16Value;
static Frac16 mf16FilteredValue;

void Isr(void);

void main(void)
{
        /* filter window size 2 ^ 2 = 4 points */
        mudtFilterMA32.w16N = 2;

        /* Filter initialization */
        GDFLIB_FilterMA32Init(&mudtFilterMA32);
}

/* Periodical function or interrupt */
void Isr(void)
{
        /* Filter calculation */
        mf16FilteredValue = GDFLIB_FilterMA32(mf16Value,
&mudtFilterMA32);
}
```

3.7.11 Performance

This section specifies the actual requirements of the function or macro in terms of required code memory, data memory, and number of clock cycles to execute.

Table 3-23. Performance of the GDFLIB_FilterMA32 Function

Code Size (bytes)	10[1]/15[2]	
Data Size (bytes)	0	
Execution Clock	Min	27/25[1]/17[2] cycles
	Max	27/25[1]/17[2] cycles

1 measurements valid for GDFLIB_FilterMA32

2 measurements valid for inline version GDFLIB_FilterMA32i

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