QuickFil 5.1

User Manual

Table of Contents

1 Introduction

1.1 Structure Of The Manual

Welcome to *QuickFil.*

This manual is divided into the following sections:

Chapter 1 “INSTALLATION” - specifies the minimum hardware requirements and shows how to install the software.

Chapter 2 “PROGRAM INTRODUCTION” - provides you with a short summary of the program’s features.

Chapter 3 “TUTORIAL” - will acquaint you with the software using a simple example.

Chapter 4 “OPERATING INSTRUCTIONS” - tells you about the operation of *QuickFil*.

Chapter 5 “PROGRAM FUNCTIONS” - provides you with detailed information on the functions of the program.

Chapter 6 "ISSPICE INTERFACE" - explains the interface to ISSPICE

“APPENDIX” - contains background information on the technical aspects of the program, instructions for creation of key macros for automatic sequences, and a list of useful literature.

“INDEX” - allows you to find specific explanations of certain key words.

1.2 Typefaces

In this manual, instructions for communicating with the program through the keyboard are given as follows:

Examples:

• 10M[↵]	Enter character sequence “10M”, followed by the ENTER key.
• [A]	Press “A” to access a menu field.
• [Ctrl+U]	Press and hold the CTRL key down while pressing “U”.

- Uppercase letters, followed by a colon, are *QuickFil* menu entries. For example, SPECIFICATION would refer to the Specification menu.
- Characters or messages which the program displays on the screen are shown as follows:

FILTER NOT YET SPECIFIED.

1.3 Installing QuickFil

QuickFil is a DOS program which can be performed in Windows environment. A comfortable installation program will reduce the efforts of the user.

There are following versions available:

- CD — The program is on a compact disk (CD ROM) available for the languages English and German. Please insert the CD and the installation program will start automatically. Please choose the language and follow the further instructions.
- ZIP-version — This is a ZIP archive for the installation program. After extracting the ZIP file (using Winzip or some other extracting program) to some temporary directory, you can start the installation program by executing SETUP.EXE. If the installation is finished, you can remove the temporary directory and all its files.
- Executable version — By starting the executable file, a temporary directory is created and all files are extracted to that directory. Afterwards, the setup program is started automatically. If the installation is finished, this temporary directory is removed.

Hardware Requirements

Minimum configuration:

- IBM PC or compatible
- CGA, HGC, EGA or VGA graphics

Optional:

- Printer
- Plotter (HPGL format)
- Serial interface for output device: only available for Windows 95, Windows 98 and Windows Me

Operating System Requirements

- Windows 95, Windows 98, Windows Me, Windows NT4, Windows 2000 or Windows XP
- A special DOS version is available on request.

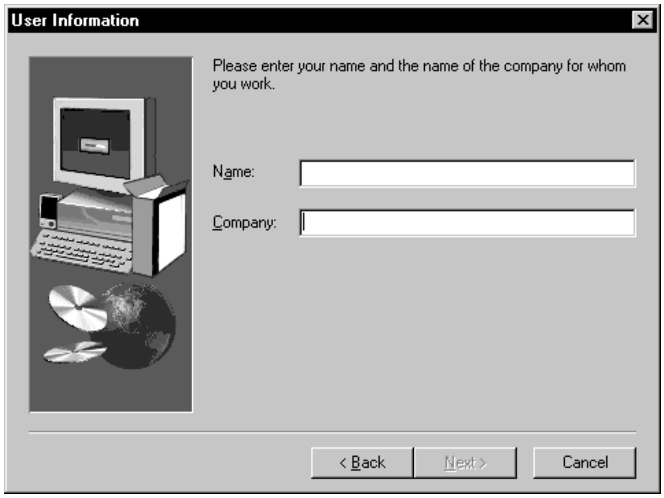
Installing the Software

Before installing *QuickFil*:

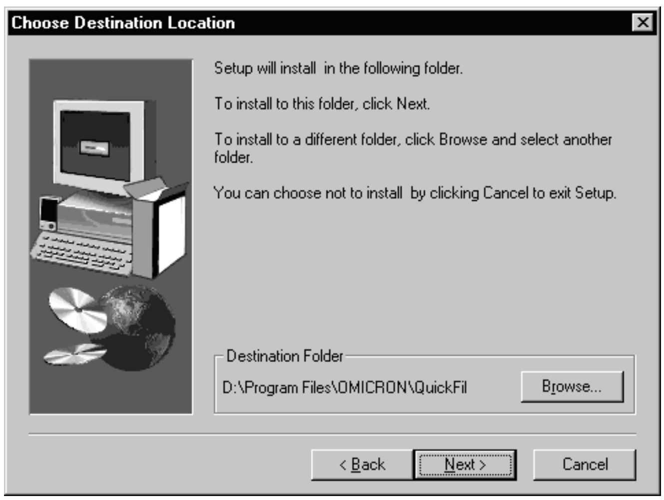
- Make a backup copy of the program.

The installation program is an up to date Windows installation program, and is self explanatory.

After agreeing to the License agreement, you must specify your name and the name of the company:



Afterwards you can specify the directory where the program will be installed:



The default path will be Drive:\Program Files\OMICRON*QuickFil*. The default drive is the drive where the operation system is installed (system drive).

The program will be installed to the specified directory and a short cut to the desktop will be created. Further, there will be a shortcut to the menu START | PROGRAMS.

1.4 Starting QuickFil

For starting the program *QuickFil*, simply click at the icon at the desktop or use the list in START | PROGRAMS, and choose the program *QuickFil*.

1.5 Remove QuickFil

Using the menu: START | SETTINGS | CONTROL PANEL | ADD/REMOVE PROGRAMS, you can remove the program in the standard way for Windows.

If you install a new version of *QuickFil*, the old version of the program is removed automatically.

1.6 Hints for the DOS version

Since the basic program is a DOS program, the following hints are for DOS users.

Graphics Related Installation Problems

Computers with color graphics adapter, but only a monochrome display (e.g. laptops):

If you are not pleased with the display representation on your screen:

- Exit the program (by entering [Q], acknowledge with [Y]) and use extended start command.
- Type either QF /bw[↵] or QF /lcd[↵].

This switches your computer’s screen to another display mode. If you start the program properly but still get “junk” on your screen: When the program starts, *QuickFil* will automatically check which graphics card is in your computer and set itself accordingly. However, there are certain circumstances that it may not be able to identify the card.

To solve this problem:

- Exit the program.
- Edit the file QF.DEF in any word processing program (e.g. IsEd, MS Word, etc.)
- After the line BEGIN GRAPHICDEFAULTS, change the entry in the line DISPLAYTYPE to suit your screen.

Possible inputs include:

CGAHi	(640x200, mono)
MCGAHi	(640x48, 2-color)
MCGAMed	(640x200, mono)
EGAHi	(640x350, 16 colors (256K))
EGA64Hi	(640x350, 4-color (64K))
EGAMonoHi	(640x350, mono)
HERCMonoHi	(720x348, mono)
ATT400Hi	(640x400, mono)
ATT400Med	(640x200, mono)

For Example:

The line, DISPLAYTYPE IBM8514HI, will set the display type to the IBM8514 1024x768 standard.

- Save the QF.DEF file as an ASCII file.

1.7 Quitting QuickFil

To quit the *QuickFil* program:

- Select Quit, or press Q, or Esc until you are at the *QuickFil* main menu.
- Press Q, while in the *QuickFil* menu. Answer yes, to confirm and exit the program.

2 Program Introduction

2.1 Features

QuickFil is a CAE software program for specifying, dimensioning and analyzing passive analog filters. The program offers the following features:

FILTER TYPES:

- Lowpass
- Highpass
- Bandpass
- Bandstop

APPROXIMATIONS:

Standard approximations

- Elliptic (Cauer filter)
- Butterworth (potence filter)
- Chebyshev
- Inverse Chebyshev
- Bessel (for lowpass only)
- Modified Bessel (for lowpass only)

General amplitude approximations

- Maximally flat
- Equal ripple

each referring to the passband.

CIRCUIT DESIGN:

Passive LC/reactance filters

- Calculation of filters with topologies suggested by the program, or
- Design of filters on a modular (piece by piece) basis. With this approach, the user can synthesize the filter step-by-step. *QuickFil* will provide selections for each element at each step.
- Modifications of synthesized circuits (Norton's transformation etc.)
- Different terminating resistances and extreme terminations may be selected.
- Graphic analysis of the characteristics of the current circuit
- Interface with the ICAP and Touchstone analysis programs Active RC filters.

ANALYSIS SECTION:

Graphical analysis of

- Transmission characteristics (magnitude, phase response, group delay etc.)
- Reflections
- Impedances
- Up to four diagrams can be displayed at one time.
- Linear or logarithmic scaling
- Auto or manual scaling
- Waveform Cursor function
- Output to plotters, printers, or file

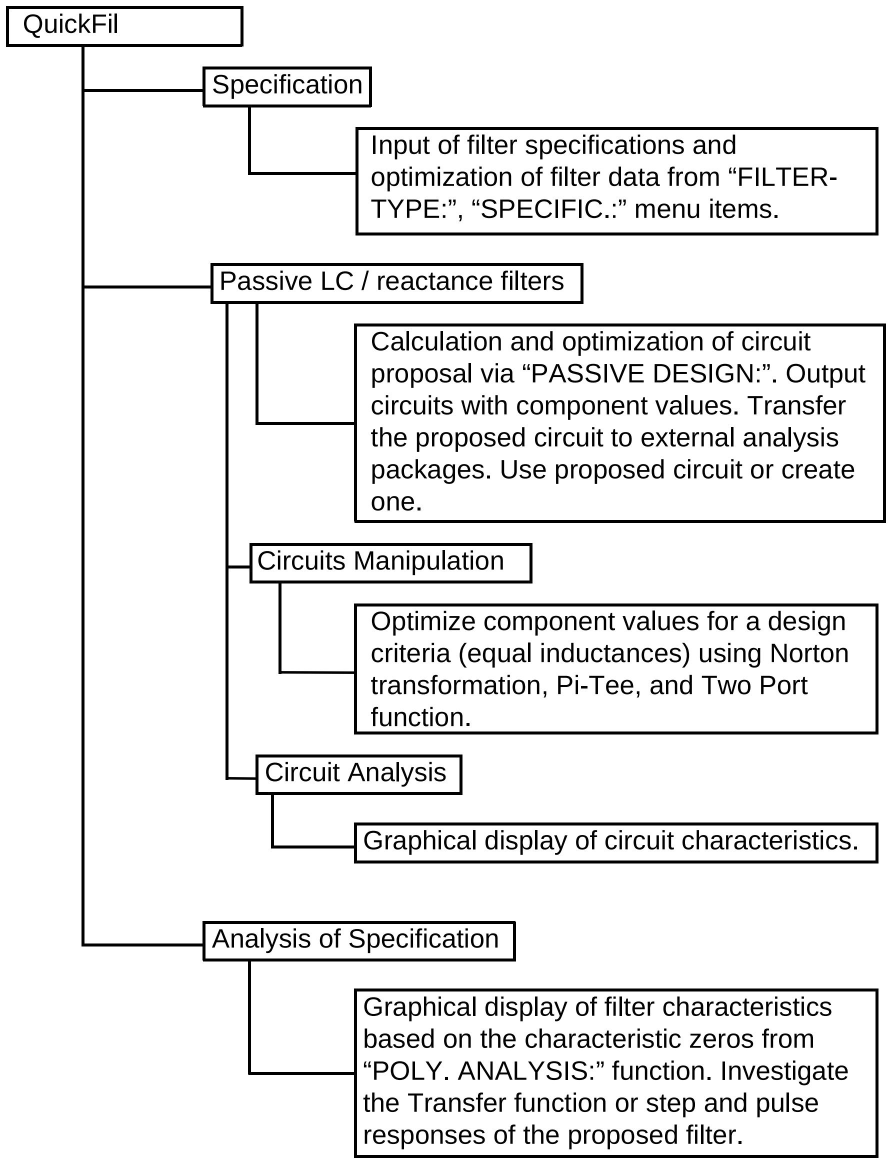
APPLICATIONS:

- Design of passive reactance filter circuits
- Optimization of filter specifications
- Rapid cost estimating
- Feasibility studies of different specifications
- Compare different realization possibilities (approximations, circuit structures etc.)

INTERFACE:

- Menu-driven
- Mouse support
- On-line help information
- Keystroke macro support

2.2 Program Structure



2.3 Program Support

QuickFil offers interfaces to the following programs:

ISSPICE/SPICENET

Output the filter circuit in an ISSPICE subcircuit format, or as a stand-alone ISSPICE netlist. When output as a subcircuit, the file created will have the same format as other SPICE model library files included in the Intusoft ICAPS package. Netlists may also contain component tolerances for use in the Monte Carlo statistical yield analysis.

Touchstone

Output the filter circuit in a Touchstone netlist format.

All SPICE netlists are in an ASCII format and can be used with virtually any SPICE program.

3 Tutorial

This chapter will acquaint you with *QuickFil* by using a simple example. You will learn how to:

- Operate the program
- Activate the on-line help
- Enter a filter's specifications
- Obtain recommendations for complete filter design
- Analyze a filter's characteristics graphically

Hint for mouse users

All examples in this manual are written based on operation from the keyboard. If you would like to use your mouse in the examples, please read the remarks in the "Selecting Options Using The Mouse" section.

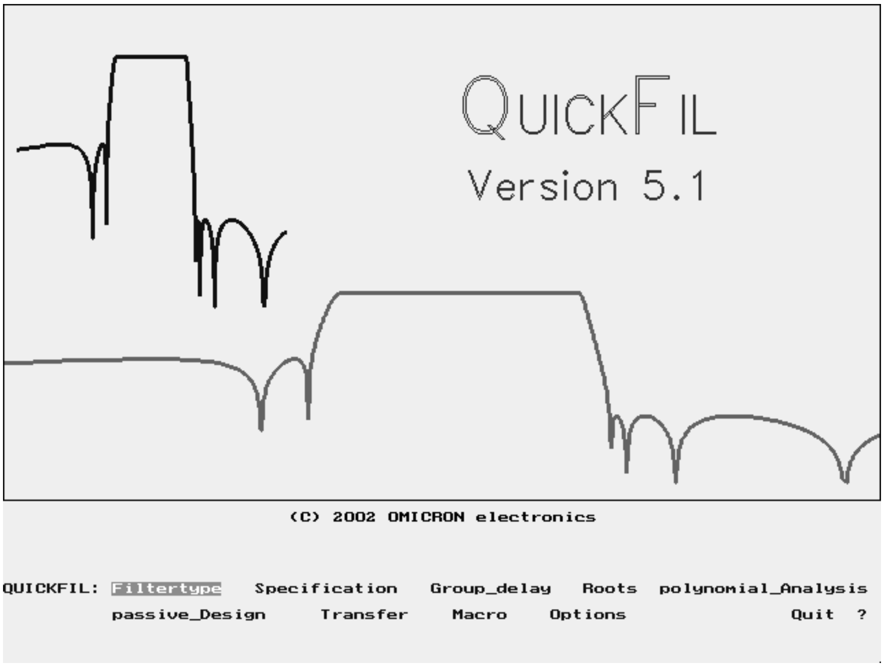
For this tutorial example, we will design a bandpass filter with the following specifications:

- Passband: 87.5 MHz ... 108 MHz
- Stopband: 0 - 70 MHz and from 140 MHz
- Stopband attenuation: 45 dB
- Return loss (matching): 16 dB

Selected approximation: Elliptic filter (Cauer)

3.1 Starting The Program

After we have installed *QuickFil*, we can start the program by clicking at the icon on the desktop. Alternatively, we can start the program from the Windows menu START | PROGRAMS. After a short period, while the program is being initialized, the Main Menu will be displayed:



We are now in the *QuickFil* main menu. To get to a defined initial state, we will reset the program:

- [O] Go to the OPTIONS menu. (You can select various menu functions by entering the appropriate capital letter. In the letter ‘O’).
- [R] Select the program_Reset option.
- [↵] Answer to query (Y/N).
- [Q] Back to the main menu.

Now we are ready to start the filter design.

3.2 Specifying The Filter

From the main menu, press:

- [↵] Change to the FILTERTYPE menu. The screen will now display the following options:

```
Filtertype                         Approximation

Lowpass                            General Equal-ripple
Highpass                           General Maximally-flat
Bandpass
Bandstop                           Butterworth
Allpass                            Chebychev
                                   Inverse Chebychev
                                   Elliptic (Cauer)
                                   Bessel (Thomson)
                                   Modified Bessel


FILTERTYPE: Type  Approximation                                  Quit  ?
```

Initially a Butterworth lowpass is selected. However, we want to calculate a bandpass filter with an Elliptic approximation. To change to the bandpass filter type:

- [↵] Go to TYPE menu.
- [B] Bandpass

Now, we specify the desired approximation.

- [A] Change to the APPROXIMATION menu.
- [E][↵] Elliptic approximation (Cauer filter)

The screen will show changed settings by highlighting the entries.

To use the mouse to activate functions:

- Just click on the desired settings on the screen.

It is now time to enter the filter specifications.

- [Q] Back to the main menu.
- [S] Go to the SPECIFICATION menu.

The following screen will appear:

```
          SPECIFICATIONS to :  Elliptic (Cauer) - bandpass filter
 (O)

 (A)  Lower passband edge frequency      :              not defined
 (B)  Upper passband edge frequency      :              not defined
 (C)  Lower stopband edge frequency      :              not defined
 (D)  Upper stopband edge frequency      :              not defined

 (E)  Passband bandedge loss             :              not defined
 (F)  Passband bandedge return loss      :              not defined
 (R)  Passband reflection factor         :              not defined
 (G)  Stopband loss                      :              not defined  ◄

 (H)  Filter degree                      :              not defined
 (I)  Case                    (a, b, c)  :                       b
 (J)  Variable value (A,B,C,D,E,F,G,H,R) :                       G

      Lower 3dB edge frequency           :              not defined
      Upper 3dB edge frequency           :              not defined
      Filter quality                     :              not defined

SPECIFICATION: A  B  C  D  E  F  R  G  H  I  J   New  cOmment   fiLe    Printer
               freqUencyrepres.  bandwithrepreS.  reL.bandwithrepres.   Quit  ?
```

If you're not clear about any of the terms on the screen, you can call up the on-line help:

- [?] Activates the on-line help.

The following help text will appear:

```
SPEC. (Butterw...):                                                    Page 1/7

Here you can specify the performance of your filter. You can define limits
for the attenuation response.

(Lower) passband limit frequency
(Lower) limit frequency at which the attenuation reaches the specified
passband attenuation.

Upper passband limit frequency
Upper limit frequency at which the attenuation reaches the specified
passband attenuation (bandpass / bandstop only).

Lower stopband limit frequency
(Lower) limit frequency at which the attenuation reaches the specified
stopband attenuation.

(Upper) stopband limit frequency
Upper limit frequency at which the attenuation reaches the specified
stopband attenuation (bandpass / bandstop only).


HELP: Resume  Next_page  Previous_page  Contents  About
```

The page number in the top right-hand corner, tells us that there are more pages available on this topic:

- [N] Next page

```
SPEC. (Butterw...):                                                    Page 2/7

For standard approximations and for bandpass and bandstop filters
(symmetrical filters) you can specify the center frequency and the
bandwidth alternatively. To change to that representation choose:
bandwithrepreS. or reL.bandwithrepres.  Using freqUencyrepres. you
will have the classical frequency definition.

Centre frequency
Geometrical meanvalue of the lower and upper passband edge frequencies.

Passband bandwidth
[Upper passband edge frequency]  minus  [Lower passband edge frequency]

Stopband bandwidth
[Upper stopband edge frequency]  minus  [Lower stopband edge frequency]

Relative passband bandwidth
[Bandwidth of the passband] divided by the [Center frequency].

Relative stopband bandwidth
[Bandwidth of the stopband] divided by the [Center frequency].

HELP: Resume  Next_page  Previous_page  Contents  About
```

Now, we have all the needed information so let's continue entering our specifications:

- [R] Return to the point in program where the help was called up.

Now, let's choose case "c" for equal terminating resistances:

- [I][C][Enter] Case "c"

Enter the following specifications:

- [A] Lower passband edge frequency
- 87.5M[↓] Value: 87.5 MHz

Note: please make sure you use the right upper case/lower case scaling units (e.g. m= milli-, M = mega- etc.) This could also be input as 87.5E6.

- 108M[↓] Upper passband edge frequency: 108 MHz
- 70M[↵] Lower stopband edge frequency: 70 MHz

When you confirm the last input, the upper stopband edge frequency will also appear in the mask (135 MHz). This occurs because with symmetrical bandpass filters, if any three frequencies are known, the fourth can be calculated. Every time you enter a value, *QuickFil* checks to see if the entry affects or generates any other values. We are happy with this value. In fact, it's less than the value we wanted! Now let's enter the required return loss:

- [F]16[↵] Return loss: 16 dB

Entering the return loss, automatically gives the passband attenuation: approximately 0.11 dB.

Next, the stopband attenuation:

- [G]45[↵] Stopband loss: 45 dB

Once this input is confirmed, the program automatically calculates the filter degree and displays it on the screen:

```
           SPECIFICATIONS to :  Elliptic (Cauer) - bandpass filter
   (O)

   (A)   Lower passband edge frequency      :           87.500 000 MHz
   (B)   Upper passband edge frequency      :          108.000 000 MHz
   (C)   Lower stopband edge frequency      :           70.000 000 MHz
   (D)   Upper stopband edge frequency      :          135.000 000 MHz

   (E)   Passband bandedge loss             :              0.110 483 dB
   (F)   Passband bandedge return loss      :                  16.00 dB
   (R)   Passband reflection factor         :                  15.85 %
   (G)   Stopband loss                      :                  53.95 dB

   (H)   Filter degree                      :                      8    ◄
   (I)   Case                    (a, b, c)  :                      c
   (J)   Variable value (A,B,C,D,E,F,G,H,R) :                      H

         Lower 3dB edge frequency           :           85.487 587 MHz
         Upper 3dB edge frequency           :          110.542 364 MHz
         Filter quality                     :                  17.12
SPECIFICATION: A  B  C  D  E  F  R  G  H  I  J   New  cOmment  fiLe    Printer
               freqUencyrepres.  bandwithrepreS.  reL.bandwithrepres.   Quit  ?
```

No doubt you've noticed that when the program calculated the degree, it also revised the stopband attenuation, increasing it to 53.95 dB. With the current value, this specification still has a margin that can be used to improve other filter specifications.

We would rather have a better return loss (matching) than a higher stopband loss. So lets' try to use this "margin" to improve the return loss.

Degree, stopband attenuation, return loss and passband attenuation vary with one another, so before we start changing any of these parameters, we need to know which of the others we can or should change. We can do this by setting any of these fields to "variable".

In this case, we want to optimize return loss:

- [J] Variable value
- [F][↵] Return loss set to "variable"

The position of the arrows, in the right-hand part of the screen, shows us that changes were made. The three values are set to variable because all three values are directly linked to one another.

Alternatively you can use the mouse:
A change of the variable value can be effected by clicking on the empty space behind the respective field. *QuickFil* then sets the marking to the selected position. Now let's enter a new stopband loss which we would prefer - let's say 50 dB.

- [G]50[↵] Stopband attenuation: 50 dB

The screen will now appear as follows:

```
          SPECIFICATIONS to :  Elliptic (Cauer) - bandpass filter
 (O)

 (A)  Lower passband edge frequency      :          87.500 000 MHz
 (B)  Upper passband edge frequency      :         108.000 000 MHz
 (C)  Lower stopband edge frequency      :          70.000 000 MHz
 (D)  Upper stopband edge frequency      :         135.000 000 MHz

 (E)  Passband bandedge loss             :            0.044 818 dB   ◄
 (F)  Passband bandedge return loss      :                19.89 dB   ◄
 (R)  Passband reflection factor         :                10.13 %    ◄
 (G)  Stopband loss                      :                50.00 dB

 (H)  Filter degree                      :                       8
 (I)  Case                    (a, b, c)  :                       c
 (J)  Variable value (A,B,C,D,E,F,G,H,R) :                       E

      Lower 3dB edge frequency           :          84.698 809 MHz
      Upper 3dB edge frequency           :         111.571 817 MHz
      Filter quality                     :                   14.50
SPECIFICATION: A  B  C  D  E  F  R  G  H  I  J   New  cOmment   fiLe    Printer
               freqUencyrepres.  bandwithrepreS.  reL.bandwithrepres.    Quit  ?
```

The return loss has changed and with it, the passband loss.These two factors are directly connected. (With loss-free filters, the wave (energy), which does not reach the output, returns to the input).

Now we're happy with our specifications, so we'll go on and design the filter:

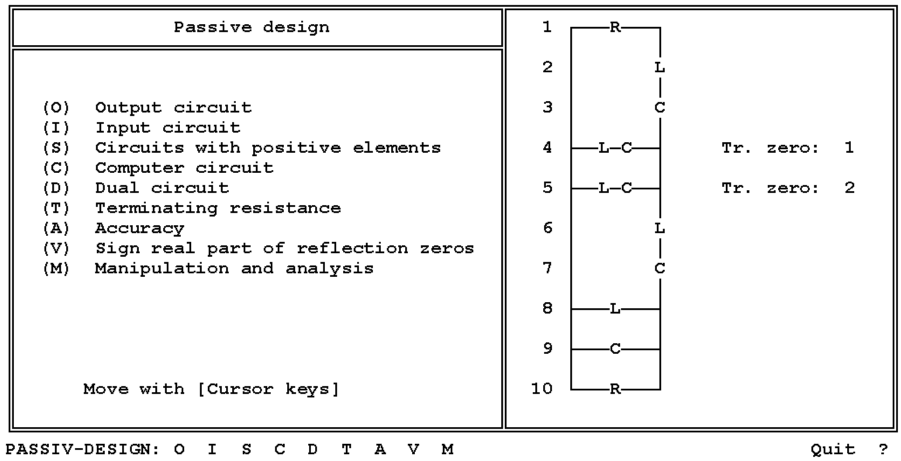
- [Q] Back to the main menu, calculating roots in the process.

3.3 Designing The Circuit

From the main menu type:

- [D] Go to the PASSIVE DESIGN menu.

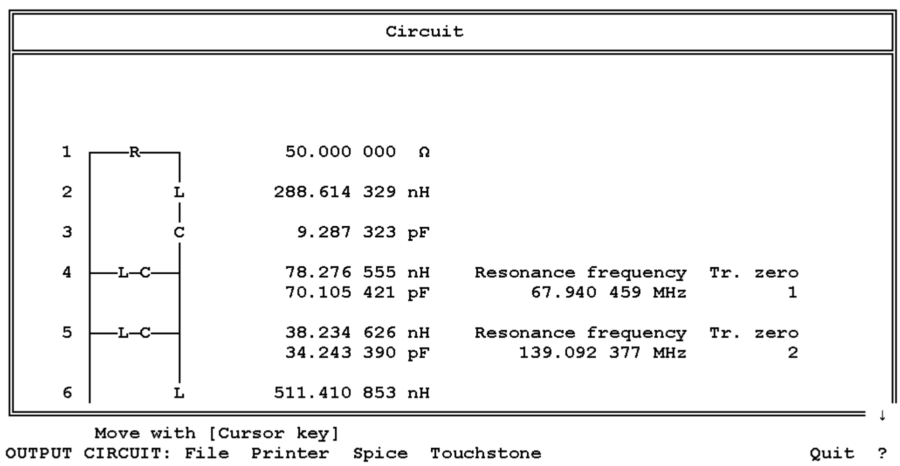
The screen will display the following:



This screen already shows the circuit topology but not the component values. To show the values, press:

- [↵] Go to the OUTPUT CIRCUIT menu

The program takes a moment while performing the calculations (reactances, components) and then, displays the circuit on the screen, along with all the component values.



To see the next part of the circuit:

- [↓] Scroll through the circuit.

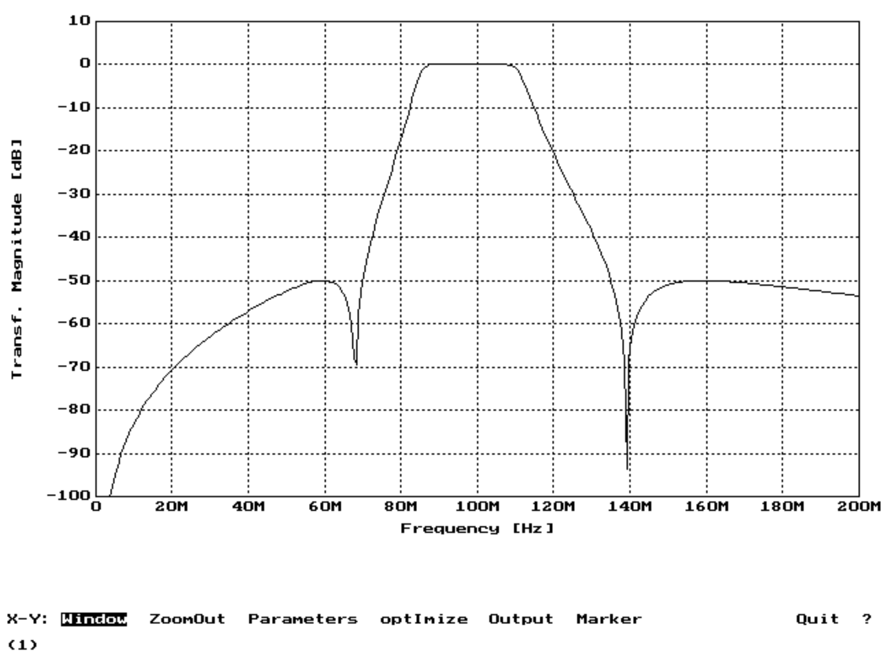
The circuit is displayed with the filter input at the top, the output at the bottom and the solid line on the left as the ground.

3.4 Analyzing Circuits

Let's take a look at the frequency response of the filter circuit we have designed:

- [Q] Back to the PASSIVE DESIGN menu.
- [M][A] Go to the CIRCUIT ANALYSIS menu.
- [X][X][X][↵]200M[↵] Upper limit X-axis = 200 MHz
- [O]300[↵] 300 calculated points
- [↵] Switch to the waveform plotting mode.

Once the screen display is complete, it will appear as follows:



Of course, this leaves a lot of room for improvement (limits, scaling etc.). Let's get back to our starting point:

- [Q][Q][Q][Q]   Back to main menu.

In the last step, we will save the filter data and program status for later use:

- [T][S]example1[↵]   The file is saved under Example1.QF.

This ends the tutorial example!

SUMMARY:

In this example, you have seen:

- How to operate the program;
- How to ask for help;
- How to specify a filter;
- How a complete circuit is designed;
- How to analyze a circuit's transmission characteristics;
- How fast *QuickFil* actually runs; and
- How easy it is to design an Elliptic filter.

Further examples are provided in the descriptions of the individual program functions (Chapter 5).

4 Operating Instructions

4.1 Selecting Menu Options

The *QuickFil* menu system is based on a hierarchical tree structure. Selecting a menu option will take you:

- to another menu;
- to an instruction; or
- to the appropriate data field.

The following key functions are valid throughout the whole program:

[ESC]	Interrupt key. Calculations are also stopped. For mouse users: The same function can be achieved by pressing both mouse buttons.
[?]	Invokes the on-line help system.
[Q]	Selects the quit command. Exits from a menu by maintaining all changes and returns to the previous menu.
[F1]	Calls up a list of files when the program expects the entry of a filename.

4.2 Selecting Options Using The Keyboard

You can select any menu option by pressing the first UPPER CASE letter in the particular menu entry (usually the first letter, but not always).

In most menu functions, each upper case letter occurs only once. If that letter is pressed, the appropriate command will be immediately executed.

For example:

To move from the main menu to the OPTIONS menu:

```
MAIN: Filtertype    Specification    Group_delay    Roots    polynomial_Analysis
      passive_Design     Transfer    Macro    Options                   Quit  ?
```

- Press [O]. The OPTIONS: menu will be activated.

In menus where the same letter applies to more than one function, the commands are executed as follows:

- Successive presses of the capital letter will alternate between the appropriate functions. To select the desired function, press Enter.

For Example:

You're in the APPROXIMATION menu and you want to select the Elliptic entry.

```
APPROXIMATION: Equal-ripple     Maximally-flat     Butterworth     Chebyshev
               Inv.-Chebychev   Elliptic                             Quit  ?
```

Procedure:

- [E] Highligth Elliptic
- [↵] Confirm selection

The items in the menu can also be selected with the [←], [→].

[Spacebar] or [Backspace] keys and confirmed with [↵].

4.3 Selecting Options Using The Mouse

To select menu functions with the mouse:

- Move the mouse pointer over the desired function.
- Click the left or right mouse button.

The function you selected will be executed immediately.

To back up one menu level:

- Press both the left and right mouse buttons simultaneously.

4.4 Dialog Fields

In many parts of the program, the dialog with the program takes place through fields in the screen. Highlighted entries can be selected directly by the user. Entries in normal video contain additional information and cannot be selected.

The highlighted entries can either be:

- numbers/texts that can be edited;
- functions that are executed after being selected; or
- the valid setting from a range of options.

4.5 Entering Data In Fields

Keystrokes Results:

[↵]	End and accept the entry.
[ESC]	End without accepting the entry.
[←], [→]	Move cursor one character to the left/right.
[Ctrl+←], [Ctrl+→]	Move cursor one word to the left/right.
[Home], [End]	Move cursor to the beginning/end of a string of characters.
[Backspace]	Delete character left of the cursor.
[Ctrl+Backspace]	Delete the whole entry in the input field.
[Ctrl+T]	Delete until the next blank.
[Del]	Delete character at the cursor position.
[Ins]	Switch from “Insert” to “Overwrite” mode.

Additional input options in numerical fields include:

min[↵]	Smallest possible value desired
max[↵]	Highest possible value desired
nn%[↵]	nn% of the presently shown value desired

Scaling abbreviations for numbers include:

a	Atto = 1.0E-18
f	Femto = 1.0E-15
p	Pico = 1.0E-12
n	Nano = 1.0E-9
u	Micro = 1.0E-6
m	Milli = 1.0E-3
k	Kilo = 1.0E+3
M	Mega = 1.0E+6
G	Giga = 1.0E+9
T	Tera = 1.0E+12
P	Peta = 1.0E+15
E	Exa = 1.0E+18

4.6 Entering Filenames

Some program functions require a filename to be entered. For these functions, you can enter filenames in two different ways:

- Direct entry of a filename in the corresponding field; or
- Selection of a filename from a list.

Filename extension note: *QuickFil* automatically adds the extension the program expects for the respective data format (see “Data Formats”). If you want to use another extension, you can enter it along with the filename.

Filename path note: If you enter the filename only, data will be transferred with the path specified in the OPTIONS menu. If you want to use another path, you can enter it along with the filename (e.g. a:\test\filter1.ini [↵]).

The scaling abbreviations differ from those used with the ISSPICE analog circuit simulation program.

Instead of entering a filename, you can call up a list of files and select the one you want.

To select a file using the keyboard:

- Press [F1] to call up the list of existing files.
- Select the file you want by entering the first letter or using the cursor keys.
- Press [↵] to confirm.

To select a file using the mouse:

- Place the mouse pointer in the filename field.
- Press the right mouse button;
- Then, either press the left mouse button for selection and Enter for confirmation; or
- Press the right mouse button for selection and simultaneous confirmation.

Options for listing files

Path names and wildcards such as “?” and “∗” can be used for listing files.

Examples of entries:

∗.∗[F1]	All files (no matter what extension) are listed in the directory as stated in the OPTIONS menu.
C:\QF\∗.FMT[F1]	All files are listed that have the extension FMT, and that are contained in the directory, as stated in the OPTIONS menu.

The first file displayed in the list on the screen is automatically copied to the input field. After confirming the filename, the data will be transferred.

4.7 On-Line Help Text

To invoke the on-line help text:

- Press “?”.

You can issue the following instructions from the menu line:

Resume

Return to the *QuickFil* menu screen.

Next_page

Display the next help screen.

Previous_page

Move back to the previous screen.

Contents

Open the HELP CONTENTS menu to obtain a list of all the help topics that you can access.

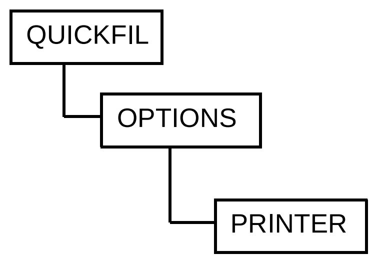
4.8 Output

QuickFil provides two types of printer output, text and graphics.

Text Output

This type of output is used in *QuickFil* for printing tables of data, such as specifications, zero positions, etc.

The printer specifications (type of font, margin, etc.) are set in the OPTIONS menu, either by entering the respective data field directly (mouse users), or by the menu items in the PRINTER submenu.



Interface printer/plotter

This item is used for selecting the desired interface for the output device.

IBM-char set

Determines if the printer is able to print the whole IBM character set (line symbols).

Lines/page

Determines the number of lines to begin a new page.

Left border

Determines the left margin in characters.

PostScript

Determines whether your printer is a postscript printer. Relevant only for text outputs.

Char/line

Determines the number of characters per line.

Init

Offers the option of sending an initialization sequence to your printer. This sequence is sent to the printer before each printout. Non printable control characters are provided in the form \nnn, with nnn being a three-digit decimal number (e.g. ESC = \027). Hint: Don’t enter a sequence if you have a postscript printer.

For example:

Control sequence for the HP laser printer IID for 10 pt Courier font is:
\027(10U\027(s0p12.00h10.0u0s0b3T

4.9 Graphic Output

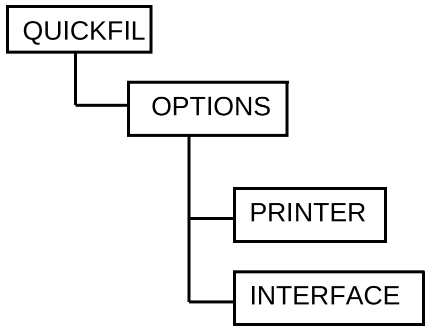
This type of output is used in *QuickFil* for printouts of the filter analysis results.

As different output formats are required for the individual printers, *QuickFil* first “translates” the respective graphics data into the printer language, and then transmits the data through the predefined interface. The program contains its own conversion software for this purpose.

The printer settings (type of printer, desired size of the drawing, etc.) are made in the PRINTER OPTIONS menu (from the menu OUTPUT GRAPHIC, select in the graphics section). For more details, please refer to the “Output Graphic” section.

4.10 Program Setup

QuickFil allows you to set certain program options. The program automatically switches to these settings at start-up. The respective settings are made in the OPTIONS menu. For settings activated from the keyboard (without mouse support), there are additional submenus.



For example, the following settings can be programmed:

- The DOS path where data is searched/saved. (If no alternate path has been added to the filename.)
- Interface parameters for the peripheral output devices (printer, plotters, etc.).
- The type of decimal point used (./,).
- Number groupings (yes/no).

The OPTIONS menu also tells how much memory is left and gives the precise version number of this program.

The entries in the OPTIONS menu are:

Drive, Path

This entry contains the path to the DOS directory where data files will be saved. (If an alternate path was not provided when the filename was entered.)

Interface Printer

This sets the printer interface parameters.

Interface Plotter

This sets the plotter interface parameters.

If you would like to print your diagrams through another interface (e.g. IEC interface), you can proceed as follows:

- Instead of printing the output on a printer, save the graphics data to a file using the HPGL format. (menu item File in the OUTPUT GRAPHIC menu)
- Exit from *QuickFil* and plot the graphics from the DOS prompt (copy the contents of the file to the respective plotter port).

Decimal-sign

Determines whether the "." or "," is used as the decimal point.

Grouping

Determines whether "longer" numbers in fixed point displays are to be divided into groups of three for clarity. When selecting the groups of numbers, the decimal separating point is used that is inverse to the selected decimal sign.

Printer Setup

Determines the printer settings for text output.

Parameters Of Serial Port(s)

Determines the parameters for the serial interface(s) COM1/ COM2:. If the mouse is installed on one of these ports, the settings cannot be altered.

The following parameters can be selected:

Baud rate: 110, 150, 300, 600, 1200, 2400, 4800, 9600

Parity: No, Even, Odd

Data bits: 7, 8

Stop bits: 1, 2

Note: Since the *QuickFil* is a DOS Program and it will communicate with the serial ports directly, you will have problems in Windows, especially for Windows NT, Windows 2000 and Windows XP. In Windows, all serial ports are normally switched off.

QuickFil automatically determines which ports are present in the computer.

The OPTIONS menu also contains the Prog_Reset menu item through which the program can be reset to a defined initial state.

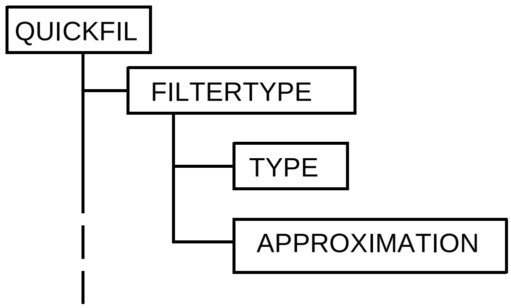
You should be absolutely sure when activating this function because all previously entered information will be lost!

Note: The saving/loading of program status (program setups with filter data, graphic parameter, etc.) is possible through the TRANSFER menu item in the main menu.

5 Program Functions

5.1 Specifications

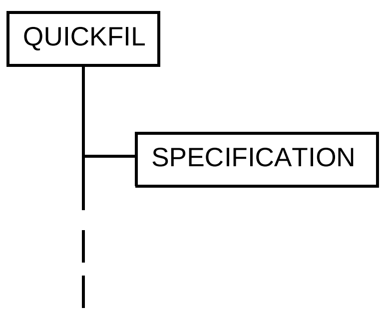
The desired type of filter and the approximation are selected in the FILTERTYPE menu. The FILTERTYPE menu is selected from the main menu.



After determining the desired filter type, various filter characteristics are entered in the SPECIFICATION menu.

5.2 Standard Approximations

For standard filter approximations(Butterworth, Chebychev, inverse Chebychev, Elliptic), specifications are made in the SPECIFICATION menu.



Depending on the filter type, the following menu will be shown:

Lowpass or highpass filters:

```
        SPECIFICATIONS to :  Elliptic (Cauer) - lowpass filter
(O)

(A)  Passband edge frequency          :          1.000 000 kHz

(C)  Stopband edge frequency          :          1.200 000 kHz

(E)  Passband bandedge loss           :           1.000 000 dB
(F)  Passband bandedge return loss    :                6.87 dB
(R)  Passband reflection factor       :               45.35 %
(G)  Stopband loss                    :               94.35 dB   ◄

(H)  Filter degree                    :                     10
(I)  Case                  (a, b, c)  :                      b
(J)  Variable value (A,C,E,F,G,H,R)   :                      G

     3dB edge frequency               :          1.004 479 kHz

     Filter quality                   :                  41.42
SPECIFICATION: A  C  E  F  R  G  H  I  J  New  cOmment  fiLe  Printer  Quit  ?
```

Bandpass or bandstop filters:

```
        SPECIFICATIONS to :  Elliptic (Cauer) - bandpass filter
(O)

(A)  Lower passband edge frequency    :          1.000 000 kHz
(B)  Upper passband edge frequency    :          2.000 000 kHz
(C)  Lower stopband edge frequency    :        800.000 000  Hz
(D)  Upper stopband edge frequency    :          2.500 000 kHz

(E)  Passband bandedge loss           :           1.000 000 dB
(F)  Passband bandedge return loss    :                6.87 dB
(R)  Passband reflection factor       :               45.35 %
(G)  Stopband loss                    :               74.87 dB   ◄

(H)  Filter degree                    :                     12
(I)  Case                  (a, b, c)  :                      b
(J)  Variable value (A,B,C,D,E,F,G,H,R) :                    G

     Lower 3dB edge frequency         :        993.696 894  Hz
     Upper 3dB edge frequency         :          2.012 686 kHz
     Filter quality                   :                  29.43
SPECIFICATION: A  B  C  D  E  F  R  G  H  I  J   New  cOmment  fiLe   Printer
               freqUencyrepres.  bandwithrepreS.  reL.bandwithrepres.   Quit  ?
```

Selections are made by pressing the appropriate key letter which corresponds to the field on the screen that you wish to edit. Depending on the filter type, only the relevant settings will be available for entry.

Following menu items are available:

(Lower) passband edge frequency

The (lower) frequency at which the loss reaches the specified passband loss (see diagram at the end of this section).

Upper passband edge frequency

The upper frequency at which the loss reaches the specified passband loss (for bandpass and bandstop only, see diagram at the end of this section).

(Lower) stopband edge frequency

The (lower) frequency at which the loss reaches the specified stopband loss (see diagram at the end of this section).

Upper stopband edge frequency

The upper frequency at which the loss reaches the specified stopband loss (for bandpass and bandstop only, see diagram at the end of this section).

As an alternative to the individual edge frequency values, *QuickFil* also allows you to enter data in the bandwidth/relative display. (Only for the standard approximations of bandpass or bandstop filters using symmetrical responses.)

Change through the menu items ***bandwidthrepreS***. and reL.bandwidthrepres.; with ***freqUencyrepres***. You can return to the frequency display.

The center frequency, stated in the bandwidth display, refers to the geometrical center with reference to the logarithmically applied frequency axis.

Passband bandedge loss

Maximum loss in the passband (see diagram at the end of this section).

Passband bandedge return loss

This entry is only relevant to passive LC filters (reactance filters). The return loss is the reflection factor in "dB" Format.

Examples:

- r = 1 return loss = 0 dB
- r = 0.1 return loss = 20dB

Return loss is directly coupled to passband attenuation.

Passband reflection factor

Ratio of the amount of the reflected signal to the amount of the input signal, in percent. The reflection factor can be calculated from the input impedance, directly (see appendix J).

The reflection factor is directly coupled to the passband and return loss.

Stopband loss

Minimum loss in the stopband (see diagram at the end of this section).

Filter degree

The number of attenuation peaks of the filter circuit (peaks where s -> infinity included). The degree of a filter is an indication of the number of components required.

Case

For even degree lowpass and highpass filters, and for bandpass and bandstop filters whose degree is divisible by four, there are frequency transformations available which are represented by cases.

a Case "a" is available for Invers Chebychev and for Elliptical filters. Case "a" means that no frequency transformation is used. Invers Chebychev and Elliptical filters are not realizable if it’s an even degree lowpass/highpass, or a bandpass or bandstop filter whose degree is divisible by four, since there is no transmission zero at zero and no transmission zero at infinity. Therefore, no passive ladder filter can be realized. By using a frequency transformation, you can get a realizable filter.

b Case "b" is available for Chebychev, Invers Chebychev and for Elliptical filters. Case "b" guarantees that there is at least one transmission zero at zero or at infinity. For Chebychev filters case "b" is the standard not transformed filter. For Invers Chebychev and for Elliptical, the largest transmission zero of the even degree prototype lowpass is transformed to infinity. For even degree filters and Chebychev or Elliptical approximation, input and output resistance are different.

c Case "c" is available for Chebychev and for Elliptical filters. Case "c" guarantees that there is at least one transmission zero at zero or at infinity, and the input and output resistance of the passive filter are equal. The largest transmission zero of the even degree prototype lowpass is transformed to infinity, and the lowest reflection zero is transformed to zero.

For more information, see the section “Case, terminating resistance ratio”.

Variable value

You can use this field to specify which characteristic of a filter *QuickFil* can, or should change, if any of the screen inputs changes (e.g. during editing the specifications). The variable value is shown by an arrow at the end of the editing field.

The screen contains the following additional information:

3dB edge frequency

The frequency at which the passband loss reaches -3 dB of its maximum value (precisely: 3.0103 dB). This value is calculated for the user as further information. It cannot be changed like parameters of input fields directly.

Filter quality

Maximum pole quality of the zeros of polynomial e(s). This value is calculated for the user as further information. It cannot be changed like parameters of input fields directly. For passive LC-filters, the inductors quality should be designed with a factor of 3 to 5 higher than the stated filter quality when realizing the filter.

The following options are also available:

New

Deletes all data contained on the screen in preparation for new input.

cOmment

QuickFil automatically labels all outputs (roots, circuits, graphs) with comments that you can enter via this option.

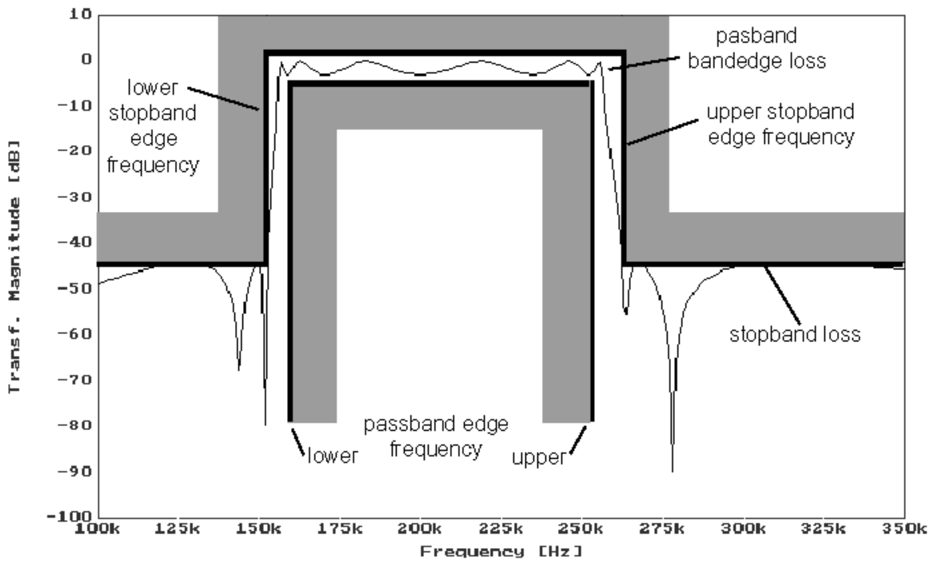
fiLe

Saves the entered filter specifications in a file. The default extension is .SPZ. For more information on this data format, see the section on “Filter Specification Protocol”.

For more details, please refer to the section on “Output”.

Printer

Prints the filter specifications.



You can specify the filter demands by simply editing the input fields (pressing the letter, defined at the beginning), using the cursor key to skip to the next input field, or by simply using the mouse. If all input fields are specified, you can change the values of the input fields. Since all possible specifications are dependent on each other, every change will produce changes to other fields. The variable field is shown by an arrow at the end of the field. You can specify the variable field which should change if you edit one input field by the special menu item ***Variable value,*** or by simply clicking on the field where you would like to have the "variable arrow".

The possibility to adjust the filter parameters to the users demands is one of the main and unique features of *QuickFil*.

5.3 Bessel Lowpass Approximations

Bessel filters are lowpass filters which have a flat group delay response. Since the transformations from lowpass to highpass, bandpass and bandstop filters do not preserve the flat group delay response, only lowpass filters make sense. There are two approximations available: Bessel and Modified Bessel.

```
          SPECIFICATIONS to: Modified Bessel - Lowpass - Filter

 (O)

 (A)  3dB edge frequency           :              1.000 000 kHz

 (B)  Group delay at zero          :            401.422 536 µs

 (C)  Stopband edge frequency      :              3.683 590 kHz       ◄

 (D)  Stopband loss                :                   73.26 dB

 (E)  Filter degree                :                      10
 (F)  Case               (a, b)    :                       b
 (G)  Variable value  (A, C, D)    :                       C

      Filter quality               :                    1.42
      Frequency at 90% group delay :              3.376 955 kHz

SPECIFICATION: A  B  C  D  E  F  G  New  cOmment    fiLe  Printer     Quit  ?
```

Selections are made by pressing the appropriate key letter corresponding to the field on the screen that you would like to edit. Depending on the type of filter, only the relevant settings will be available for entry.

Following menu items are available:

3dB edge frequency

The frequency at which the loss reaches -3 dB of its maximum value (precisely: 3.0103 dB). Here you can specify a kind of cut off frequency of the lowpass filter.

Group delay at zero

The group delay at the frequency zero. Here, you can specify the group delay at the frequency zero. The group delay is maximally flat and will decrease with increasing frequency.

Stopband edge frequency

The frequency at which the loss reaches the specified stopband loss.

Stopband loss

Minimum loss in the stopband.

Filter degree

The number of attenuation peaks of the filter circuit (peaks where s -> infinity included). The degree of a filter is an indication of the number of components required.

Case

For Modified Bessel approximations and even degree filters, we differentiate between various “cases”.

a This is the basic approximation without any frequency transformation. Since even degree Modified Bessel lowpass filters have no transmission zeros at infinity, the required transmission characteristic cannot be realized by a passive ladder filter. This feature is included for analysis possibilities.

b The largest transmission zero of the Modified Bessel lowpass of even degree is transformed to infinity. Therefore, there is at least one transmission zero at infinity and the transfer function can be realized by a passive ladder circuit.

For more information, see the section “Case, terminating resistance ratio”.

Variable value

You can use this field to specify which characteristic of a filter *QuickFil* can, or should change if any of the screen inputs change (e.g. during editing the specifications). The variable value is shown by an arrow at the end of the editing field.

The screen contains the following additional information:

Filter quality

Maximum pole quality of the zeros of polynomial e(s). This value is calculated for the user as a further information. It cannot be changed like parameters of input fields directly. For passive LC-filters, the inductors quality should be designed with a factor of 3 to 5 higher than the stated filter quality when realizing the filter.

Frequency at 90% group delay

The frequency at which the group delay will be reduced to 90% of the delay at zero frequency. This item is only for information of the user so he can estimate the bandwidth of the flat delay.

The following options are also available:

New

Deletes all data contained on the screen in preparation for new input.

cOmment

QuickFil automatically labels all outputs (roots, circuits, graphs) with comments that you can enter via this option.

fiLe

Saves the entered filter specifications in a file. The default extension is .SPZ. For more information on this data format, see the section on “Filter Specification Protocol”.

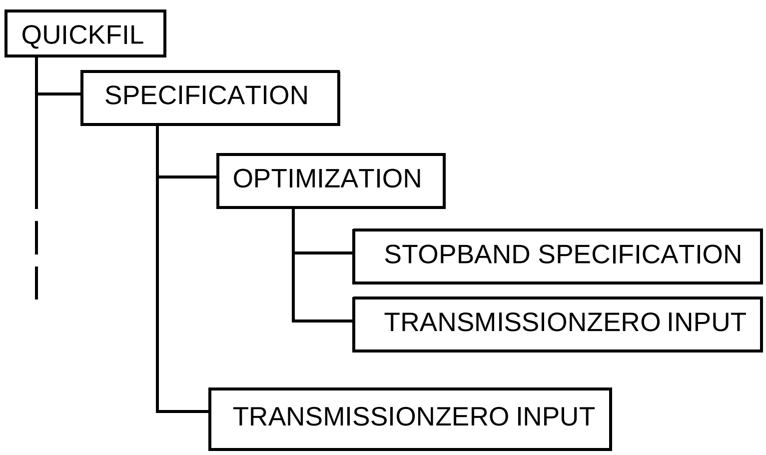
For more details, please refer to the section on “Output”.

Printer

Prints the filter specifications.

5.4 General Amplitude Approximations

When selecting a general amplitude approximation (equal ripple or maximally flat), further specifications for the filter are made in the SPECIFICATION menu and its respective submenus:



5.5 SPECIFICATION Menu (for general amplitude approx.)

Depending on the type of the filter, the following menu will be shown:

Lowpass or highpass filters:

```
            SPECIFICATIONS to :  Equal ripple - lowpass filter
  (O)

  (A)  Passband edge frequency          :            1.000 000 kHz

  (E)  Passband bandedge loss           :             1.000 000 dB
  (F)  Passband bandedge return loss    :                  6.87 dB
  (R)  Passband reflection factor       :                 45.35 %

       Transm. zeros at zero      :   0 | Fixed transm. zero pairs:   0
  (H)  Transm. zeros at infinity  :   2 | Var.  transm. zero pairs:   4 opt.

  (V)  Loss at zero                     :             0.000 000 dB
  (W)  Resistance ratio                 :                1.000 000

       Filter degree                    :                       10

       3dB edge frequency               :            1.004 594 kHz

       Filter quality                   :                    40.36
SPECIFICATION: A  E  F  R  H  V  W                              qualitY
               Stopbandopt. fiXtrans.zeros  cOmment  New  fiLe Printer  Quit  ?
```

Bandpass filters:

```
           SPECIFICATIONS to :  Equal ripple - bandpass filter
  (O)

       Kind of bandpass                 :                parametric
  (M)  Parameter                        :            1.414 214 kHz

  (A)  Lower passband edge frequency    :            1.000 000 kHz
  (B)  Upper passband edge frequency    :            2.000 000 kHz

  (E)  Passband bandedge loss           :             1.000 000 dB
  (F)  Passband bandedge return loss    :                  6.87 dB
  (R)  Passband reflection factor       :                 45.35 %

  (G)  Transm. zeros at zero      :   4 | Fixed transm. zero pairs:   1
  (H)  Transm. zeros at infinity  :   2 | Var. transm. zero pairs :   4 opt.

       Filter degree                    :                       16
       Lower 3dB edge frequency   (≈)   :          993.788 055  Hz
       Upper 3dB edge frequency   (≈)   :            2.006 880 kHz
       Filter quality                   :                    54.11
SPECIFICATION: M  A  B  E  F  R  G  H   qualitY   Conventional    Defaultparam.
               Stopbandopt. fiXtrans.zeros  cOmment  New  fiLe Printer  Quit  ?
```

Bandstop filters:

```
          SPECIFICATIONS to :  Equal ripple - bandstop filter
 (O)

 (A)  Lower passband edge frequency    :              1.000 000 kHz
 (B)  Upper passband edge frequency    :              2.000 000 kHz

 (E)  Passband bandedge loss           :              0.500 000 dB
 (F)  Passband bandedge return loss    :                   9.64 dB
 (R)  Passband reflection factor       :                  32.98 %

 (M)  Transm. zero pairs at centre:  2 | Fixed transm. zero pairs:   0
                                       | Var. transm. zero pairs :   4

 (V)  Loss at zero                     :              0.100 000 dB
 (W)  Resistance ratio                 :              1.355 361

      Filter degree                    :                     12
      Lower 3dB edge frequency         :              1.011 710 kHz
      Upper 3dB edge frequency         :              1.976 852 kHz
      Filter quality                   :                  22.27

SPECIFICATION: A  B  E  F  R  M  V  W                            qualitY
               Stopbandopt. fiXtrans.zeros  cOmment  New  fiLe Printer  Quit  ?
```

Once you have entered the filter type and the approximation (FILTERTYPE menu), use this menu to specify the filter. Selecting one of the letters in the menu will take you to the corresponding field on the screen. The program will only allow you to enter the characteristics that apply to the selected filter type. The available characteristics are:

(Lower) passband edge frequency

The (lower) frequency at which the loss reaches the specified passband loss.

Upper passband edge frequency

The upper frequency at which the loss reaches the specified passband loss (only for Bandpass and Bandstop filters).

Note: After completing the specification for bandstop filters, it is only possible to change one of the edge frequencies. When one edge frequency is changed, the other edge frequency is automatically changed while the center frequency is kept constant. This is due to the fact that bandstop filters are always symmetrical.

Note: If you would like to specify a very narrow band bandpass or bandstop filters, there are limits in the program to avoid wrong data and numerical problems. But if you need this very narrow band filters, you can change that limit in the *QuickFil* Defaults File (QF.DEF) (see Appendix)

Parameter

This parameter is only available for parametric bandpass filters. In bandpass filters, you have to distinguish between conventional or parametric types of responses (change in Parametric/Conventional menu item). If the parametric form is selected, the screen will contain an input field for the parameter. This gives you more freedom for dimensioning. (For more details please refer to the section “Parametric Bandpass Filter” and the explanations in the Appendix “Conventional/Parametric Bandpass Filters”).

By selecting the ***Defaultparam*** menu item you may accept the default parameter provided by the program. It is the geometric mean of the lower and upper passband edge frequencies.

A negative parameter value can also be entered. However, it will only have an influence on the structure of the circuit in parametric filters of odd degree. Passband bandedge loss

Passband bandedge loss

Maximum allowable loss in passband.

Passband bandedge return loss

This entry is relevant only to passive LC filters (reactance filters). The return loss is the reflection factor in "dB" Format.

Examples:

• r = 1	return loss = 0 dB
• r = 0.1	return loss = 20dB

Passband reflection factor

Ratio of the amount of the reflected signal to the amount of the input signal in percent. The reflection factor can be calculated from the input impedance, directly (see appendix J).

The reflection factor is directly coupled to the passband and return loss.

Transm. zeros at zero/infinity (Bandstop: Transm. zero pairs at center)

The number of transmission zeros desired can be entered here in the “extreme” frequencies, depending on the filter type. The entries are understood as fixed transmission zeros.

Default settings:

Lowpass: 1 transmission zero at infinity

Highpass: 1 transmission zero at zero

Bandpass: 1 transmission zero at zero and one at infinity

Bandstop: 1 pair of transmission zeros at center frequency

When entering the transmission zeros at zero in conventional bandpass filters, the number of transmission zeros at infinity is corrected in such a way that the filter always has an even degree, since odd degree conventional bandpass filters don’t exist. Parametric bandpass filters require at least one transmission zero at zero and one at infinity.

How the number of fixed transmission zeros can be determined in a given circuit is discussed in the “Design With User-DefinedCircuits” section.

Note: If a sufficient number of transmission zeros is set at zero and/or infinity, for achieving the minimum losses (with no fixed or variable transmission zeros in the finite stopband), you will receive a filter with a monotonic loss curve in the stopband.

Loss at zero/infinity

This parameter is only available for equal ripple approximation but not for bandpass filters. For lowpass and highpass filters, the degree of the filter has to be even. For bandstop filters the degree is divisible by four. You can adjust the resistance ratio of the ladder circuit by this parameter. If you would like to have the best filter for the given limits of loss in the passband, please choose the same value as you took for the passband bandedge loss. If it is important to have a passive ladder filter which has the same terminating resistance at the input and at the output, please take the value 0 dB.

For more details please refer to the section on “Equal Ripple Approximations”.

Resistance ratio

This parameter is only available for equal ripple approximation but not for bandpass filters. For lowpass and highpass filters, the degree of the filter has to be even. For bandstop filters the degree is divisible by four. Here you can explicitly adjust the resistance ratio of the passive ladder circuit. The "resistance ratio" is directly coupled with the "loss at zero/infinity".

For more details please refer to the section on “Equal Ripple Approximations”.

For a better overview of the overall filter specifications, the following data is also available:

Fixed transm. zero pairs

This entry informs you about how many fixed pairs of transmission zeros (these are notch frequencies) you have provided with finite frequencies (menu item ***fiXtrans.zeros***).

Var. transm. zero pairs

This entry informs you how many variable pairs of transmission zeros are within the finite stopband. If the position of the variable pairs of transmission zeros was optimized by *QuickFil* (OPTIMIZATION menu), the remark opt. can be seen in this screen (field directly behind their number).

Note: *QuickFil* exclusively deals with symmetrical bandstops. The number of pairs of transmission zeros is always even in the finite frequency range because there is no theory available for bandstop filters, which are not symmetrical and can be realized by ladder filters.

Filter degree

The degree is the number of attenuation peaks of the circuit (the transmisssion zeros at zero and at infinity are also counted). Beware: Pairs of transmission zeros always mean two transmission zeros. The degree of a filter is an indication for the number of components required. Maximum degree: 50

The entry in this screen depends on the settings in the OPTIMIZATION: menu.

Note: In conventional bandpass and bandstop filters the degree must be even.

3dB edge frequency

Provides the setting at which frequency/frequencies the loss passes the 3dB edge (precisely: 3.0103 dB).

Note: The calculation of the 3dB limits after changing the data requires a certain amount of time. In higher degrees, however, waiting periods can become quite lengthy. Therefore, it is also possible to turn off the calculation of the 3dB limits (For more details see the Appendix on the *QuickFil* Defaults File (QF.DEF)). The calculation will take place when the filter quality, if desired, is to be calculated.

Filter quality

Maximum pole quality of the zeros of polynomial e(s). This value is calculated for the user as a further information. It cannot be changed like parameters of input fields directly. For passive LC-filters the inductors quality should be designed with a factor of 3 to 5 higher than the stated filter quality when realizing the filter.

Note: The filter quality is only calculated if the ***qualitY*** menu item is selected. The calculation of the filter quality requires the calculation of the whole filter characteristics and will take time. If you leave the specification menu and enter it again, the calculation of the filter characteristics is calculated too and you can see the filter quality on the screen.

Stopbandopt.

Activates the OPTIMIZATION menu. Here, the required minimum loss curve in the stopband (entry of the tolerance scheme) is specified and the optimization of the position of the transmission zeros lying in the finite stopband is performed. For further details please refer to OPTIMIZATION: Menu section.

fiXtrans.zeros

Activates the TRANSMISSIONZERO INPUT submenu for entering the position of fixed transmission zeros. These will also be considered when optimizing the position of the variable transmission zeros in the OPTIMIZATION menu. A practical application would be the suppression of pilot frequencies. For further details, please refer to the “TRANSMISSONZERO INPUT” section.

cOmment

QuickFil automatically labels all outputs (roots, circuits, diagrams) with comments which you can enter via this option, e.g. If you specify a project number, you will always know which printout belongs to which project.

New

Clears the screen for the specification of a new filter.

fiLe

Saves the filter specifications in a file. Default extension: .SPZ

Printer

Sends the filter specifications to a printer.

5.6 OPTIMIZATION Menu

```
                          stopband optimization

 (A) Var. transm. zero pairs in lower stopband  (proposal :  2 ) :    1
 (B) Var. transm. zero pairs in upper stopband  (proposal :  4 ) :    4
     Variable transmission zero pairs           (proposal :  6 ) :    5

 (C) Transmission zeros at zero              :                        3
 (D) Transmission zeros at infinity          :                        1
     Filter degree                           :                       14

     Minimum of loss reserve                 :                  8.73 dB
     Difference of the minima                :                  0.00 dB

 (G) Limit of the optimization               :                  0.10 dB

OPTIMIZATION: A  B  C  D  G      Lower_stopband_spec.    Upper_stopband_spec.
              Variable-transmissionzeros   Optimization   Iteration     Quit  ?
```

In this menu you can enter the a curve of the minimum losses in the stopband (tolerance scheme). You can also optimize the number and position of the transmission zeros which are arranged within the stopband, allowing the required stopband loss curve to be reached (modified Remez algorithm).

General Sequence Of Entries

The “classical” sequence of entries in this program takes place in the following order:

Entry Of The Tolerance Scheme

From the OPTIMIZATION menu...

For lowpass, highpass and bandstop filters, use the sTopband-specification menu item.

For bandpass filters, use the Lower/Upper_stopband-spec menu item.

This brings you to the STOPBAND SPECIFICATION menu.

The tolerance scheme has a stepped profile. In the case of an individual curve, the tolerance scheme can be approximated in small steps. The tolerance scheme serves as a unilateral edge, i.e. the optimization tries, through the optimal arrangement of the positions of the transmission zeros, to move the real loss curve at each point as far away as possible from the tolerance scheme.

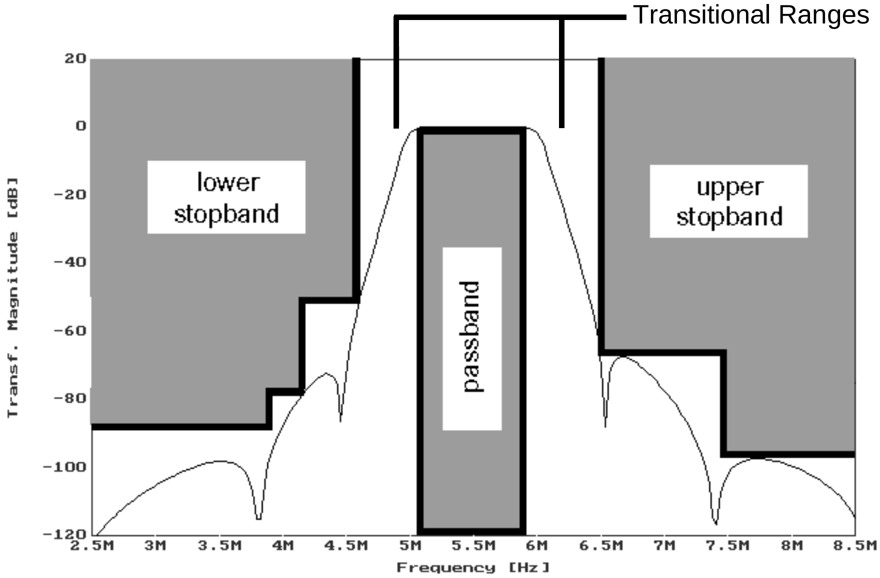
The tolerance scheme is only valid in the stopband and must be defined before the optimization can proceed. In bandpass filters, you have to distinguish between the upper and lower stopbands. If no particular requirements are set on one of the two bands, the optimization can only be carried out with one specified stopband (e.g. only the lower tolerance scheme).

Note: A maximum of fifty steps is allowed for the tolerance scheme.

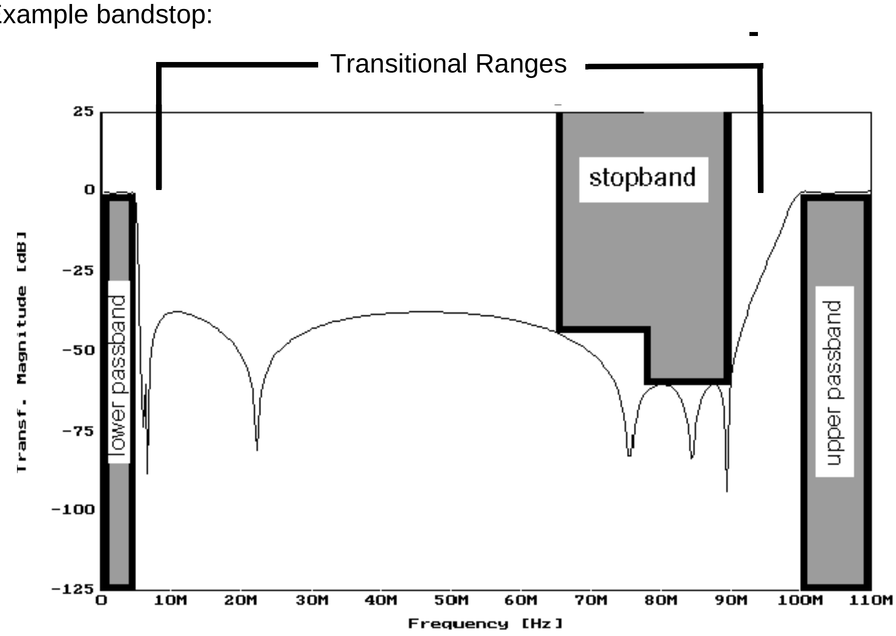
QuickFil does not care about the filter's loss outside of the stopband frequency range (transitional range between the stopband and the passband) and this range will not be taken into account for determining the loss reserve.

Explanation of the terms

Example bandpass:



Example bandstop:



The specified tolerance scheme is automatically saved. It can be included in the graphical display of the loss curve in dB, along with the actual filter response. For more details, please refer to “Circuit Analysis” section.

Automatic Initialization Of The Variable transmission zeros

When exiting the STOPBAND SPECIFICATION menu, *QuickFil* will make a proposal for the required number and position of the variable transmission zeros that will achieve the required stopband loss curve in the respective frequency range. These values are the initial values that will be used for the subsequent optimization process.

Note: The evaluation does not take into account the number of already specified fixed transmission zeros at zero, infinity, and the center frequency (bandstop filters). It also does not take into account the pairs of fixed transmission zeros in the finite frequency range already entered by the user.

More details on how *QuickFil* establishes the number and position of the transmission zeros for the initial values of the optimization can be found in the “Determination Of Variable Transmission zero Pairs” section.

Changing the Number of Proposed Transmission Zeros

The determined number of variable transmission zeros will be stated in the OPTIMIZATION screen. This value can be increased or decreased, as desired, before the optimization is started. A new initialization is made whenever the number of transmission zeros is changed in this manner.

Note: From this menu it is also possible to change the number of transmission zeros at zero, infinity, or the center frequency (in bandstop filters). When making these changes, the entry in the SPECIFICATION menu is also altered. If no variable transmission zeros are entered, the loss reserve can, nevertheless, be determined by selecting ***Optimization*** or ***Iteration***. If there are also no fixed transmission zeros, the loss curve in the stopband is monotonic.

Viewing and Editing Variable Transmission Zeros

For special cases, *QuickFil* allows you to preset the individual positions of the variable transmission zeros in the stopband, or to modify the result of the program's initialization. In order to perform these functions, enter the TRANSMISSIONZERO INPUT menu through the ***Variable_transmissionzeros*** menu function. *QuickFil* presumes that the user would like to pursue a particular goal, for example, changing the initial condition for the optimization. Hence, there is no automatic initialization of the transmission zeros afterwards. For more details on the TRANSMISSIONZERO INPUT menu, please refer to the “Transmissionzero Input” section.

Note: Variable transmission zeros can only be positioned within the range of the tolerance scheme.

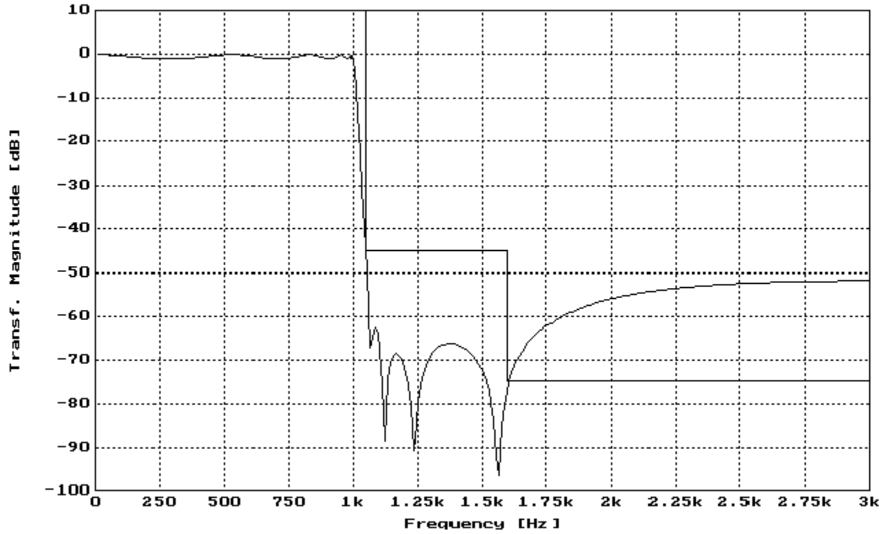
Optimization of the Transmission Zeros

After having finished determining the number and the initial positions of the variable transmission zeros in the stopband, the optimization can be carried out by the program. The optimization is started by the OPTIMIZATION menu function.

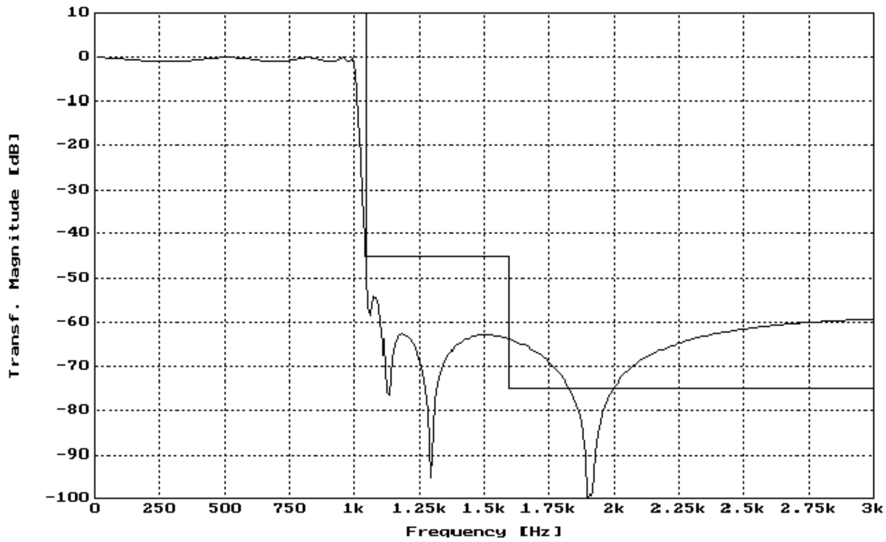
The variable transmission zeros lead to "arcs” in the loss curve. Each of these “arcs” comprises a minimum distance to the predetermined tolerance scheme (in most cases the “cusp point” of the arc). During the optimization the variable transmission zeros are moved in iterative steps in the stopband in such a way that all the minimum distances have the same distance and that the difference between the highest and the lowest minimal distance converges towards zero. The program proposes a difference of 0.1 dB as the stop limit for the optimization. However, this value can be changed individually, if desired. If after concluding the optimization, better values can be achieved, further single iterations can be carried out through the Iteration menu item.

Example for a lowpass:

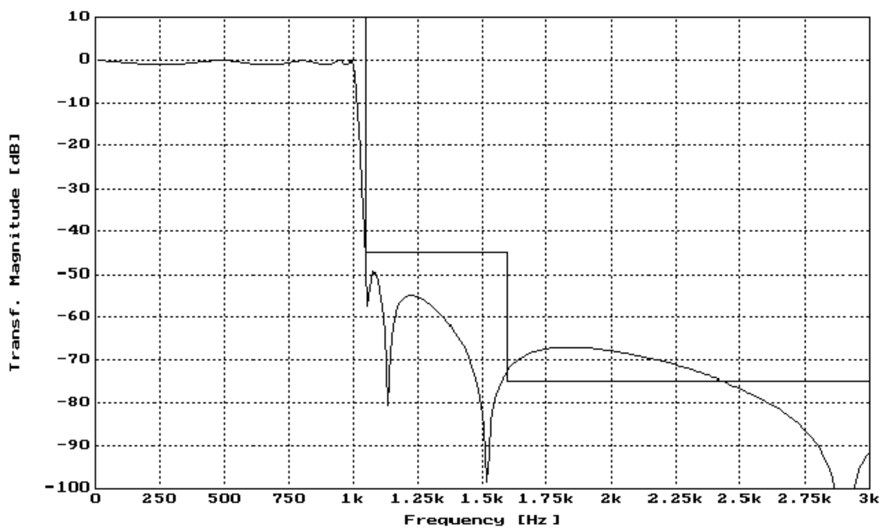
1. Arrangement after the initialization:



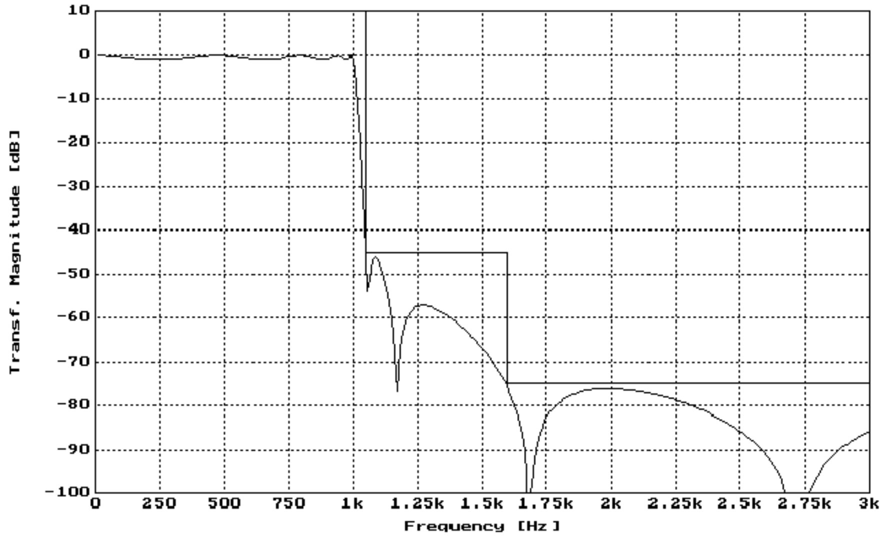
2. After the first iteration step:



3. After the third iteration step:



4. After completing the optimization (6 iterations):



The optimization may lead to a loss reserve that is greater than required. If this is the case, then either the degree of the filter can be reduced by decreasing the number of variable transmission zeros, or a reduction in the passband loss can be made. One can usually increase the return loss by the approximate value of the reserve. In both cases, however, it is necessary to carry out a new optimization and check whether the change makes the loss fall below the required value. This would be indicated by a negative sign in the value of the minimum loss reserve after completing the optimization.

Bandpass filters

During the optimization of certain bandpass filters, it is possible for a pair of transmission zeros to assume real values instead of the required imaginary values. You can recognize this by the number of pairs of transmisssion zeros in the upper and lower stopband. The total of which does not result in the set number of variable pairs of transmission zeros. In any case, the optimization can maximally yield one real pair of transmission zeros.

However, there are cases where such real pairs of transmission zeros occur during the optimization and then disappear. It is also possible that such a real pair of transmission zeros is required for the optimum solution. But, with a passive circuit of the form presently used, it is not possible to realize it.

In order to remedy this problem, you can displace the real pair of transmission zeros to zero or infinity, or split them to zero and infinity. This solution is successively requested by *QuickFil* in such a case. If you do not agree with any proposal, the optimization is ignored and *QuickFil* will use the values provided before the optimization as the variable transmission zeros.

It is also possible to make only one iteration. Again, the same problems with real pairs of transmission zeros might occur. Variable transmission zeros can also be moved back and forth between the upper and lower stopband during the optimization.

5.7 STOPBAND SPECIFICATION Menu

In this menu, the stopband settings for general equal ripple and maximally flat approximations are entered.

First entry

The edge frequencies of the stopband specifications are numbered consecutively and listed according to frequency.

Proceed as follows for the first entry:

- Select the Add menu item.
- Confirm the proposed number in the range.

The following steps will depend on the type of filter being synthesized:

Lowpass

- Enter the frequency from which a certain stopband loss is to be achieved.
- Then, enter the desired loss value. The range ends at infinity if no further range is specified afterwards or at the beginning of the next range.

Highpass

- Enter the frequency up to which a specific stopband loss is desired.
- Then, enter the desired loss value. The range begins at zero, if a range was not entered before or from the end of a previous range.

Bandpass

Lower stopband: like highpass

Upper stopband: like lowpass

Bandstop

- Enter the frequency from which a specific loss is to be reached, and then the frequency up from which this is desired.
- Then, enter the desired loss value. In the next range, *QuickFil* will use the upper edge frequency of the last range as the “initial frequency” and expects the entry of the frequency up to which the new loss is set. Then, enter the loss value for this range.

Note: The number of frequency ranges that can be entered for lowpass, highpass and bandpass filters is 50 (bandstop 49).

Change stopband settings

To change individual values:

- Click on the fields in the screen with your mouse (scroll with the mouse pointer on the screen edge if the table is longer);

or

- Select the Edit menu item, then select the desired value with up and down arrows and make the pertinent changes.

Insertion of a new range

A range with another loss can also be “inserted” afterwards by entering the number of the range within which a new range is to be specified. *QuickFil* will then allow you to enter frequency values which lie within the range that has already been specified.

Delete

Deletes individual ranges by entering the respective number.

New

Deletes all stopband settings shown on the screen.

When exiting the menu, the transmission zeros are initialized first, i.e. *QuickFil* makes a first proposal for the number and positions of the variable transmission in the stopband. This serves as the initial setting for the following optimization. For more details please refer to the OPTIMIZATION menu section.

5.8 TRANSMISSIONZERO INPUT Menu

This menu is used for entering and changing fixed, as well as, variable transmission zeros.

In lowpass, highpass and bandstop filters, the entered transmission zeros are sorted in two columns with rising frequencies. In bandpass filters, the transmission zeros of the lower stopband are entered in the left column and the transmission zeros of the upper stopband in the right column.

Bandpass filters: If there is insufficient space in the column for all of the stopband transmission zeros, the sorting arrangement according to the upper and lower stopband is dropped. The frequencies will also be sorted according to rising frequencies in the two columns. For optical separation, a blank line is inserted between the transmission zeros of the upper and lower stopband.

Bandstop filters: In bandstop filters, two transmission zeros are always treated symmetrically to the center frequency. It is only possible to have an even number of pairs of transmission zeros. When making entries or changes, the complementary frequency belonging to a specific transmission zero is also updated automatically.

Add

Used to add a transmission zero to the existing list.

Edit

To edit a specific field:

- Select the field with the up or down arrows and enter the data.
- Then, confirm the entry with [↵].

Under certain circumstances, the table might be sorted again (in accordance with rising frequencies).

New

Deletes all entries in preparation for a new filter.

To delete individual transmission zero frequencies:

- Switch to the editing mode using Edit menu item.
- Select the transmission zero to be deleted using the down arrow.
- Delete the entry in the field using the control backspace key combination.
- Then, confirm the entry with [↵].
- Mouse users: Instead of performing the first two items, just click on the respective field with the mouse pointer.

5.9 Example 1: Lowpass Filter

For this filter, we desire a loss curve in the passband with equal ripple. The stopband will be optimized to a tolerance scheme with stepped profile.

Specifications:

Passband edge frequency: 1 kHz

Maximum allowed ripple in dB: 1 dB

Stopband loss:

from 1.05 kHz: 45 dB

from 1.6 kHz: 75 dB

From the main menu proceed as follows:

- [↵] Change to the FILTERTYPE menu.
- [↵] Change to the TYPE submenu.
- [↵] Set the type of filter to lowpass.
- [A] Change to the APPROXIMATION submenu.
- [↵] Set the equal ripple approximation.

The following entries are selected in the screen:

```
Filtertype                          Approximation

Lowpass                             General Equal-ripple
Highpass                            General Maximally-flat
Bandpass
Bandstop                            Butterworth
Allpass                             Chebychev
                                    Inverse Chebychev
                                    Elliptic (Cauer)
                                    Bessel (Thomson)
                                    Modified Bessel


FILTERTYPE: Type  Approximation                                        Quit  ?
```

Now select:

- [Q] — Return to main menu (If any other requirements were entered, confirm the request with [Y]).
- [S][N] — Change to the SPECIFICATION menu and prepare for a new filter.
- [↵]1k[↓] — Passband edge frequency: 1 kHz
- 1[↵] — Passband bandedge loss: 1 dB

The following will be seen on the screen:

```
            SPECIFICATIONS to :  Equal ripple - lowpass filter
 (O)

 (A)  Passband edge frequency          :              1.000 000 kHz

 (E)  Passband bandedge loss           :              1.000 000 dB
 (F)  Passband bandedge return loss    :                  6.87 dB
 (R)  Passband reflection factor       :                 45.35 %

      Transm. zeros at zero      :   0 | Fixed transm. zero pairs:   0
 (H)  Transm. zeros at infinity  :   1 | Var.  transm. zero pairs:   0



      Filter degree                    :                        1

      3dB edge frequency               :              1.965 227 kHz

      Filter quality                   :                 not defined
SPECIFICATION: A  E  F  R  H                                          qualitY
               Stopbandopt. fiXtrans.zeros  cOmment  New  fiLe Printer  Quit  ?
```

Now enter the stopband specifications:

- [S] Enter the OPTIMIZATION menu
- [T] Enter the STOPBAND SPECIFICATION menu
- [↵][↵]1.05k[↵] 45[↵] Enter the stopband loss of 45 dB from 1.05k
- [↵][↵]1.6k[↵]75[↵] Enter the stopband loss of 75 dB from 1.6k

The following screen will appear:

```
                         Stopband specifications

 No.                Frequency range                        Loss

   1        1.050 000 kHz    to     1.600 000 kHz       45.00 dB
   2        1.600 000 kHz    to      infinity           75.00 dB

STOPBAND SPECIFICATION: Add  Edit  Delete  New                    Quit  ?
```

Now continue with the optimization of the transmission zero positions:

- [Q] Return to the OPTIMIZATION menu.

In the OPTIMIZATION menu, *QuickFil* proposes as the initial values for the optimization four variable pairs of transmission zeros in the finite stopband and one transmission zero at infinity. This will result in a filter of the 9th degree. We accept this proposal and continue.

- [O] Select the optimization function.

After six iterations (see the entry in the Iteration field), the variable pairs of transmission zeros are positioned in the finite stopband in such a manner that the resultant loss curve fills the tolerance scheme throughout the whole stopband.

The screen after the optimization:

```
                        stopband optimization

 (A) Variable transmission zero pairs          (proposal :  4 ) :    4

 (B) Transmission zeros at infinity        :                         1

     Filter degree                         :                         9

     Minimum of loss reserve               :                   1.10 dB
     Difference of the minima              :                   0.04 dB

     Iteration                             :                         6

 (G) Limit of the optimization             :                   0.10 dB

OPTIMIZATION: A  B  G                                  sTopband_specification
              Variable-transmissionzeros   Optimization   Iteration     Quit  ?
```

The screen shows us that there is a loss reserve of 1.1 dB. We will use this reserve to obtain a slightly higher return loss.

- [Q] — Return to the SPECIFICATION menu.
- [F]7.7[↵] — Increase the return loss by a little less than the value of the loss reserve.
- [S] — Enter the OPTIMIZATION menu.
- [O] — Optimize of the position of the variable pairs of transmission zeros.

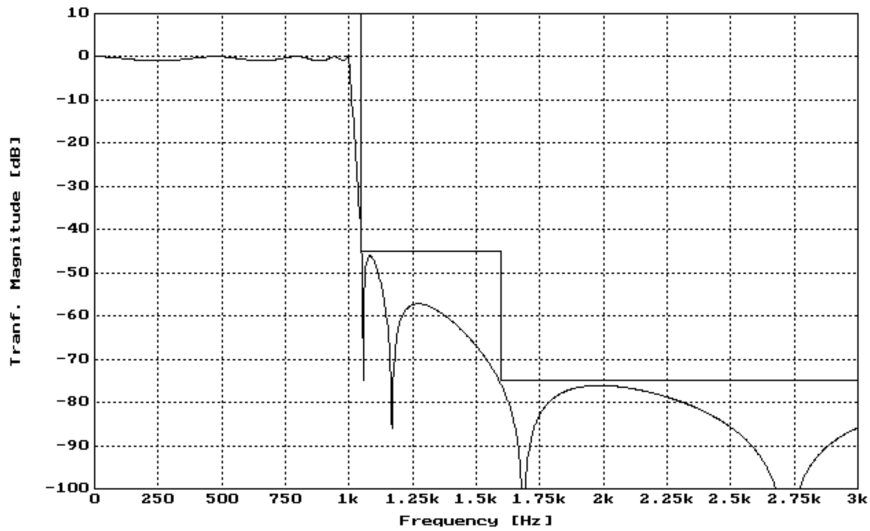
Note: Since the results of the last optimization are used as initial values, it is not necessary to repeat all of the iterations.

We can see on the screen that there is only a minimal loss reserve of 0.07 dB, which is sufficient for our purposes.

Let's return to the main menu.

- [Q][Q] — Return to the main menu.

This completes the specification of the filter. Using these values we would have developed a filter with the following loss curve:



5.10 Example 2: Bandpass Filter

For this filter, we desire the loss curve in the passband to be maximally flat. The stopband will have a fixed blocking point at a certain frequency. Otherwise, the loss in the stopband will rise monotonically.

Specifications:

Passband: 100 kHz to 150 kHz

Passband loss: 1 dB

Stopband:

up to 70 kHz: 35 dB

from 190 kHz: 45 dB

fixed blocking frequency at 210 kHz

First enter the specifications:

Starting point: Main menu

• [↵]	Enter the FILTERTYPE menu.
• [↵]	Enter the TYPE submenu.
• [B]	Select Bandpass.
• [A]	Enter the APPROXIMATION menu.
• [M]	Select Maximally-flat.
• [Q][Y][S][N]	Enter the SPECIFICATION menu, delete all entries (if any).
• [A]100k[↓]	Lower passband edge frequency = 100 kHz
• 150k[↓]	Upper passband edge frequency = 150 kHz
• 1[↵]	Passband bandedge loss: 1 dB
• [S]	Enter the OPTIMIZATION menu.
• [L]	Enter the STOPBAND SPECIFICATION menu for entering the lower tolerance scheme.
• [↵][↵]70k[↵]35[↵]	up to 70 kHz: minimum loss 35 dB
• [Q][U]	Enter the STOPBAND SPECIFICATION menu for entering the upper tolerance scheme.
• [↵][↵]190k[↵]45[↵]	from 190 kHz: minimum loss 45 dB
• [Q][Q]	Return to the SPECIFICATION menu.

The following screen will be displayed:

```
          SPECIFICATIONS to :  Maximally flat - bandpass filter
 (O)

     Kind of bandpass                :               conventional

 (A)  Lower passband edge frequency   :            100.000 000 kHz
 (B)  Upper passband edge frequency   :            150.000 000 kHz

 (E)  Passband bandedge loss          :              1.000 000 dB
 (F)  Passband bandedge return loss   :                   6.87 dB
 (R)  Passband reflection factor      :                  45.35 %

 (G)  Transm. zeros at zero     :   1 | Fixed transm. zero pairs:   0
 (H)  Transm. zeros at infinity :   1 | Var. transm. zero pairs :   4

      Filter degree                   :                     10
      Lower 3dB edge frequency        :             97.243 638 kHz
      Upper 3dB edge frequency        :            153.766 389 kHz
      Filter quality                  :                not defined
SPECIFICATION: A  B  E  F  R  G  H                     qualitY    Parametric
               Stopbandopt. fiXtrans.zeros  cOmment  New  fiLe Printer  Quit  ?
```

We will now set the desired losses in the stopband:

Enter the fixed blocking frequency at 210 kHz.

- [X] Jump into the TRANSMISSIONZERO INPUT menu.
- [↵]210k[↵] Fixed transmission zero at 210 kHz

Now, enter the setting that transmission zeros are only desired at extreme frequencies.

- [Q][S] Jump into the OPTIMIZATION menu.
- [↵]0[↓]0[↵] The number of the variable pairs of transmission zeros in the lower and upper stopband is set to 0 for each of these ranges.

It is quite clear that we can no longer manage with only one transmission zero at zero and infinity. Let's set their number to 4.

- [C]4[↓]4[↵] 4 transmission zeros each at frequency, zero and infinity
- [O] Selects the OPTIMIZATION menu in order to calculate the minimal loss reserve.

The negative value of the minimum of loss reserve shows us that we have to slightly increase the number of transmission zeros and thus, the degree of the filter.

- [C]5[↵] 5 transmission zeros at zero

You can see that the number of transmission zeros at infinity was reduced to three. This was due to the fact that conventional bandpass filters can only have an even degree, therefore, *QuickFil* corrects the number of transmission zeros accordingly. As we are not able to fulfill the requirements with only 3 transmission zeros at infinity (check by selecting Optimization), we also set the number of transmission zeros at infinity to 5.

- [D]5[↵] 5 transmission zeros at infinity

Let's see whether our requirements have been met.

- [O] Evaluation of the minimal loss reserve.

It fits because we receive a loss reserve of 3.44 dB. The filter has a degree of 12 (equal to the total number of the transmission zeros: one pair of transmission zeros at 210 kHz, five transmission zeros at zero and five transmission zeros at infinity).

We have thus specified our filter.

- [Q] Return to the SPECIFICATION menu.

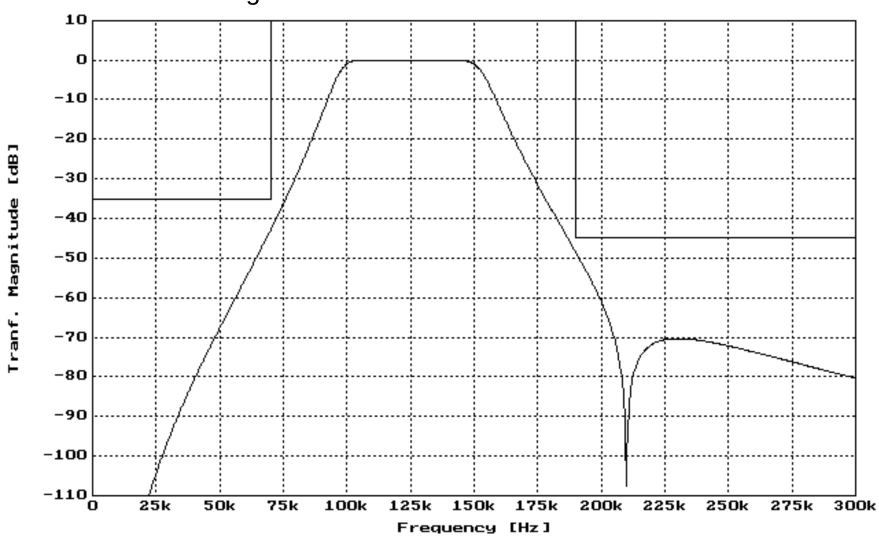
Let's calculate the filter quality:

- [Y][↵] Calculate the filter quality.

The following screen appears:

```
         SPECIFICATIONS to :  Maximally flat - bandpass filter
 (O)

     Kind of bandpass               :              conventional

 (A) Lower passband edge frequency  :            100.000 000 kHz
 (B) Upper passband edge frequency  :            150.000 000 kHz

 (E) Passband bandedge loss         :              1.000 000 dB
 (F) Passband bandedge return loss  :                   6.87 dB
 (R) Passband reflection factor     :                  45.35 %

 (G) Transm. zeros at zero     :  5 | Fixed transm. zero pairs:  1
 (H) Transm. zeros at infinity :  5 | Var. transm. zero pairs :  0

     Filter degree                  :                        12
     Lower 3dB edge frequency       :             97.327 399 kHz
     Upper 3dB edge frequency       :            153.024 850 kHz
     Filter quality                 :                     10.35
SPECIFICATION: A  B  E  F  R  G  H                        qualitY    Parametric
               Stopbandopt. fiXtrans.zeros  cOmment  New  fiLe Printer  Quit  ?
```

The filter we have designed will have the characteristic shown below:



5.11 Group delay

In this section of the program, you can design allpass filters having arbitrary group delay response.

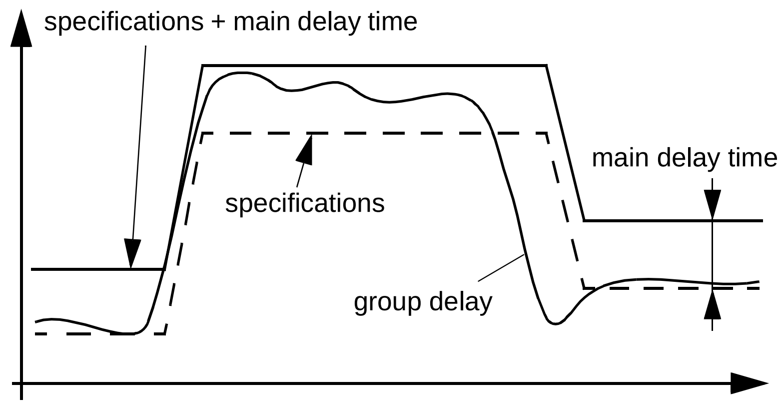
There are two possible applications:

- You can add the allpass filter to an existing filter. The groupdelay of the whole filter is optimized. You design a filter as described before and enter the menu ***Group_delay*** in the main menu.
- You can design an allpass filter standalone. You specify an allpass filter in the menu ***Filtertype*** and enter the menu ***Group_delay*** directly. You can enter the specifications of the allpass in that menu.

The approximation of the group delay is performed in following steps:

- You specify the interval of the frequency where you would like to optimize the group delay.
- You define the response of the group delay (normally flat).
- You specify the degree of the allpass filter (*QuickFil* will give you a suggestion).
- *QuickFil* will initialize the real and complex transmission zeros of the allpass filter.
- You can optimize the transmission zeros using least-p optimization.

Normally, for the response of the filter, only the relative group delay is important. It does not matter if you add a fixed group delay to the group delay response since the shape of the step response will be the same. Its only delayed which does not matter in most cases. Therefore, you can specify a ***Main delay time (Tg0)*** which is a fixed delay time added to the specified response of the group delay. You take the main delay time as fixed, or you can define it as a further parameter of the optimization and you will get better performance for the allpass filter.



Now, lets look at the main menu of the group delay optimizer:

```
                                Group delay correction

  (A)  Lower frequency bound (proposal:   0.000 000  Hz ):    0.000 000  Hz
  (B)  Upper frequency bound (proposal:   1.000 000 kHz ):    1.000 000 kHz

  (C)  Filter degree         (proposal:              6):               6
  (D)  Exponent p                                      :               2
  (E)  Number of points                                :              50

  (F)  Main delay time (Tg0)                           :     3.321 915 ms

       Original group delay difference                 :     1.338 808 ms
       Group delay difference with correction          :   341.815 457 µs
       Error norm                                      :    49.599 993 µs
                                                                optimized
       With optimization of main delay time            :             Yes
       With group delay specifications                 :              No

GROUP DELAY: A  B  C  D  E  F  cOment   Optimize    Iteration    Specifications
             Transm.-zeros     Tg0_on   Tg0_off       fiLe   Printer    Quit  ?
```

There are following menu items available:

Lower frequency bound

Here, you can specify the lower edge of the frequency interval for the optimization. There is a proposal available if you add the allpass filter to a standard filter. Its the lower edge frequency of the filter.

Upper frequency bound

Here, you can specify the upper edge of the frequency interval for the optimization. There is a proposal available if you add the allpass filter to a standard filter. It is the upper edge frequency of the filter.

Filter degree

Here, you can specify the degree of the allpass filter. The degree is a measure of the number of components needed for the allpass filter. After you have specified the Filter degree, the transmission zeros of the allpass filter are initialized. There is a proposal available which is an estimate of *QuickFil*, depending on the group delay specifications and the group delay performance of the filter, which should be corrected.

Exponent

The optimization of the group delay is using an error norm. The difference of the group delay specifications and the real group delay is the error group delay or the error function. Using a norm, you can determine for every error function a real number, the error norm of the error function. The goal of the optimization is to set the best parameters of the allpass filter to get the lowest error norm.

The error norm is the least-p norm which is well know in the literature. At this menu item, you can specify the exponent of the error norm. The limits for the exponent are 2 and 100. Using the exponent 2, you will get least-square optimization, using exponent 100, you will get about min-max optimization.

Number of points

The group delay is not considered a continuous function, it is investigated only at discrete frequency points. You have to do that discrete optimization because it is not possible to treat the continuous function in the algorithm. Therefore, you define points at the frequency axis which are equally spaced. At each of these frequency points, the group delay is calculated and an approximation of the error norm is built using all these data. At this menu item, you can specify the number of points used at the frequency axis. The lower limit is 10, the upper limit is 250. For increasing calculation speed, decrease the number of points. For increasing the accuracy and avoiding that some peaks of the group delay could happen and are not considered by the optimization, take a big number of points.

Main delay time (Tg0)

Here, you can specify the main delay time. However, it only makes sense if you have switched off the ***optimization of the main delay time.*** Here, you can shift the group delay response by a fixed delay time. If the ***optimization***

of the main delay time is switched on, the main delay time is a parameter of the optimization and is initialized if you specify the degree of the allpass.

cOmment

QuickFil automatically labels all outputs (roots, circuits, diagrams) with comments which you can enter via this option, e.g. If you specify a project number, you will always know which printout belongs to which project.

Optimization

If the all specifications of the allpass filters are defined, the optimization is started. There are some results available for the user which are updated in each cycle of the optimization:

- group delay difference with correction
- error norm

The optimization is necessary to get good results for the group delay. If you forget the optimization and leave the menu, there will be a message:
Group delay is not optimized! Leave nevertheless? Yes No

Iteration

An iteration is a single step of the optimization. Here, you can perform the optimization step by step and watch what happens with the error norm.

Specifications

Here, you can define the specifications of the group delay of the overall filter. You will enter the menu GROUP DELAY SPECIFICATIONS:. If you do not specify any group delay, the program will assume that the delay response is zero independent of the frequency.

Transm.-zeros

Here, you can investigate and change the transmission zeros of the allpass filter. You will enter the menu GR.-TRANSMISSION ZEROS:.

With optimization of main delay time

Here, you can specify if you want to add the main delay time as a further parameter for the optimization (yes/no). If the overall delay time does not matter, you can include the main delay time to the optimization parameters to get better results. If you want to get an exactly specified group delay response, please switch off the optimization of the main delay time.

fiLe

Saves the entered filter specifications in a file. The default extension is .SPZ. For more information on this data format, see the section on “Filter Specification Protocol”.

For more details, please refer to the section on “Output”.

Printer

Prints the filter specifications.

On the screen you can see following information which is helpful to get an overview of the group delay optimization:

Original group delay difference

Here, the maximum difference of the target group delay to the actual group delay in the defined frequency span is shown, considering that there is no allpass filter.

Group delay difference with correction

Here, the maximum difference of the target group delay to the actual group delay (including the allpass filter) in the defined frequency span is show.

Error norm

Here, the error norm of the difference of the target group delay and the actual group delay (including the allpass filter) is shown.

With group delay specifications

Here, you can see if you have specified any target group delay response in the menu GROUP DELAY SPECIFIACTION:.

5.12 Group delay specification

In the menu GROUP DELAY SPECIFICATION: you can define the response of the group delay.

```
                    Specifications of group delay

   No.           Frequency                Group delay

     1             0.000 000  Hz             0.000 000  s
     2           200.000 000 kHz             2.700 000 ns
     3           250.000 000 kHz             5.000 000 ns
     4           300.000 000 kHz             7.700 000 ns
     5           400.000 000 kHz            13.800 000 ns
     6           600.000 000 kHz            27.600 000 ns
     7           800.000 000 kHz            41.000 000 ns
     8             1.000 000 MHz            53.000 000 ns
     9             1.200 000 MHz            63.200 000 ns
    10             1.400 000 MHz            71.900 000 ns
    11             1.600 000 MHz            79.300 000 ns
    12             1.800 000 MHz            85.300 000 ns
    13             2.000 000 MHz            90.000 000 ns
    14             2.200 000 MHz            92.800 000 ns
                                                                     ↓
     Move with [Cursor keys]
GROUP DELAY SPECIFICATION: Add  Edit  Delete  New                  Quit  ?
```

You can specify discrete points of the group delay function by defining the frequency and the group delay at that frequency.

- The response of the specified group delay results by connecting the defined points in the frequency - group delay response graph (linear interpolation).
- For frequency below the point which frequency is the lowest one, the group delay is assumed to be constant and has the value of that point (constant extrapolation).
- Similar for frequencies above the point which frequency is the highest one, the group delay is assumed to be constant and has the value of that point (constant extrapolation)

Using these assumptions, the response of the specified group delay is defined for all frequencies.

The target response of the optimization of the group delay is defined by:

target group delay = specified group delay + main delay time

Note: If you do not specify any group delay specifications, the specifications are assumed to be zero group delay, independent of the frequency. In this case, you will get a flat group delay response.

In the menu GROUP DELAY SPECIFICATION: you can enter and change the group delay specifications.

Add

You can add one point of the group delay specification by defining the frequency and the group delay. This new point is added in the list of group delay points, which is sorted by the frequency.

Edit

Here, you can edit the list of group delay specification by following steps:

- Select the menu item Edit and the curser jumps to the first editing field.
- Using the cursor key, [↓] and [↑], select the input field.
- edit the input field.
- Finish the editing with [↵] key.

While editing the group delay specification points, the frequency is limited by the frequency of the neighboring points.

Delete

Here, you can delete single lines in the list, that is to say, you delete one point of the group delay specification. You have to specify the number of that point which you would like to delete.

New

The whole list of group delay specification is deleted.

5.13 Group delay transmission zeros

The transmission zeros of the allpass filter are the parameters of the optimization of the group delay response. There are real transmission and conjugated complex transmission zeros available.

In this menu, you can change the transmission zeros and investigate and initialize the optimization for better results.

Note: You will need experience to get the best start positions for the optimization that are proposed by *QuickFil*. If you have bad start values for the transmission zeros, the optimization will not converge or converge to a local minima which can prove to be very bad.

In the menu GR.-TRANSMISSION ZEROS: you can investigate and edit the transmission zeros of the allpass filter.

```
                Transmission zeros of group delay equalization

    No.     normalized transmission zeros (f0 =    4.800 000 MHz )

     1          0.204 740   +/-j     0.100 466
     2          0.194 233   +/-j     0.265 619
     3          0.185 066   +/-j     0.429 392
     4          0.176 782   +/-j     0.587 540
     5          0.169 757   +/-j     0.746 365
     6          0.153 750   +/-j     0.909 271

GR.-TRANSMISSION ZEROS: Add  Edit  Delete  New                        Quit  ?
```

Following menu items are available:

Add

Here you can add further transmission zeros to the list of transmission zeros in two steps:

- Enter the real part of the transmission zero. The real part has to be positive, negative values and zero are not allowed.
- Enter the imaginary part of the transmission zero. The imaginary part is positive or zero. If it is zero, there is only one real transmission zero otherwise, there are two conjugated complex transmission zeros.

Edit

Here, you can edit the transmission zeros by following steps:

- Select the menu item Edit and the curser jumps to the first editing field.
- Using the cursor key, [↓] and [↑], select the input field.
- edit the input field
- Finish the editing with [↵] key.

New

All transmission zeros are deleted. After leaving the menu, the filter degree is zero.

5.14 Example for Group delay correction

Now, let’s look at an example for group delay correction. In communications engineering, digital signal transfer is often used. The importance of digital modulations will increase more and more. Often, the modulation QPSK is used for the transfer of data via satellites.

Now, let’s specify a IF (intermediate frequency) filter for 70 MHz for a DVB-S signal:

- data rate: 42 Mbit / sec.
- roll off factor: 0.4
- characteristics: halve Nyquist
- centre frequency: 70 MHz
- stopband loss: 40 dB

For this task, the "maximally flat" approximation is best suited but you need group delay correction to reduce the inter symbol interference (ISI).

Let’s start the example beginning at the main menu:

- [O] Jump in the menu OPTIONS: For getting a defined starting position for our example we will perform a program reset.
- [R][[↵][↵] select the menu item: ***program_Reset***
- [Q] quit the OPTIONS menu
- [↵] jump to the menu FILTERTYPE
- [↵] jump to the menu TYPE
- [B] choose Bandpass
- [A] jump to the menu APPROXIMATION
- [M][Q] choose "maximally flat" and quit

Now let’s define the specifications of the filter:

- [S][↵] jump to the menu SPECIFICATION
- [P][↵] choose parametric bandpass filter
- [M][25M][↓] parameter = 25 MHz
- [59.5M][↓] lower passband edge frequency = 59.5MHz
- [80.5M][↓] upper passband edge frequency = 80.5 MHz
- [3.01][↵] passband bandedge loss = 3.01 dB
- [G][↵][3][↓] transmission zeros at zero = 3
- [5][↵] transmission zeros at infinity = 5

Further, we will specify two fixed transmission zeros:

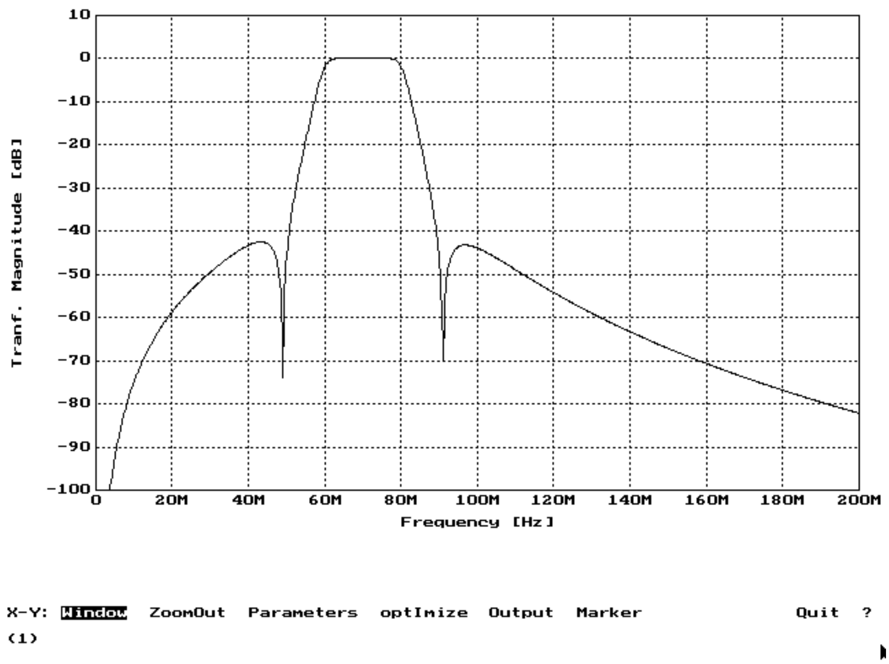
- [X] jump to the menu TRANSMISSION ZERO INPUT:
- [↵][49M][↵] transmission zero at 49 MHz
- [↵][91M][↵] transmission zero at 91 MHz
- [Q] return to main menu

```
          SPECIFICATIONS to :  Maximally flat - bandpass filter
 (O)

      Kind of bandpass                 :                 parametric
 (M)  Parameter                        :              25.000 000 MHz

 (A)  Lower passband edge frequency    :              59.500 000 MHz
 (B)  Upper passband edge frequency    :              80.500 000 MHz

 (E)  Passband bandedge loss           :                3.010 000 dB
 (F)  Passband bandedge return loss    :                     3.01 dB
 (R)  Passband reflection factor       :                    70.71 %

 (G)  Transm. zeros at zero      :   3 | Fixed transm. zero pairs:   2
 (H)  Transm. zeros at infinity  :   5 | Var. transm. zero pairs :   0

      Filter degree                    :                      12
      Lower 3dB edge frequency         :              59.499 871 MHz
      Upper 3dB edge frequency         :              80.500 126 MHz
      Filter quality                   :                 not defined

SPECIFICATION: M  A  B  E  F  R  G  H   qualitY   Conventional    Defaultparam.
               Stopbandopt. fiXtrans.zeros  cOmment  New  fiLe Printer  Quit  ?
```

- [Q] jump back to the main menu, the zeros of the transfer function are calculated

Using these specifications, we will get following loss function:

- [A] jump to the menu polynomial_Analysis
- [O][1000] number of points = 1000
- [G] show the loss performance graphically

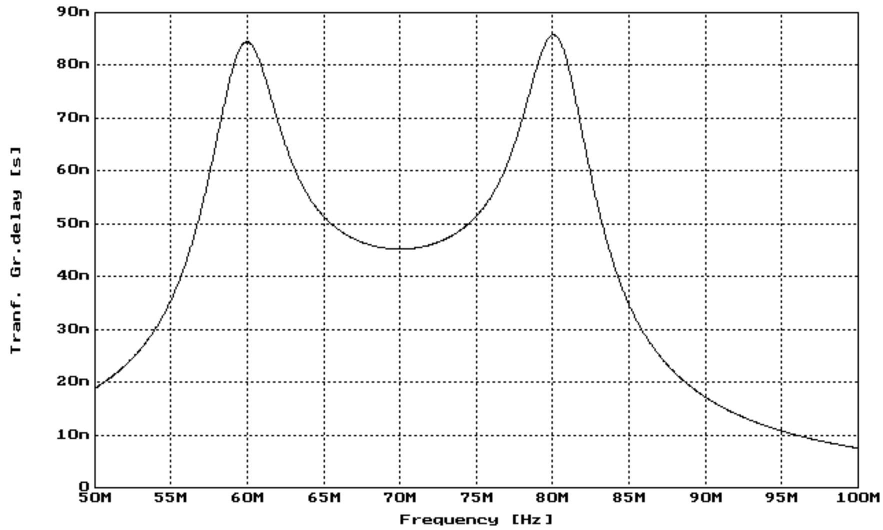


The loss response is good for our application. For a digital transmission, you have to demand that the impulses which represent the data, don't disturb each other (ISI inter symbol interference). Therefore, the impulse of the transmitter has to use a specified form.

Further, the receiving filter has to have a constant group delay in the range of the two 3 dB edge frequencies.

First, we will investigate the group delay of the bandpass filter.

- [Q] jump back to the menu POLY. ANALYSIS
- [X][X][↵][50M][[↵] X-from 50 MHz (lower frequency limit of the graph)
- [X][X][X][↵][100M][[↵] X-to 100 MHz (upper frequency limit of the graph)
- [R][D][↵][Q] Representation: ***group_Delay***
- [G] Show the graphic
- [I] Optimize



The group delay response differs from a flat response relatively heavy. Using the group delay correction of *QuickFil* you can improve the performance.

Now, we change the program section GROUP DELAY:

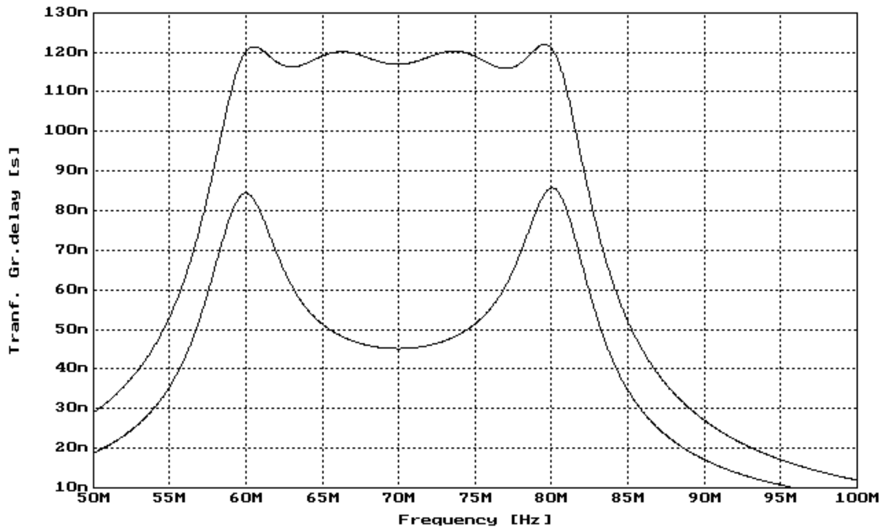
- [Q][H] return to the menu POLY. ANALYSIS and store the last diagram using the hold function. (HOLD.GDF)
- [Q][G] Jump to the menu GROUP DELAY.
- [A][↵] lower frequency bound: accept the proposal: 59.5 MHz
- [B][↵] upper frequency bound: accept the proposal: 80.5 MHz
- [C][4] Filter degree = 4
- [O][O][↵] Chose the menu item ***Optimize***. The group delay correction allpass filter is calculated.

```
                          Group delay correction

 (A)  Lower frequency bound (proposal:  59.500 000 MHz ):   59.500 000 MHz
 (B)  Upper frequency bound (proposal:  80.500 000 MHz ):   80.500 000 MHz

 (C)  Filter degree         (proposal:                4):                4
 (D)  Exponent p                                       :                2
 (E)  Number of points                                 :               50

 (F)  Main delay time (Tg0)                            :   118.554 513 ns

      Original group delay difference                  :    40.557 337 ns
      Group delay difference with correction           :     6.028 755 ns
      Error norm                                       :     1.613 934 ns
                                                                optimized
      With optimization of main delay time             :              Yes
      With group delay specifications                  :               No

GROUP DELAY: A  B  C  D  E  F  cOment   Optimize    Iteration    Specifications
             Transm.-zeros      Tg0_on   Tg0_off       fiLe   Printer    Quit  ?
```

Now, *QuickFil* has calculated the allpass filters for correcting the group delay distortions.

Now, we will compare the group delay response of the original filter with the group delay corrected filter.

- [Q][A][G] Jump to the menu POLY. ANALYSIS.
- [O] Optimize the diagram.



- [Q][Q] Return to main menu

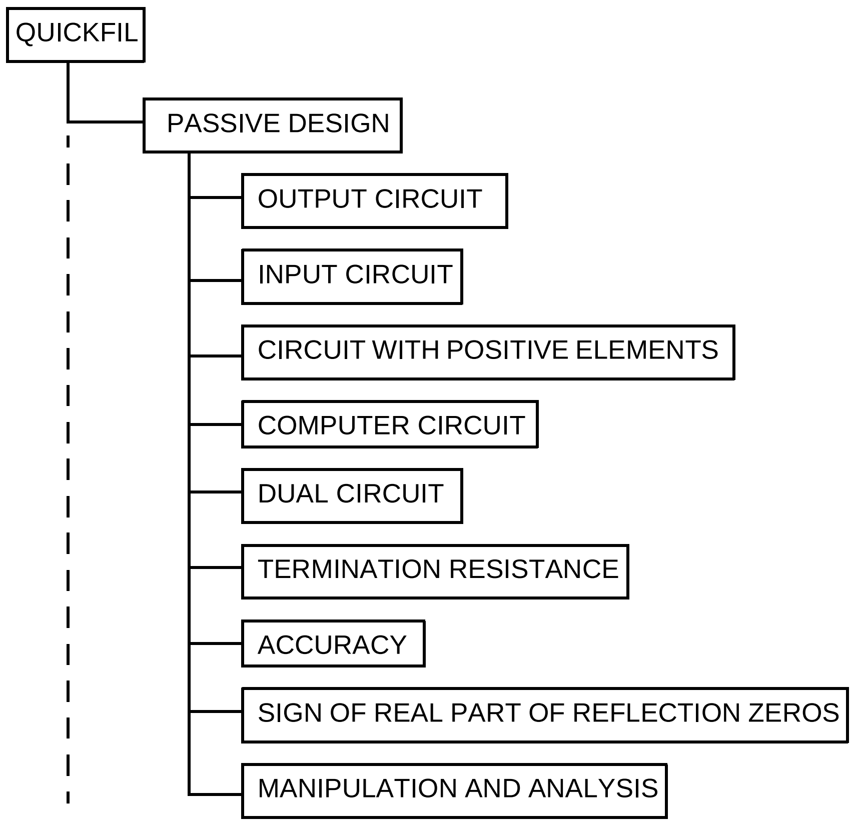
Now, we have finished this example.

5.15 Circuit Design

The polynomial roots are automatically calculated when the SPECIFICATION menu is exited. By using these roots, *QuickFil* can calculate a complete proposal for the LC (reactance) filter.

5.16 PASSIVE DESIGN menu

This menu is used to access various synthesis and analysis features.



Following menu items are available:

(O) Output circuit

In this menu, you can send the actual circuit topology (including component values) to the screen, printer, or file. File formats include ASCII, SPICE, (for programs such as *ISSPICE*) or Touchstone.

(I) Input circuit

Here, you can construct your own filter piece by piece.

(S) Circuits with positive elements

Since the synthesis of ladder circuits does not guarantee that you will get a circuit with realizable elements (positive element value for each component), a special algorithm is implemented for searching ladder circuits which have only positive elements. All possible circuits are investigated. Only the circuits having positive elements are extracted.

(C) Computer circuit

Returns you to *QuickFil*'s initial circuit suggestion.

(D) Dual circuit

Switches to the corresponding dual circuit. The duality constant used for this transformation is the reference resistance entered via option (T) Terminating resistance. For more details on dual circuits, see the “Dual Circuits” section.

(T) Terminating resistance

Switches to the TERMINATION menu where you can specify the reference resistance and the type of filter termination.

(A) Accuracy

Switches to the ACCURACY menu where you can adjust the accuracy of calculations by specifying the number of significant decimal places.

(M) Manipulation and analysis

Switches to the MANIPULATION AND ANALYSIS menu where you can modify the filter and analyze the circuit characteristics graphically.

Note: If the message:

```
The filter cannot be realized without mutual inductances!
It is at least one transmission zero at zero or at infinity necessary.
Press any key to continue!
```

appears when you enter the PASSIVE DESIGN menu from the main menu, then you cannot realize the filter with classical passive ladder structure. But you can realize an active or digital filter for that transfer function. These approximations are included only for analyzing features and for comparison with other approximations.

For passive ladder structure, you need at least one transmission zero at zero or at infinity. To get realizable passive filters, please change:

In standard approximations:

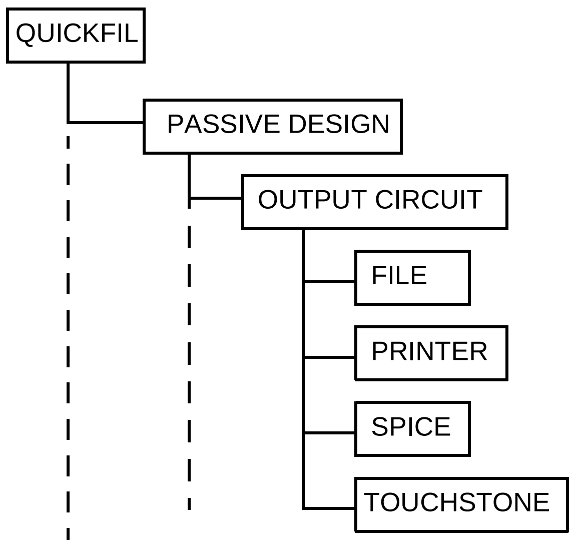
- Select a case that can be realized passively (case b or c); or

In equal ripple or maximally flat approximations

- Select at least one transmission zero at zero or at infinity.

5.17 OUTPUT CIRCUIT Menu

This menu is used to output a circuit with component values. The circuit will be shown on the screen as soon as you switch to this menu. Other destination devices can be selected via the menu line.



File

Outputs the circuit to an ASCII file. Default extension: .SCH

Printer

Outputs the circuit to the printer.

Spice

Outputs the circuit, shown on the screen, to a file in a standard SPICE format that is compatible with all ISSPICE versions. Default extension: .CIR

Touchstone

Outputs circuit data to a file in a form which can be read by Touchstone. Default extension: .CKT

Note: The following messages may appear:

*** The circuit calculation is inaccurate! ***

This means inaccuracies were found when the circuit was constructed. The component values are calculated from the input and output sides of the circuit. The quotient of the component values derived from both calculations is obtained. An average of the quotients is used to determine the output resistance. For calculations to be valid, they must fall within a specified range around the calculated average. If the quotients vary from the average too much, this error message will appear.

Remedy: Increase the accuracy in the ACCURACY menu.

*** Attention: Negative component values! ***

Negative components are (physically) impossible. This is not a calculation error - filter table books would have equivalent entries except that they do not show them very often! Remedy: Manipulate the circuit using the MANIPULATION AND ANALYSIS menu. *QuickFil*'s analysis section can handle negative component values.

5.18 SPICE Menu

Here you can preview and manipulate the *ISSPICE* netlist files generated by *QuickFil*.

Save

Saves the output in one of the two following formats:

Standalone

The output is written to a single circuit file, including terminating resistors, input voltage source and analysis statements (.AC and .PRINT). It can be used directly for in an *ISSPICE* simulation. Default extension: .CIR

Subcircuit

The output is added to a library file in an *ISSPICE* subcircuit format and can be identified by specifying a subcircuit name. Default extension of the library file: .LIB

Tol R-tol L-tol C-tol

The component tolerances [in %] can be set for individual components with the Tol function or for component groups with the R-tol, L-tol and C-tol functions. These tolerances are for use with the Monte Carlo analysis features of Intusoft's *ICAPS* package.

qualitY L-qual. C-qual.

The quality can be entered for individual components (qualitY) or for all inductors or capacitors simultaneously (L-qual. or C-qual.). The corresponding series or parallel resistors are added automatically.

fBegin fEnd Points Lin Dec

These menu options define the *ISSPICE* .AC frequency analysis.

fDefault

Set the .AC frequency sweep parameters to *QuickFil* defaults.

Digits

Select the number of digits used for numerical representation.

5.19 TOUCHSTONE Menu

In this menu, you can preview and manipulate the Touchstone output generated by *QuickFil.* The data window allows you to scroll through the entire file.

Save

Saves the output to a file in Touchstone format. Default extension: .CKT

Name

Allows you to enter an individual name for the subcircuit.

fBegin fEnd Step

These menu options define the AC-sweep.

fDefault

Set the AC-sweep parameters back to *QuickFil's* defaults.

Digits

Select the number of digits used for numerical representation.

5.20 INPUT CIRCUIT Menu

This menu allows you to design your own circuits by putting the filter together piece by piece in single steps.

Bandpass filter circuits allow a wide variety of representations - lowpass, highpass and bandstop filters offer fewer representations.

At each stage, *QuickFil* checks which components can be used without losing the desired filter characteristics and will only offer valid components.

Definition of the screen entries:

Transmission zeros act./req.

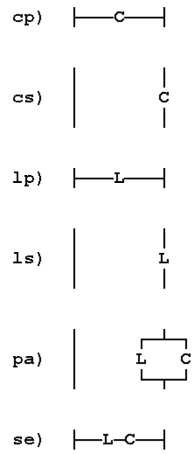
Indicates the number of transmission zeros at:

- zero
- infinity
- pairs at finite frequencies

The first number indicates the actual number of transmission zeros that will be achieved with the circuit, which you have already entered. It is assumed that the circuit terminates with a resistor.

The second number indicates the required number of the transmission zeros as specified in the specification menu. The entered filter must have the same number of transmission zeros as the transfer function defines.

You can combine the following two ports:



If you select ***pa*** or ***se***, the system will ask which transmission zero to use. Once a transmission zero has been used in the circuit, it cannot be selected a second time.

The cascading of the two-ports is limited by some rules. All these rules are implemented and you can only possibly choose the two-ports. If you have a bandpass filter circuit and if it is possible and would like to enter it, the circuit is not the simplest one because there are some Norton’s transformations included. In that case, please leave the components which are not possible to enter and perform later on Norton’s transformation to get the desired ladder structure.

Note: In this menu, you cannot specify an allpass filter. The structure of allpass filters are fixed.

5.21 Circuits with positive elements

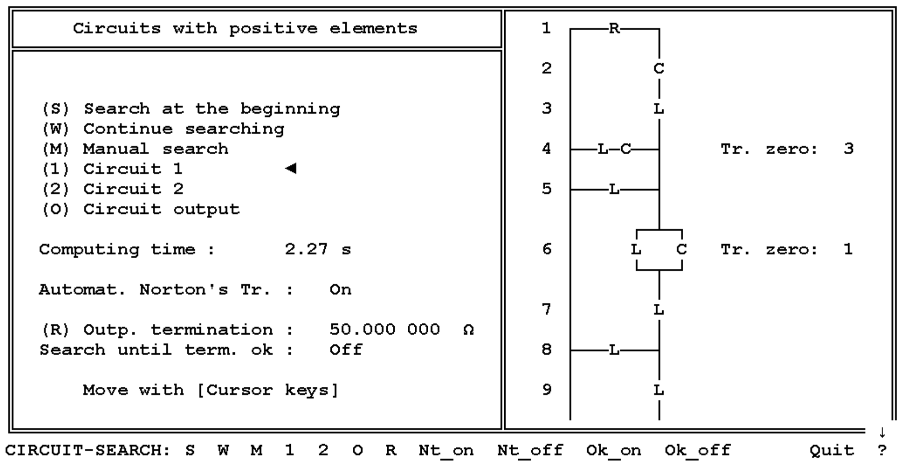
Since there is no guarantee that you will get a ladder structure which only has positive element values, you will need help for finding realizable circuits. For that purpose, *QuickFil* has included a powerful tool to find circuits which only have positive elements.

The method of synthesis of reactance filters is based on the realization of the input reactance if the output is open or shorted. This reactance is synthesized using the constraint that the transmission zeros of the transfer function have to be realized too. The process is based on extraction of elements from the reactance, step by step, while the degree of the reactance function is decreasing until it is zero. In that way, you will get every element.

There is some freedom by this realization process since there are many possible extraction subcircuits. Now *QuickFil* will try all possible subcircuit combinations until there is a circuit which only has positive elements.

The search algorithm can be treated as a tree with many branches, each with its own branches, and those with their own branches, etc. The branches leading to the end represents a ladder circuit and some of these circuits are realizable. If there is one negative element at a branch, the remaining branching from that branch will not be of use because no realizable circuits are on that branch. You might say the branch is broken. Now the algorithm is searching the entire tree until it finds a path to the end of the tree which represents a realizable circuit.

The menu screen looks like this:



The menu items are as follows:

Search at the beginning

The circuit search algorithm will be started at a defined starting circuit, which will be the same every time.

Continue searching

If you have found a realizable circuit and you would like to find another circuit possibility, you can continue the search procedure. For many approximations, you will get thousands of realizable circuits. Some approximations do not have a realizable ladder circuit available.

Manual search

If you are not happy with the automatic search, you can perform a manual search using some more intelligent algorithm. You will enter the menu

MANUAL-SEARCH. In this menu *QuickFil* will show you all possible subcircuits, which you can choose at this part of the synthesis process. You can make a manual synthesis of the first part of the circuit and continue with automatic synthesis further on. Perhaps you can find a solution in a much shorter time, because the systematic search can take a long time.

Output circuit

Here you can investigate the circuit more in detail, including the element values. This menu is identical to the menu in the PASSIV-DESIGN menu.

Automatic Norton's transformation

For bandpass filters you can add Norton's transformations into the circuit to get a specified termination circuit. Only simple Norton's transformation are used (no series or parallel circuits). The termination resistance can be specified.

Output termination

The termination resistance of the output of the filter for automatic Norton's transformation can be specified here.

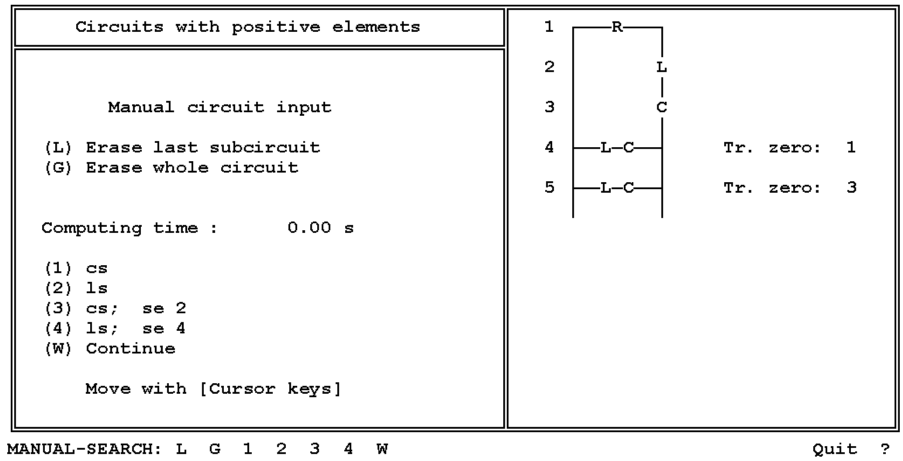
Search until termination OK

This features makes any sense only for bandpass filters and if the automatic Norton's transformation is switched on. The automatic search will be terminated only, if the termination resistance is OK.

Note: Allpass filters arn‘t considered in this menu, because every allpass filter has positive element values.

5.21.1 Circuits with positive elements - Manual Search

This is a submenu of the menu Circuits with positive elements.



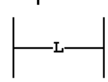
Here you can choose subcircuits from a list of possible subcircuits, delete subcircuits or the whole circuit. If there are more than four subcircuits possible you can choose the menu item ***Continue*** to get more possible subcircuits.

All subcircuits, which are shown, are realizable, that is to say, all elements of the subcircuit are positive. Since the whole circuits consists of such subcircuits, the whole circuit will have positive elements only.

There are following abbreviations used:

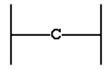
- ls; inductor series

- lp; inductor parallel

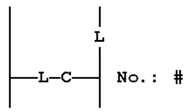


- cs; — capacitor in series

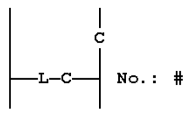
- cp; — capacitor parallel



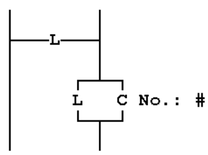
- ls; se #; — series inductor and parallel a serial resonant circuit using the transmission zero number #



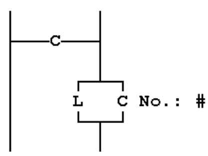
- cs; se #; — series capacitor and parallel a serial resonant circuit using the transmission zero number #



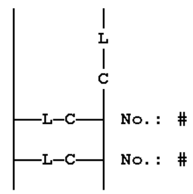
- lp; pa #; — parallel inductor and series a parallel resonant circuit using the transmission zero number #



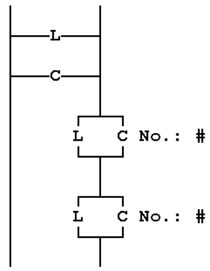
- cp; pa #; — parallel capacitor and series a parallel resonant circuit using the transmission zero number #



- ls; cs; se #; se #; series inductor, series capacitor parallel two serial resonant circuits using two transmission zeros specified by their numbers # #.



- lp; cp; pa #; pa #; parallel inductor, parallel inductor, series two parallel resonant circuits using tow transmission zeros specified by their numbers # #.



These abbreviations are about the same as used in the menu CIRCUIT-INPUT, but there are some more restrictions, because you can design finite transmission zeros only by a two-port combination.

5.22 Computer Circuit

If you choose the computer circuit, the program will build a circuit, which is realizable normally. This computer circuit is chosen, if you enter the ***PASSIV*** menu the first time or if you change the transfer function in such a way, that it cannot be realized with the last used circuit. That is to say, the number of transmission zeros at zero, infinity and at finite frequencies has changed.

5.23 Dual Circuit

Here you can choose the dual circuit of the actual circuit. The duality constant is fixed to the termination resistance specified in the termination menu.

5.24 TERMINATION Menu

In this menu you can determine whether the filter is doubly terminated or is provided with an extreme termination either on the input or output side (short-circuit or open, depending on the selected type of circuit (dual types of circuits!)). You can also determine the value of the reference resistance.

Only the real resistance of the source, or the subsequently added network, is taken into account.

Note: If the load impedances are complex, which happens from time to time, it is possible to partly take this into account. If the load impedances can be simulated by a model of an ohmic resistance and a parallel capacitor, it is possible to synthesize the desired filter with a filter structure that starts with a parallel capacitor. The capacity determined by the program is then distributed by value to the capacity of the model's load impedance and a residual capacity of the filter. Naturally, other models that are based on analog principles can also be realized.

5.25 ACCURACY Menu

You can use this menu to specify the accuracy of calculations by entering the number of significant decimal places.

General remarks on the accuracy of filter design:

By its nature, circuit design is highly sensitive to round-off errors. This means that almost all filter design programs fail with low pass filters from the 9th degree on - even if they use the math coprocessor's high-resolution data format. *QuickFil* uses its own computing algorithm, written in assembler, with a very high internal accuracy (up to 120 decimal places). This enables filters up to the 50th degree to be calculated. *QuickFil*'s accuracy is one of its main strengths.

Following menu items are available:

Accuracy

Indicates the number of significant decimal places being used. If the number of places you enter is greater than the previously specified value, the roots will be reiterated before the circuit is designed.

Accuracy of the refinement

This shows what accuracy was previously used.

QuickFil proposal

QuickFil makes its own proposal in light of the approximation and degree selected. The system will then use this accuracy unless you specify a greater accuracy.

Note: Internal calculations for *QuickFil* use 16-bit words. Therefore, the program rounds accuracy up to the next highest number of decimal places.

Refinement of the roots is only necessary for doubly-terminated filters. The Feldtkeller equation (see “Zero Output” section) must be fulfilled.

5.26 Sign of the real part of the reflection zeros

This menu is very special for passive filters. If you define a transfer function, the reflection factor isn’t defined uniquely in general. If the reflection zeros have a non zero real part you can choose the sign of the real part of the reflection zeros without any effect to the transfer function.

Note: The theoretical background can be found in the Feldtkeller equation. If you know the transfer function G(s) you know the polynomials e(s) and p(s) and you can calculate the polynomial f(s) * f(-s) using the Feldtkeller equation. You can determine the roots of the product polynomial f(s) * f(-s) uniquely, but you can’t allocate the roots to f(s) in an unique manner.

This special case can occur only for Bessel Filters, Modified Bessel filters and for parametric bandpass filters.

If you analyze the filter characteristics you will see only a difference in the phase of the reflection factor and in the input and output impedance. All other parameters will be the same. Further you will see different element values for the passive circuit. It is possible, that some sign combinations of the reflection zeros will produce realizable circuits, while others sing combinations will have no realizable circuit.

```
                Sign of the real part of the reflection zeros

   1.)  (R)  -       16.) (C)  +
   2.)  (C)  -       17.) (C)  +
   3.)  (C)  -       18.) (C)  +
   4.)  (C)  -       19.) (C)  +
   5.)  (C)  -       20.) (C)  +
   6.)  (C)  -       21.) (C)  +
   7.)  (C)  +       22.) (C)  +
   8.)  (C)  -       23.) (C)  +
   9.)  (C)  -       24.) (C)  +
   10.) (C)  +       25.) (C)  +
   11.) (C)  +
   12.) (C)  +
   13.) (C)  +
   14.) (C)  +
   15.) (C)  +

SIGN-POLYNOM-F: Change  Optimize                                  Quit  ?
```

Here you will find a list of real and complex reflection zeros. The first number is a counting number for identification of the reflection zero. The second item will show you, if it is a real or a complex reflection zero:

- (C) complex reflection zero
- (R) real reflection zero

The following sign is the sign of the real part of the reflection zero.

Change

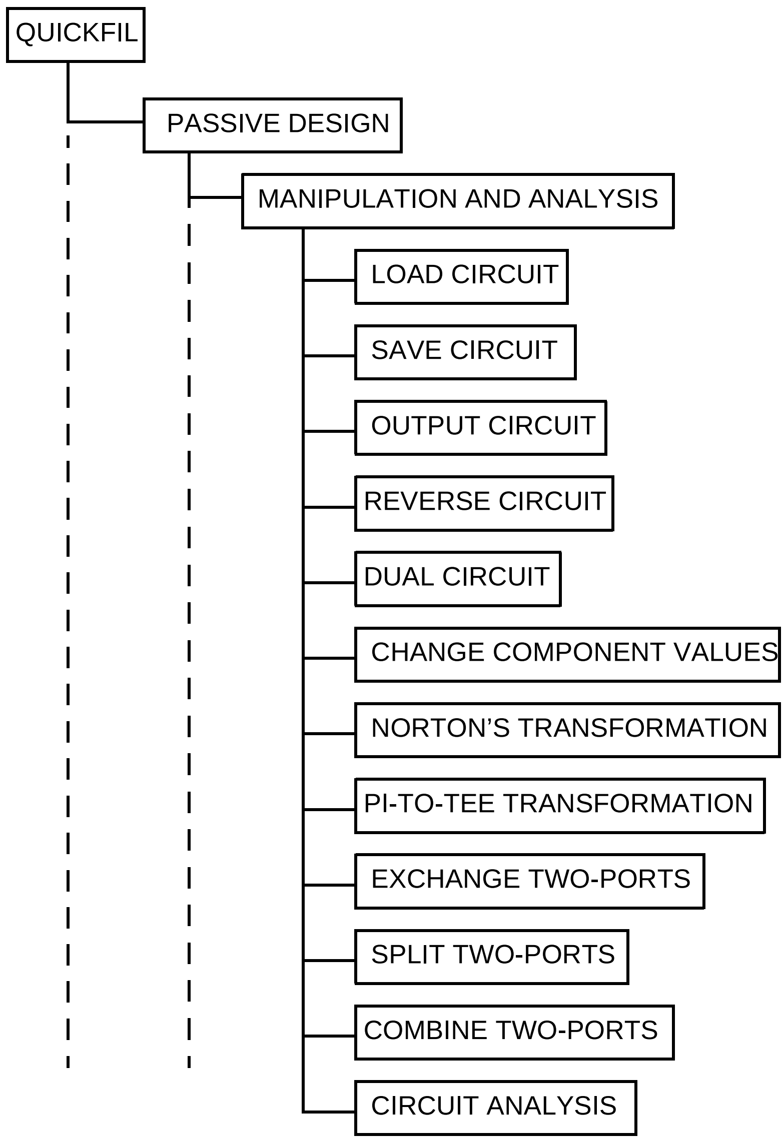
Here you can change the sign of the real part of the reflection zero. You need specify the number of the reflection zero and the sign will change.

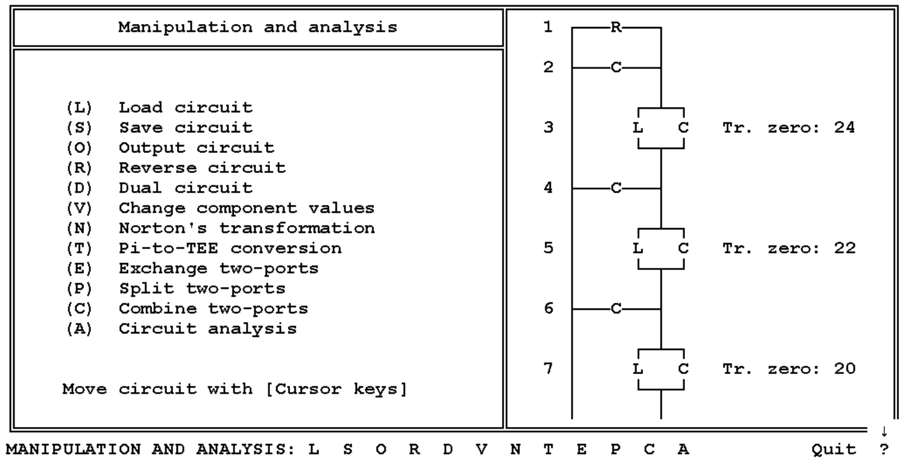
Optimize

A special optimization algorithm is used to find the optimum sign combination. This algorithm tries to make the phase of the reflection factor at the input and at the output at about equal.

5.27 MANIPULATION AND ANALYSIS Menu

You can use this menu to modify the calculated circuits. Beware: Any changes are lost upon exiting this program section, but the circuit is saved in the file "LAST.SH", if you leave the menu. So you can load the circuit later on. The following selections are available:





(L) Load circuit

Loads circuits previously saved via the (S) Save circuit function. If you would like to load the circuit, which you have used before you left the menu MANIPULATION AND ANALYSIS: last time, load the file "LAST.SH".

(S) Save circuit

Saves the current circuit in a binary format. Default extension: .SH

(O) Output circuit

Switches to the OUTPUT CIRCUIT menu to output circuits (with component values) to the:

- Screen
- Printer
- File in
- ASCII format (as displayed on screen)
- ISSPICE (SPICE analysis software)
- Touchstone format (analysis software)

(R) Reverse circuit

Circuit input and output are reversed (mirror-image).

(D) Dual circuit

Used for selecting the dual circuit. Before *QuickFil* proposes the respective dual circuit, the entry of the duality constant is requested. The reference resistance determined in the PASSIVE DESIGN menu is used as the initial proposal.

(V) Change component values

Allows the component values to be changed.

(N) Norton's transformation

Switches to the NORTON'S TRANSFORMATION menu, where you can optimize component values for bandpass filters without changing the filter's transmission characteristics.

(T) PI-to-TEE conversion

Switches to the PI-TO-TEE CONVERSION menu where PI arrangements can be transformed into TEE structures (or vice-versa), if such arrangements arise due to Norton's transformation.

(E) Exchange two-ports

Switches to the EXCHANGE TWO-PORT menu, where you can switch any two two-ports.

(P) Split two-ports

Switches to the SPLIT TWO-PORTS menu, where you can split any two-port into two two-ports (L -> parallel circuit, C -> series circuit), giving another degree of freedom for Norton's transformation.

(C) Combine two-ports

The system examines the circuit and combines similar two ports (switched in series or parallel).

(A) Circuit analysis

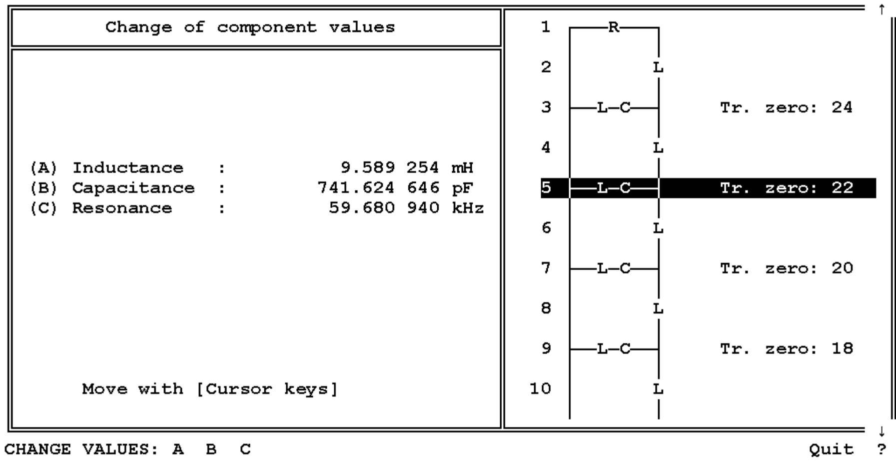
Switches to the CIRCUIT ANALYSIS menu, where you can analyze the characteristics of the filter.

5.28 DUAL CIRCUIT Menu

In this menu, you will be asked for the value of the duality constant to be used in the transformation to the dual version of the current circuit. The transformation is executed when you exit the menu using Quit.

5.29 CHANGE VALUES Menu

From this menu you can change component values.



To change a component's value;

- Use the arrow keys to move the cursor to the component you want to change ([↓] or [↑])
- Go to the appropriate field.
- Enter the required value and execute the change by pressing [↵].

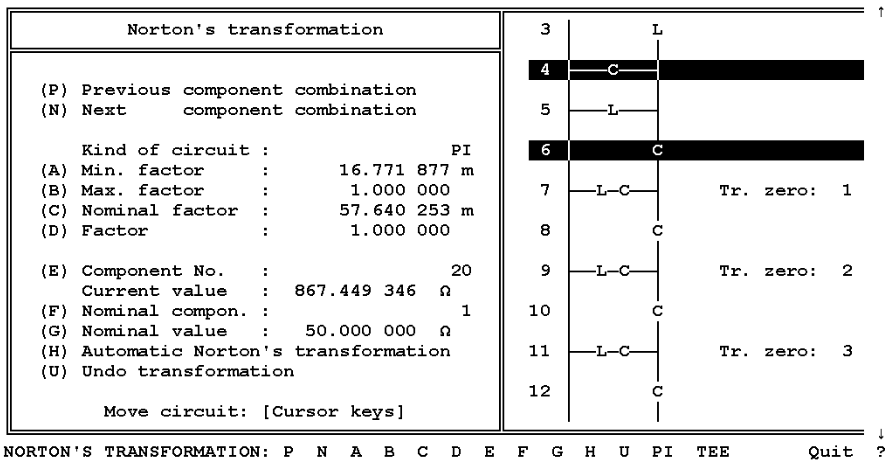
Note: *QuickFil* will take the new value as fixed, and will modify the filter's characteristics accordingly.

Normally you don‘t have the exact values for each component available. Normally the components are available in some standard values and you can combine elements in series and in parallel to get a better approximation for the desired component value. Depending on the values of the components the filter characteristics may be destroyed, more or less. In this menu you can change the component values and test, what happens to the filter characteristic.

Note: The component values of an allpass filter can't be changed in this version of the program *QuickFil*.

5.30 NORTON'S TRANSFORMATION Menu

You can use this menu to optimize component values without affecting the filter's transmission characteristics. Since Norton's transformation invariably creates circuit components with bandpass characteristics, it is only available with bandpass filters.



Norton's transformation:

- Multiplies (L's and R), or divides (C), for all subsequent component values, by the transformation factor (t), and
- Gives a new component arrangement within the chosen combination of components.

The following options are available:

(N) Next component combination

Norton's transformation can only be used with certain circuit combinations. Selecting this command switches you to the next combination which can be subjected to Norton's transformation.

(V) Previous component combination

Switches to the previous component combination on which Norton's transformation can be performed.

(A) Min. factor, (B) Max. factor

Positive component values can only be achieved if the transformation is kept within certain limits. These limits are calculated for the selected combination and entered into this field. If you select (A) or (B), that value will be used as the transformation factor (entered field (D) Factor), and the transformation will be performed using that factor.

(C) Nominal factor

QuickFil uses this field to enter the transformation factor required to give the value entered in (G) for the component in (E). If (G) is blank, then field (C) is missing.

Nominal factor (C) = Nominal value (G) / Current component value (E)

This function replaces the pocket calculator commonly used for this purpose.

If you select (C), the corresponding value will be used as the transformation factor (entered in (D) Factor), and the transformation will be carried out accordingly. Values outside the limits are also allowed. (see notes on (D)).

(D) Factor

The transformation will be carried out using the factor entered here.

Entries can be made in two ways:

- By using the value of (A), (B), or (C). This is accomplished by selecting one of these options. *QuickFil* will assume that value and carry out the transformation immediately.
- By entering a value directly into the input field. The transformation will run as soon as you confirm the value.

QuickFil also allows transformations with factors outside the limits in each case. This usually generates negative component values, but if negative values already exist, it may be possible to convert them to positive ones.

(E) Component No.

This is used to enter the number of the circuit component whose value is to be modified to the nominal value in (G). Each component number is visible in the circuit. If a number is allocated to a resonant circuit, then the number always refers to the inductance. The current value of the component selected at any time is shown under “Current value”. Only components, which are in the component combination selected for transformation, can be selected.

(F) Nominal component

This is used to enter a component whose value will be given to the component shown under (E) once the transformation is complete. If a nominal component is selected, its value is used as the nominal value (G). Only those components, which are placed before the component combination selected for transformation, can be used as nominal components.

(G) Nominal value

Specifies the value that the component shown under (E) is to have once the transformation is complete. This nominal value can be input in two ways:

- By selecting a nominal component via (F), whose value is then used as the nominal value (useful in circuits with equal L values)
- Entering the individual value directly.

(H) Automatic Norton's transformation

The program will perform Norton's transformations in automatic mode. *QuickFil* will search for possible element combinations for Norton's transformation and will perform Norton's transformation in such a manner, that the component specified in (E) will achieve the value specified in (G). Only simple Norton's transformations are used, that is to say, no resonant circuits are used for the elements, which are used for the transformation.

(U) Undo transformation

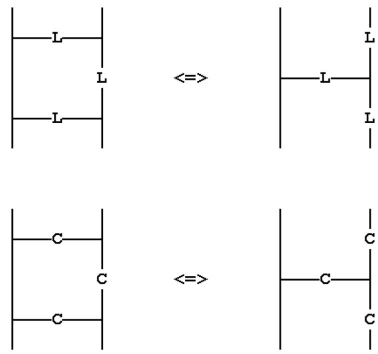
You can undo one Norton's transformation. *QuickFil* will store the circuit, before it will perform the Norton's transformation. If you will undo one Norton's transformation, the circuit is stored back and is equal as before the transformation.

PI and TEE

There are two possible subcircuits, which the Norton's transformation can build, the PI and the TEE circuit. These subcircuits are equivalent. Using this menu options you can choose, which subcircuit you desire.

5.31 PI-TO-TEE CONVERSION Menu

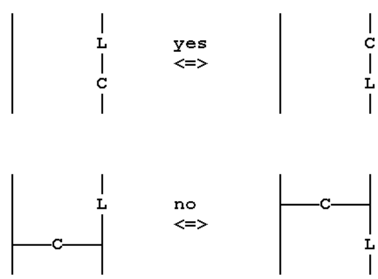
You can use this menu to transform component combinations in a circuit from a PI to a TEE configuration (or vice versa) provided that Norton's transformation produced these for L's or C's.



5.32 EXCHANGE TWO-PORTS Menu

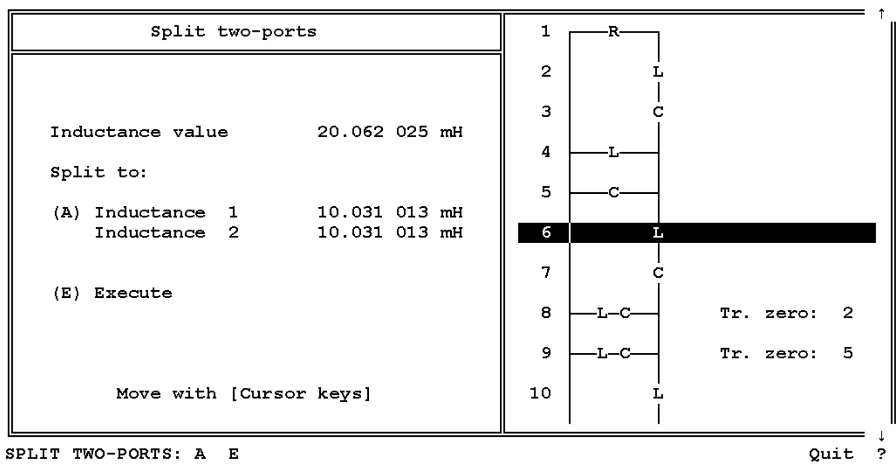
Use this menu to exchange adjacent two-ports (only allowed if this does not change the filter characteristics).

Examples:

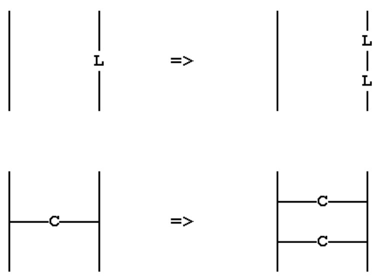


5.33 SPLIT TWO-PORTS Menu

In this menu, you can split two-ports into two two-ports (L - parallel circuit, C - series circuit). This gives an extra degree of freedom for Norton's transformation. The value of component 1 can be entered; *QuickFil* will then automatically calculate the value of component 2. Together, these two two-ports will then have the same value as the original.



Examples:



5.34 Example: Bandpass Filter

In this example, we will design a filter for a satellite receiver's second intermediate frequency.

The example will demonstrate how to:

- Make changes in a circuit proposed by *QuickFil* (for example, to achieve equal inductor values),
- Make use of the Norton's transformation option,
- Perform a PI-to-TEE transformation, and
- Design a combinational filter structure (for example, for developing a bandpass by placing a lowpass and a highpass in sequence).

Specifications:

Center frequency: 70 MHz

Bandwidth: 27 MHz

Upper stopband edge frequency: 97 MHz

Return loss: 20 dB

Stopband loss: 25 dB

Chosen approximation: Chebyshev

Additional requirement: For manufacturing reasons all inductors should have the same value.

Let's enter the filter specifications first.

Enter the following from the main menu:

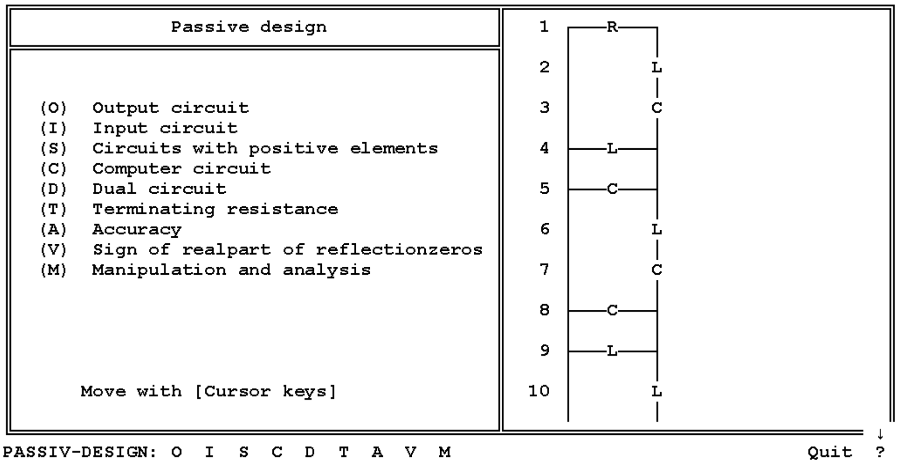
- [↵] Activate the FILTERTYPE menu
- [↵][B] Select Bandpass
- [A][C] Select the Chebyshev approximation
- [Q][Y][S][N] Activate the SPECIFICATION menu, delete all existing entries (if any)
- [A]56.5M[↓] Lower passband edge frequency: Value 56.5 MHz
- 83.5M[↓][↓] Upper passband edge frequency: 83.5 MHz
- 97M[↓][↓] Upper stopband edge frequency: 97 MHz
- 20[↓][↓] Passband bandedge return loss: 20 dB
- 25[↵] Stopband loss: 25 dB

This completes the specification with the following results on the screen:

```
              SPECIFICATIONS to :  Chebychev - bandpass filter
 (O)

 (A)  Lower passband edge frequency      :          56.500 000 MHz
 (B)  Upper passband edge frequency      :          83.500 000 MHz
 (C)  Lower stopband edge frequency      :          48.636 598 MHz
 (D)  Upper stopband edge frequency      :          97.000 000 MHz

 (E)  Passband bandedge loss             :            0.043 648 dB
 (F)  Passband bandedge return loss      :                20.00 dB
 (R)  Passband reflection factor         :                10.00 %
 (G)  Stopband loss                      :                25.59 dB

 (H)  Filter degree                      :                   10    ◄
 (I)  Case                       (b, c)  :                    b
 (J)  Variable value (A,B,C,D,E,F,G,H,R) :                    H

      Lower 3dB edge frequency           :          54.537 582 MHz
      Upper 3dB edge frequency           :          86.504 569 MHz
      Filter quality                     :                13.28

SPECIFICATION: A  B  C  D  E  F  R  G  H  I  J   New  cOmment   fiLe    Printer
               freqUencyrepres.  bandwithrepreS.  reL.bandwithrepres.    Quit  ?
```

Now let's move on to the "Passive design" program section:

- [Q][D] Activate the PASSIVE DESIGN menu The following screen appears:

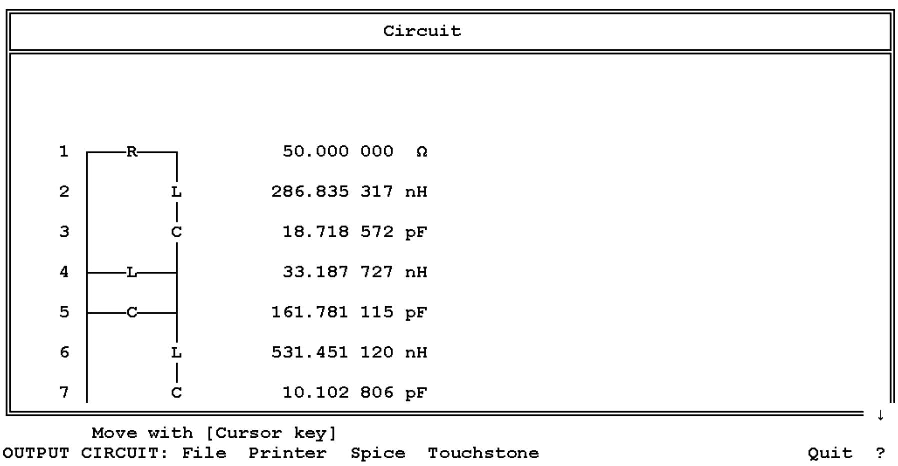


Note: If your screen shows another circuit please enter [C][Y] to reset *QuickFil*.

Take a look at the component values for the proposed circuit.

- [↵] Activate the OUTPUT CIRCUIT menu

After a short calculation phase the following screen appears:

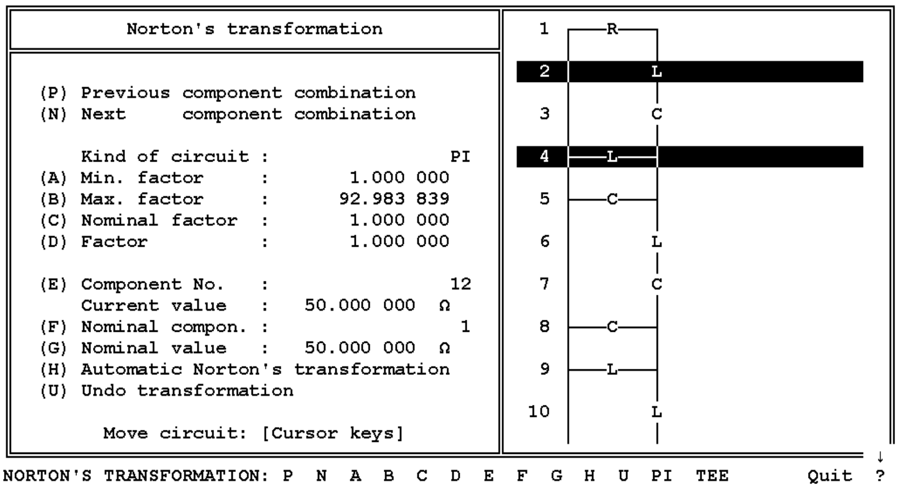


Use the arrow keys to move around screen. We can use the [↓] key to run through the circuit step by step. Notice that our ‘equal inductances’ specification is unfulfilled.

The next step is to modify the circuit with the appropriate transformations in order to optimize the component values to suit our requirements without affecting the filter’s transmission characteristics. To do this, we will use Norton’s transformation.

- [Q][M][N] Switch to NORTON’S TRANSFORMATION menu

The screen appears follows:



The position of the “bars” in the circuit shows which combination of components can be subjected to Norton’s transformation. These two elements are transformed to 3 elements and the impedance level of the circuit behind these elements is changed.

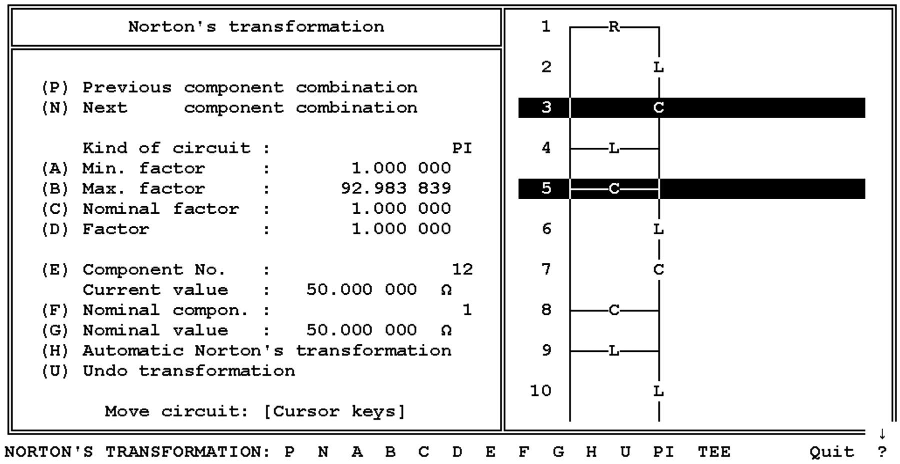
However, this is an L combination - which, of course, we can’t use for a transformation (after all, we want equal inductances).

So let’s move to the next possible combination of components:

- [N] Next combination of components which can be subjected to Norton’s transformation

Let's stop here. The components currently numbered 3 and 5 will be used for the transformation (numbers are reallocated each time the circuit is transformed).

Note: Don’t let the fact that the inductance we want to adjust is between the capacitors mislead you. *QuickFil* is clever enough to put L4 behind C5.



Now let’s enter the current number of the component we want to change.

- [E]4[↵] Indicates that component 4 should have the “nominal value” to be defined under “G” after the transformation

The current value of component 4 is approx. 33.19 nH. We will use the calculated value of the first inductance, L2, as the nominal value.

- [F]2[↵] After the transformation, component 4 should have the same value as component 2

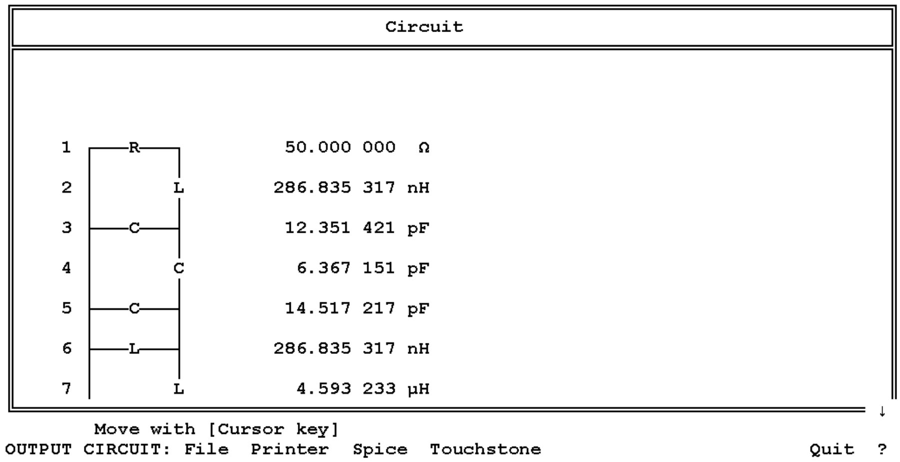
We enter the necessary transformation factor in field (C). Since it is between the limits specified in (A) and (B) (anything outside these limits will usually give negative component values), we can run the transformation using this nominal factor.

- • [C] Enter nominal factor as the transformation factor in field (D) and run the transformation

QuickFil transforms the two capacitors into a new circuit with three capacitors. The inductance, which was formerly numbered 4, now appears as 6. Now, let’s check whether this does in fact have the value we want.

- [Q][O] Switch to the CIRCUIT OUTPUT menu

The screen shows that it has clearly worked:



Now change the value of the next inductance, component 7.

- [Q][N] Back to NORTON’S TRANSFORMATION menu

Since the capacitor 8 and inductor 7 can be exchanged without any effect and the capacitor 5 and inductor 6 too, we can use the capacitor 5 and capacitor 8 for the next transformation. Using these two elements we don’t change the inductor 6 but we can change the inductor 7.

Let’s go on to this combination of components eligible for Norton’s transformation.

- [N] Next combination

We can’t use this either as it would change the value of 6.

- [N] Next combination

This combination will work because *QuickFil* is clever enough to avoid changing 6 (arranged before C5) and treat 7 as though it were past the components to be transformed (after C8).

We can now specify our transformation.

- [E]7[↵] Indicates that the component currently numbered 7 should have the nominal value entered in “G” after the transformation The current value of component 7 is approx. 4.59 uH; but we want this inductance to have the calculated value of L2. So we enter this as a nominal value once again.

- [F]2[↵] After the transformation, component 7 should have the value as component 2

We enter the transformation factor required in (C) once again. This is also within the limits set in (A) and (B); so there is nothing to stop the transformation.

- [C] Enter the nominal factor as the transformation factor in field (D) and execute transformation

QuickFil has transformed the two capacitors into a Pi circuit. The inductance, which was numbered 7, is now numbered 9. If you like, you can switch to the CIRCUIT OUTPUT menu again to see if this value is what we want.

Now if you try to transform the next inductor L11 to the same value as the other ones, you don’t find a element combination for Norton’s transformation. But there is a little trick. If you change the last PI circuit, which was built by the last Norton’s transformation, to a TEE circuit, you will find further possibilities for Norton´s transformation. You can change the circuit by the undo command in Norton´s transformation, change the circuit from PI to TEE and repeat the Norton’s transformation.

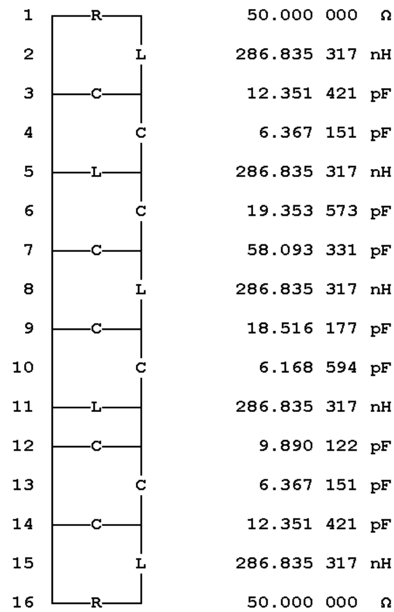
Alternatively you can transform the PI-circuit to a TEE-circuit by the special menu: PI-TEE conversion. we proceed the synthesis in following way:

- [Q][T][N][E] PI to-TEE conversion of capacitors 6, 7 and 8
- [Q][N][N][N][N] Select capacitors 8 and 10 for the next transformation, to optimize the value of L11
- [E]11[↵] Component 11 selected
- [F]2[↵] Value of component 2 selected as nominal
- [C] Execute transformation with nominal factor
- [Q][N][N] Select capacitors 11 and 14 for the next transformation, to optimize value of L13
- [E]13[↵] Component 13 selected
- [F]2[↵] Value of component 2 selected as nominal
- [C] Execute transformation with nominal factor

That's it. Let's look at the result.

- [Q][O] Switch to the circuit output

We can use the cursor keys to run through the circuit. We find that all the L's do in fact have the same value.



We will end this example by having a quick look at how you can use *QuickFil* to generate your own circuits piece by piece.

Instead of using the circuit *QuickFil* suggests, our aim here is to build up our own circuit as follows:

1st filter section: lowpass

2nd filter section: highpass

The first thing we need to do is to switch from the OUTPUT CIRCUIT menu, where we were at the end of the last section, back to the PASSIVE DESIGN menu.

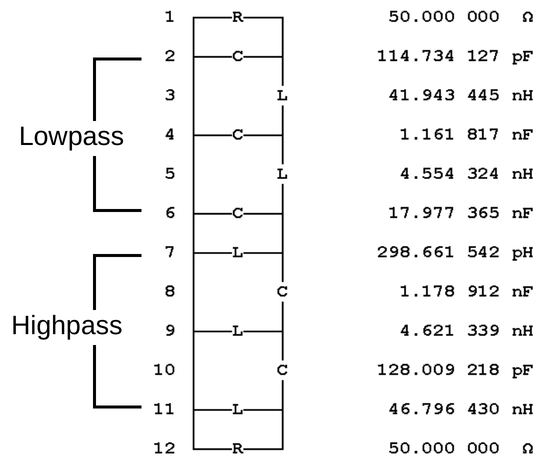
• [Q][Q][Y]	Back to the PASSIVE DESIGN menu, with the screen showing *QuickFil's* first suggested circuit
• [I]	Switch to the INPUT CIRCUIT menu
• • [H]	Delete the complete circuit

Now *QuickFil* will expect us to specify which components to put together on a modular basis. At each stage, *QuickFil* will check which components can be used next without affecting the filter's transmission characteristics.

We will only be allowed to select from among those components (highlighted). Our inputs run:

• [A]	Parallel capacitor
• [D]	Series inductance
• [A]	Parallel capacitor
• [D]	Series inductance
• [A]	Parallel capacitor
• [C]	Parallel inductance
• [B]	Series capacitor
• [C]	Parallel inductance
• [B]	Series capacitor
• [C]	Parallel inductance

Our circuit will now look like the following:



We could keep on rearranging this circuit to suit our own requirements, but that is outside the scope of this example. So let's end by going back to the main menu.

- [Q][Q]   Back to the main menu

SUMMARY

In this example, you have learned how to:

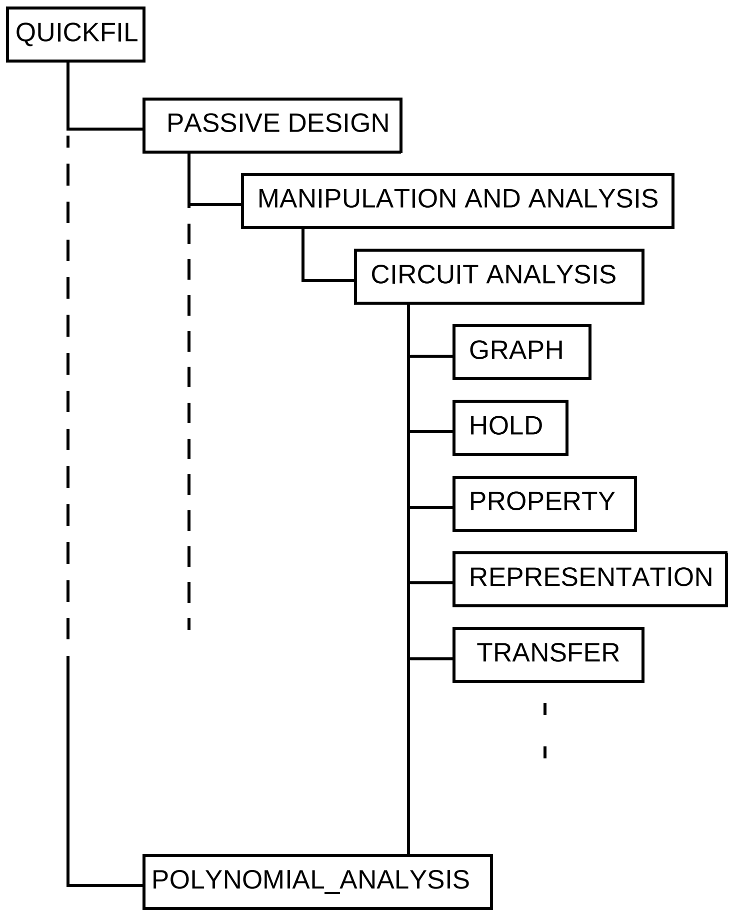
- Change a circuit suggested by *QuickFil*
- Use Norton's transformation facility
- Execute PI-to-TEE transformations, and
- Design circuits on a piece by piece basis.

5.35 Analysis

The filter's properties can be investigated graphically in one of two levels:

- Analysis based on the determined polynomial roots.
- Analysis of calculated circuits.

Before entering the actual graphics section, the waveforms which are to be entered in the diagrams and their mode of display, are determined. The respective program section has the following structure:



You can use this menu to analyze the characteristics of *QuickFil* circuits (passive analysis) or simple analyze the characteristics of the transfer function. Both menus POLY. ANALYSIS and CIRCUIT analysis are about the same menus, there are little differences only.

```
Diagram 1: Transfer function Magnitude [dB]
Diagram 2:
Diagram 3:
Diagram 4:
                                        Select diagram with [PgUp],[PgDn]
   10.000 000 dB
                   Property:                     Representation:
                                                 Stopband spec.
                   Transfer function             Magnitude [1]
                   Input impedance               Magnitude [dB]
                   Output impedance              Phase [rad]
                   Input reflection factor       Phase [°]
                   Output reflection factor      Group delay [s]
                                                 Real part [1]
                                                 Imaginary part [1]
 -100.000 000 dB
                  0.000 000  Hz                            200.000 000 MHz
Merged graphs:    0              100 Points           linear

Quality factor Induct.: infinite                  Capacit.: infinite
CIRCUIT ANALYSIS: Graph  Hold    Property  Repres.  Transfer  XDefault  YDefault
                  Y-from Y-to  X-from X-to  pOints  Lin Log   Ind Cap    Quit  ?
```

To analyze a filter;

- Select a diagram 1 - 4
- Select the property
- Select the representation
- Select the limits, scale, and number of analysis points
- Select the quality factors (optional)

Following menu items are available:

Graph

Switches to the graphics mode. The characteristic(s) will be displayed in accordance with the parameters entered in the dialog.

Hold

This menu item saves the diagram(s) last shown in the graphics section to a file called HOLD.GDF and then automatically integrates this file as future curves are loaded. This process can be repeated several times. HOLD.GDF will be overwritten, but the new data will be linked to the curves already loaded. The data will remain integrated until they are deleted in the GRAPH_LOAD menu.

Property

Here you can specify which property to use in the current diagram (highlighted). The required diagram is selected by using the arrow keys.

Note: The Property selected will only be entered in the diagram field once the Representation has also been specified.

Representation

Switches to the REPRESENTATION menu. Here you can specify the form by which the desired Property is to be shown in the diagram. The entry will be made in the appropriate field as soon as this is selected.

Note: When selecting the Magnitude [dB] of the transmission function, *QuickFil* allows you to include the tolerance scheme as specified in the STOPBAND SPECIFICATION menu. The respective setting is made in the ***withOutspec/ With_spec*** menu items, which is additionally offered in the menu line.

To delete a diagram field's input;

From the keyboard:

- Select the diagram field using [PgUp]/[PgDn] keys, then select Delete from the REPRESENTATION menu.

Using the mouse:

- Point the mouse at the diagram field and click on the corresponding input in the Representation section of the screen.

Transfer

Switches to the TRANSFER menu. This function is used to load and save graphics parameters and curves.

XDefault

Sets the X limits of the current diagram automatically using the following rules:

Linear scale:

X-from = 0

X-to: next highest default value (1 2 5 10 20 50 100) from 2 x f0

Logarithmic scale:

X-from = X-to/100

X-to: next highest default value from 5 x f0 f0 = reference frequency

In lowpass and highpass filters, the passband edge frequency.

In bandpass and bandstop filters, the center frequency (geometric mean of the passband edge frequencies).

YDefault

Sets the Y limits of the current diagram automatically. The system will try to find appropriate values in light of the desired diagram contents. The Y limits will be reset to the default values each time the desired display content is changed (property/representation).

Y-from, Y-to, X-from, X-to

Enables you to enter limits manually for each diagram.

pOints

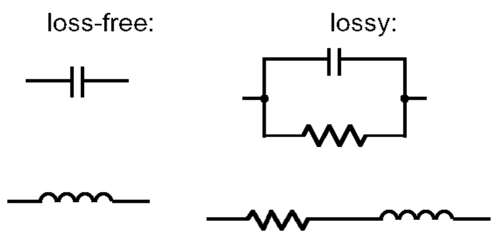
Indicates the number of computation points within the specified frequency range.

Lin, Log

Specifies the scale required for the diagram.

Ind, Cap

This menu item is available only for the menu CIRCUIT ANALYSIS:.
This feature enables you to consider losses in components. If a quality factor is entered, *QuickFil* allocates parallel resistors to capacitances and series resistors to inductances:



$$R = \frac{Q}{\omega C}$$

$$R = \frac{\omega L}{Q}$$

It's assumed, that the quality is independent of the frequency. The equivalent resistances is dependent on the frequency.

5.36 PROPERTY menu

For circuit analysis:

```
PROPERTY: Transfer_function          Input_impedance         Output_impedance
          Input_reflection_factor    Output_reflection_factor         Quit  ?
```

For polynomial analysis:

```
PROPERTY: Transfer_function      Pulse_response      Step_response      Quit  ?
```

You can use this menu to specify which property of the filter will be represented in the selected diagram. The menu line shows only those characteristics applicable in any given case.

Transfer_function

Here you can analyze the transfer function of the filter as defined in the appendix.

Input_impedance, Output_impedance

This menu item is only available for the menu CIRCUIT ANALYSIS: (passive filter analysis). Here you can analyze the input impedance or the output impedance, if the opposite port of the filter is terminated by the defined termination (resistor, open, short circuit).

Input_reflection_factor, Output_reflection_factor

This menu item is only available for the menu CIRCUIT ANALYSIS: (passive filter analysis). Here you can analyze the input reflection factor or the output reflection factor, if the opposite port of the filter is terminated by the defined termination (resistor, open, short circuit).

Pulse_response, Step_response

This menu item is only available for the menu POLY. ANALYSIS: (analysis of the transfer function directly). Here you can analyze the pulse and step response of the filter using the Laplace transformation.

Note: No entry will be made in the current diagram field (highlighted) until the desired representation has been selected.

5.37 REPRESENTATION Menu

You can use this menu to specify which form of the already chosen property you want to display. To insert the waveform in the appropriate diagram field, exit the menu by using Quit.

Note: The menu line will only show the representations available for the selected property.

Delete

Used to delete diagram field inputs.

To delete a diagram;

- Select a diagram field using the [PgUp] or [PgDn] keys.

Then select Delete. You don't have to leave the REPRESENTATION menu.

Magnitude [1]

The magnitude in linear scale of the complex function defined in the ***Property*** menu is shown in the diagram.

Magnitude [dB]

The magnitude in logarithmic dB-scale of the complex function defined in the ***Property*** menu is shown in the diagram. This representation isn't available for the input impedance and the output impedance in the CIRCUIT ANALYSIS menu.

When selecting the display “Magnitude [dB]” of the transmission function, *QuickFil* allows you to include the tolerance scheme as specified in the STOPBAND SPECIFICATION menu. The respective setting is made in the ***withOut_spec/With_spec*** menu items, which is additionally offered in the menu line.

Phase [rad]

The phase of the complex function defined in the ***Property*** menu is shown in the diagram (mathematical: arg function). In the menu POLY. ANALYSIS: it is possible to show a continuous phase response.

Phase [°]

The phase in degrees of the complex function defined in the ***Property*** menu is shown in the diagram (mathematical: arg function). In the menu POLY. ANALYSIS: it is possible to show a continuous phase response.

Group delay [s]

The group delay of the complex function, defined in the ***Property*** menu, is shown in the diagram. The group delay is the negative derivative of the phase response. If you have selected group delay as representation, it is also possible to show the specifications, if you add a group delay correction.

Real part [1]

The real part of the complex function, defined in the ***Property*** menu, is shown in the diagram.

Imaginary part [1]

The imaginary part of the complex function, defined in the ***Property*** menu, is shown in the diagram.

Voltage [V]

This representation is only available in the menu POLY. ANALYSIS. The voltage of the pulse response or the step response is shown in a diagram.

PulseStrength [Vs]

This representation is only available in the menu POLY. ANALYSIS. The pulse strength is the voltage-time integral (area) of the very short input pulse (Dirac pulse) and, is available if you select the property ***Pulse_response.***

Stepstrength [V]

This representation is only available in the menu POLY. ANALYSIS. The step strength is the voltage of the input signal step and, is available if you select the property ***Step_response.***

Phase_bounded, Phase_continuous

This representation is available only in the menu POLY. ANALYSIS if you select the property ***transfer function.*** If the option Phase_bounded is selected, you will get the standard phase representation, which is usually used in every analysis program. The phase is limited to the interval [$-\pi$ rad, π rad] or [-180°, 180°]. If you choose the option continuous phase, the program will build a phase function, which does not have jumps of a full period (2π or 360°) but, it can have jumps of half of a period.

5.38 TRANSFER menu

Menu line:

```
TRANSFER: Load(Parameters)       Save(Parameters)           Reset(Parameters)
          Load(Data)       Save(Data)          Delete(Data)              Quit  ?
```

This menu allows you to:

- Save diagrams.
- Link diagram contents.
- Save graphics settings for fast resetting.

Load(Parameters)

Loads settings saved with the Save(Parameter) option.

Save(Parameters)

Saves all settings (but not the data).

Reset(Parameters)

Resets graphics parameters to their default values and deletes selected properties in diagram fields 1 - 4.

Load(Data)

Activates the LOAD_GRAPH menu where you can load saved curves and merge them with the current diagram(s).

For more details see next section.

Save(Data)

Saves the last diagram(s) displayed in graphics mode. Default extension: .GDF.

Note: Curves can only be re-merged with diagrams in which they were included at the time they were saved. For example, a curve from diagram 2 cannot be displayed in diagram 3.

Delete(Data)

Deletes all diagram contents previously merged via the LOAD_GRAPH menu.

5.39 GRAPH_LOAD Menu

Use this menu to merge stored graphics data from memory with current diagram(s).

The menu options are as follows:

Load

Loads previously saved curves (together with their diagram's data) into the lower part of the screen where you can check their characteristics and units before merging them with the current display.

Data from the following functions can be loaded:

- Save(Data) in the TRANSFER menu.
- Save(Curve) in the MARKER menu.
- Hold in the POLY. ANALYSIS or CIRCUIT ANALYSIS menus (file HOLD.GDF).

Note: It is necessary to invoke the Merge function in order to include graphics in the diagram.

Merge

Use this command to merge loaded graphics data with the current display. Merged graphics are numbered in sequence. Newly merged data is added onto any already existing data.

Diagram parameters (units, axis data) are displayed in the upper part of the screen, but are lost in the display. The display continues to use the units and labels of the original diagram(s). Comments (input through the SPECIFICATION menu) are retained to identify the curve. If units conflict when merging, a message will be displayed.

Delete_last

Deletes last curve block merged at any time.

Delete_all

Deletes all merged curves (equivalent to the Delete(data) command in TRANSFER menu).

Selection(merged_graphs)

Displays a merged curve block at the top of the screen by selecting the number *QuickFil* allocated to it when it was merged.

Selection(loaded_graphs)

If the files INIPOLY.GDF or INIPASS.GDF are loaded (explanation below), the curve block determined by this command can be displayed at the bottom of the screen.

Note: You do not have to “copy and merge” stored graphics data each time you want to display them together. If a number of curve blocks have been merged when you exit the POLY. ANALYSIS or PASSIVE ANALYSIS menus then, these will be saved. They will be automatically retrieved the next time you go into analysis mode.

When you exit from the menu, the merged curve blocks will be stored in:

POLY. ANALYSIS: INIPOLY.GDF

PASSIVE ANALYSIS: INIPASS.GDF

5.40 Example: Elliptic Bandpass

In this example, we will investigate the Elliptic bandpass specified in Chapter 3 more closely.

If you have saved the data for this example, it is fairly easy to make the necessary initial settings.

Enter from the main menu:

- [T] Activate the TRANSFER menu.
- [↵] Select the Load menu item.
- [F1] List all filenames with the extensions.

Now select EXAMPLE1.QF (↑) and confirm it with [↵].

Briefly check the data in the SPECIFICATION menu.

- [S] Activate the SPECIFICATION menu.

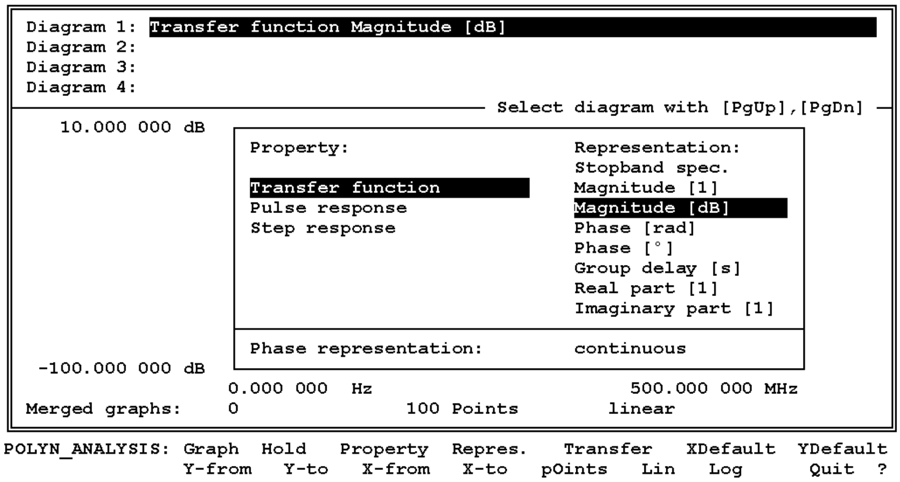
You should see the following screen:

```
          SPECIFICATIONS to :  Elliptic (Cauer) - bandpass filter
 (O)

 (A)  Lower passband edge frequency     :          87.500 000 MHz
 (B)  Upper passband edge frequency     :         108.000 000 MHz
 (C)  Lower stopband edge frequency     :          70.000 000 MHz
 (D)  Upper stopband edge frequency     :         135.000 000 MHz

 (E)  Passband bandedge loss            :             0.044 818 dB   ◄
 (F)  Passband bandedge return loss     :                 19.89 dB   ◄
 (R)  Passband reflection factor        :                 10.13 %    ◄
 (G)  Stopband loss                     :                 50.00 dB

 (H)  Filter degree                     :                        8
 (I)  Case                   (a, b, c)  :                        c
 (J)  Variable value (A,B,C,D,E,F,G,H,R) :                       E

      Lower 3dB edge frequency          :          84.698 809 MHz
      Upper 3dB edge frequency          :         111.571 817 MHz
      Filter quality                    :                   14.50
SPECIFICATION: A  B  C  D  E  F  R  G  H  I  J   New  cOmment   fiLe    Printer
               freqUencyrepres.  bandwithrepreS.  reL.bandwithrepres.   Quit  ?
```

Note: If the data on the screen does not coincide with this sample, please make the necessary changes. The procedure details are shown in Chapter 3.

Let’s investigate some of the filter’s properties more closely. Carry out an analysis on the basis of the calculated polynomial roots through which the filter is uniquely described:

- [Q] Return to the main menu.
- [A] Activate the polynomial_Analysis menu.



Accept the amount of the transfer function displayed in Diagram 1. Also include the tolerance scheme.

- [R] Activate the REPRESENTATION menu.
- [W][Q] Select the With_spec menu item.

Hint for mouse users: Click on the word “stopband spec.” directly on the screen to carry out these steps with the mouse.

Now read the diagram limits and the number of points to be calculated:

- [X][X][X][↵]200M[↵] Reduce the upper limit X to 200 MHz.
- [O]300[↵] Increase the number of points to be calculated to 300.

You can also click on the entry fields directly with your mouse.

Now on to Diagram 2. Here the phase-frequency characteristics are entered.

- [PgDn] Change to Diagram 2.
- [R][P][P][↵][Q] Set the transfer function phase [°].
- [X][X][X][↵]200M[↵] Reduce the upper limit X to 200 MHz.
- [O]300[↵] Increase the number of points to be calculated to 300.

Enter the group delay in Diagram 3:

- [PgDn] Change to Diagram 3.
- [R][D][↵][Q] Set the transfer function Group delay.
- [X][X][X][↵]200M[↵] Reduce the upper limit X to 200 MHz.
- [O]300[↵] Increase the number of points to be calculated to 300.

This completes the settings. However, one diagram is still free.

Let’s use this for displaying the pulse response.

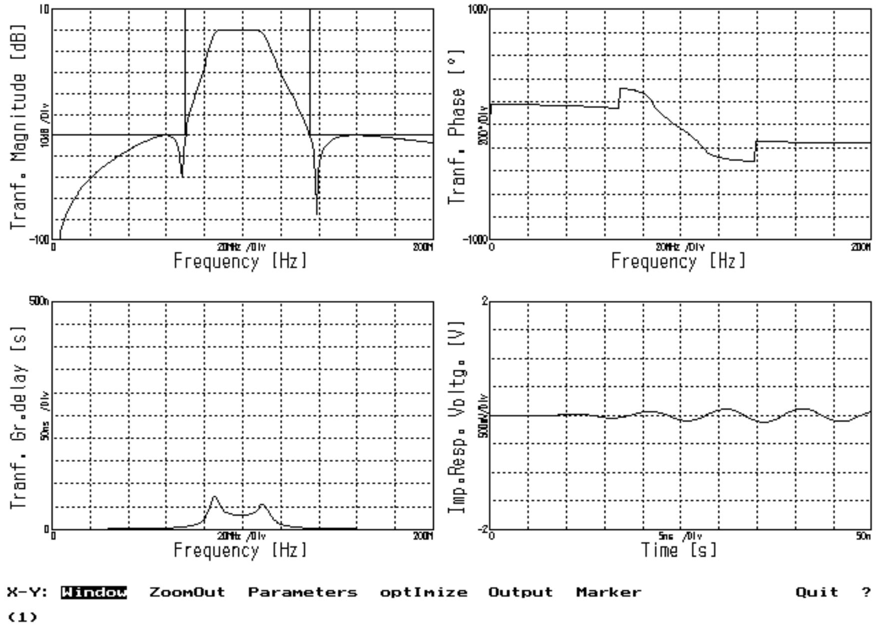
- [PgDn] Change to diagram 4.
- [P] Activate the PROPERTY menu.
- [P][Q] Select pulse response.
- [R][V][Q] I include in the diagram field.
- [O]300[↵] Increase the number of points to be calculated to 300.

After all settings are made, activate the graphics section.

- [↵] Select the Graphic menu item.

After a few moments (depending on the type of computer you have), you will see the following diagrams on your screen:

:



Naturally, you could make many changes to these diagrams. However, let’s only do this for the Y-limits of Diagrams 2 and 3.

• [PgDn]	Select Diagram 2 as the current diagram (stated on the lower left hand side of the screen in brackets).
• [I]	Optimize the Y-limits of Diagram 2.
• [PgDn]	Select Diagram 3 as the current diagram.
• [I]	Optimize the Y-upper limits of Diagram 3.

The contents of Diagram 3 are of particular interest. Let's zoom in on this diagram.

• [↵]	Activate the X-Y WINDOW menu.
• 3	Display Diagram 3 enlarged.

That is enough for the moment. Let’s have a look at how to delete the diagrams:

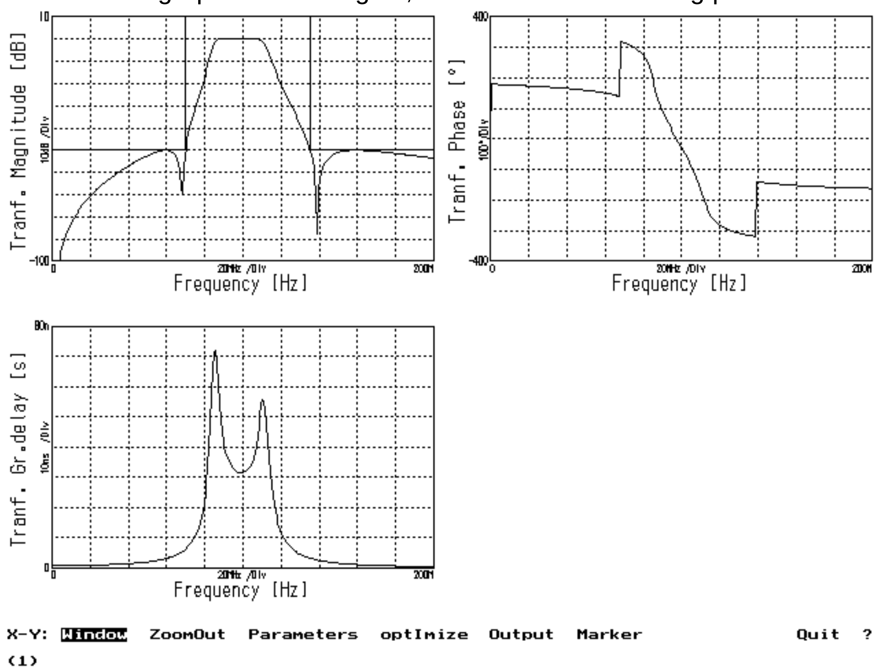
- [Q] Return to the POLY. ANALYSIS menu.

Let’s assume that we are no longer interested in the contents of Diagram 4 and want to delete it.

- [R] Activate the REPRESENTATION menu.
- [↵][Q] Select the Delete menu item for deleting the respective contents of the diagram.

Mouse users: The same effect can be achieved by clicking directly on the display that you want to delete. For example: click on entry Voltage [V].

If we enter the graphics mode again, we will see the following picture:



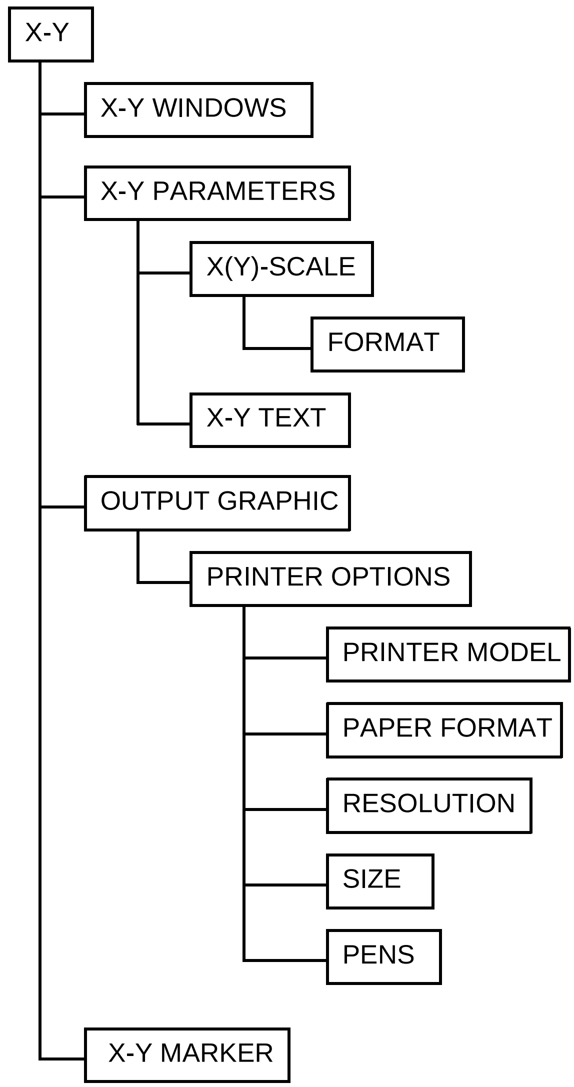
This completes our example. More details on the graphics section are provided in the next section.

Last step:

- [Q] Return to the main menu.

5.41 Graphics

After determining which properties of the filter should be displayed in the PASSIVE DESIGN or POLY. ANALYSIS menus, we can enter the graphics section of *QuickFil* through the Graph menu item. It is arranged as follows:



5.42 X-Y Menu

This menu can be used to:

- Change diagram parameters.
- Output graphics to plotter, printer, or file.
- Activate a waveform cursor.
- Select which diagram(s) are to be displayed.

QuickFil provides a special zoom function for enlarging portions of the diagram.

To zoom in on a portion of a graph:

- Place the mouse pointer on one corner of a desired section and click the left mouse button.
- Move the mouse. As the mouse is moved, a box will be created. Pull the box outline around the portion of the graph that you want to zoom in on.
- Click the left mouse button again to zoom in on the section.

You can return to the previous display by selecting the ZoomOut menu item.

The X-Y menu offers the following options:

Windows

After selecting this menu item (if there is more than one diagram displayed), the X-Y WINDOW submenu will be activated. This menu allows you to display either a “full view” of one of the displayed diagrams or, if this has already happened, to return to the display of all the diagrams.

Note: If you are currently in the “enlarged” mode, you can switch back to the “full” display of the previous or following diagram in the X-Y menu by using the [PgUp]/[PgDn] keys. You do not have to return to the simultaneous display of all the diagrams through the X-Y WINDOWS menu.

Mouse users: You can select the full display of a diagram by clicking on the diagram with the right mouse button. A further click on the right mouse button will bring you back to the display of all the diagrams.

ZoomOut

Only works after zooming in with the mouse. This menu item returns you to the display mode that was set before the “zoom” was executed.

Parameters

Takes you into the X-Y PARAMETERS menu where you can alter various diagram parameters. The current diagram is the one whose number is indicated in brackets on the lower left-hand side of the screen. The current diagram can be changed through the cursor keys [PgUp], [PgDn], [Home] and [End].

optImize

Automatic scaling of diagram axis/axes is set to “Auto” in XY PARAMETERS screen.

Output

Takes you into the OUTPUT GRAPHIC menu where you can output the diagrams to a:

- Plotter (in HPGL format);
- Printe; or
- File (HPGL format).

Marker

Switches to the X-Y MARKER menu. Here you can:

- Analyze curves in more detail using various cursor/marker functions.
- Optimize diagram limits with respect to a particular curve.
- Save individual curves from a diagram.

5.43 X-Y PARAMETERS Menu

This menu allows you to:

- Optimize diagram limits.
- Switch to other menus where you can modify other diagram parameters.
- Change the suggested diagram.

Options include:

Window

Indicates which diagram you wish to use.

optimize

Optimizes the diagram scaling.

X-Scale

Switches to X-SCALE menu where you can modify the following x-axis parameters:

- Axis limits.
- Scaling Grid (linear or logarithmic).
- Number of divisions.
- Method used for number display.

Y-Scale

Same as the X-Scale function.

Text

Switches to X-Y TEXT menu where you can change the suggested diagram labels.

Note: If desired, you can vary the size of the labels used for printing and plotting. For more details see the section on the default file (QF.DEF).

Data_points

Indicates whether the data points in the diagram should be connected by lines or not.

5.44 X(Y)-SCALE Menu

You can use this menu to set the following parameters:

- Axis limits.
- Scaling grid (linear/logarithmic).
- Number of subdivisions.
- Method used for number display.

Options Include:

Auto

Determines whether the y-axis can be automatically rescaled from the X-Y menu.

From

Accesses the lower y-axis limit input field.

To

Accesses the upper y-axis limit input field.

Lin

Selects linear y-axis scaling for this diagram.

Log

Selects logarithmic y-axis scaling for this diagram. If the spread between the upper and lower limits is less than one decade, only linear scale is allowed.

Divisions

Indicates the y-axis grid divisions. The variable “Units/ division” changes each time this field is changed.

format(Number)

Takes you to the FORMAT menu where you can specify how numbers are to be shown in the fields and on the y-axis.

Decimals

If you select this option, the system will ask for the number of decimal places you want. This applies to all display formats. If, for example, you specify 3 decimal places, the various display modes will appear as follows:

Fixpoint

Use this option to indicate that all figures must appear in fixed format mode. For example: 12345.000

Scientific

Display in powers of 10. For example: 1.234E+04

Engineer

Display in powers of 10 with the exponent divisible by 3. For example: 12.345E+03

Technical

Technical display mode. For example: 12.345 k

Y(X)-Scale

Takes you to the Y(X)-SCALE menu where you can specify the y(x)-axis parameters for the current diagram including:

- Axis limits.
- Scaling Grid (linear or logarithmic).
- Number of divisions.
- Method used for number display.

5.45 OUTPUT GRAPHIC Menu

You can use this menu to output the screen contents to a:

- Plotter, (HPGL format)
- Printer, or
- File (HPGL format)

The following options are available:

prInter

Prints the diagram to printer. The printer settings are made via the printerOptions menu item.

printerOptions

Move to the PRINTER OPTIONS menu where various printer settings can be changed.

Plotter

Selecting this option causes *QuickFil* to start to plot the diagram on the plotter (HPGL format).

File

Saves the diagram in a file using the HPGL format (Hewlett-Packard Graphics Language). This file can then be plotted at a later time from the DOS prompt, moved to another system for plotting, or inserted into a desk-top publishing program. Default extension: .PLT

5.46 PRINTER OPTIONS Menu

The printer settings for graphics output are made in this menu.

Options include:

to_Printer/to_File

Determines whether the output is made directly to the printer or to a file.

When selecting File, the contents of the file can be sent to the printer at a later time through the DOS copy command.

For example: “Copy filename LPT1”

Model

Selects the type of printer.

Format

Moves to a submenu where the paper format can be selected. The menu items “Length” and “Width” allow you to set any desired size.

Resolution

In this submenu, you can determine your printer’s resolution. Some printers only allow one resolution.

Invert

Switches from normal to inverse video display.

Rotate

Rotates the output by 90°.

Copies

Determines the number of copies to be printed.

Size

Switches to a submenu used for setting scale factors and displacements.

Pens

Move to the PENS submenu where the line attributes (width, scale of grey) for the individual diagram components can be set.

Pen Number Object

1 ... 8 curve 1 ... 8

9 logo

10 diagram border

11 gridpoint (at logarithmic scaling)

12 gridline (at linear scaling)

13 text y-axis

14 text x-axis

15 label y-axis (for printer/file output)

16 axis number

Options in the PENS menu include:

- Selection of the desired “pen” with the arrow keys.
- Width: Sets the line width in pixels.
- Grayscale: Sets the grey scale (1...light grey, 9....black).
- Copy: The values of the current pen are copied to all other pens.

5.47 X-Y MARKER Menu

This menu allows you to analyze the curve(s) displayed in more detail by using a marker. The marker is also known as a waveform cursor. The following paragraphs answer most common cursor questions.

Where and how do the cursors appear when I switch to this menu?

When you select the Marker option from the X-Y menu, a crosshair cursor will appear in the diagram marked “current”. The current diagram is shown on the screen on the bottom left-hand side and can be changed by using the [PgUp], [PgDn], [Home] and [End] keys. The cursor can be moved to the left or right using the cursor keys. The cursor will stop at the calculation points. The X and Y coordinates of the cursor’s position will be displayed at the bottom of the screen.

Mouse users: The cursor can also be positioned in the X-Y MARKER menu with the mouse (left mouse button). Only the X-position of the mouse is taken into account. The Y-position will be determined by the selected curve.

What happens if I select another option?

Coarse

Causes the step width between the calculated points on the x-axis to be increased by a factor of 2.

Fine

Causes the step width to be reduced by a factor of 2.

optimize(Curve)

Optimizes the limits of a diagram for a given curve on that diagram (the curve on which the cursor is positioned). This diagram must have the “autoscale” mode set for at least one axis in the parameters menu.

If there is more than one diagram, how do I switch from one to another?

To move the cursor between diagrams, use the [PgUp], [PgDn], [Home] and [End] keys.

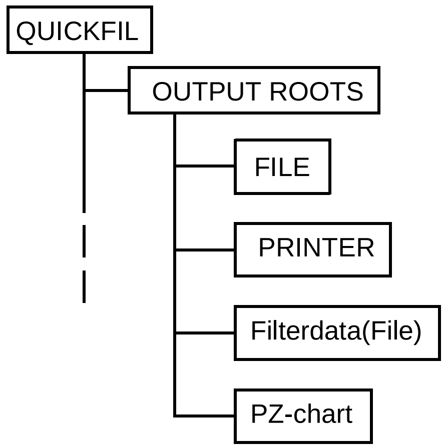
If there is more than one curve in a diagram, how do I switch to another curve?

You can use the arrow keys ([↑] and [↓]) to move the cursor from one curve to another. If you are using a color monitor, the appropriate line and coordinate details are shown in different colors, depending on which curve the cursor is positioned.

5.48 Output of the roots of the polynomials

(QUICKFIL:/Roots)

In the menu OUTPUT ROOTS: you can investigate the roots of the different polynomials. You can save the roots as a text file or print them.




```
                                Output Roots

  Reference frequency      :        69.208 020 MHz
  Reference quality factor :             3.295 620


  Normalized natural modes
  ------------------------

     -0.361 644 716 246
     -0.360 816 734 677
     -0.170 512 680 076  +-j    1.014 336 462 353
     -0.128 820 623 915  +-j    0.908 973 458 806
     -0.125 762 679 983  +-j    1.116 255 589 789
     -0.042 752 448 941  +-j    0.863 910 978 937
     -0.041 732 165 666  +-j    1.159 101 490 813
     -0.087 960 364 744  +-j    0.950 884 795 310
     -0.087 529 407 857  +-j    1.070 581 733 932

        Move with [Cursor keys]
OUTPUT ROOTS: File  Printer  Filterdata(File)  PZ-chart                Quit  ?
```

Following menu items are available:

File

Here, you can save the roots of the polynomials, as seen on the screen, as a text file in ASCII format.
Default file extension: *.FDT

Printer

Here, you can print the roots of the polynomials, as seen on the screen, to a line printer.

Filterdata(File)

Here, you can save the roots of the polynomial in a ASCII text format, which can be read by other programs.
Default file extension: *.QFT

PZ-chart

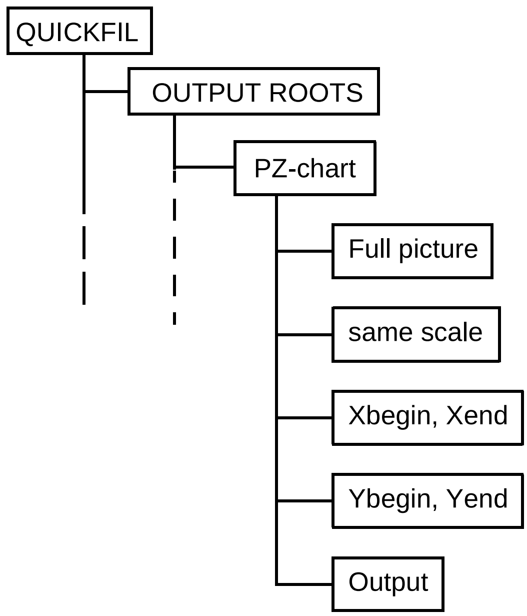
Here, you will jump to the menu PZ-CHART:, where you can show the poles and zeros of the transfer function G(s).

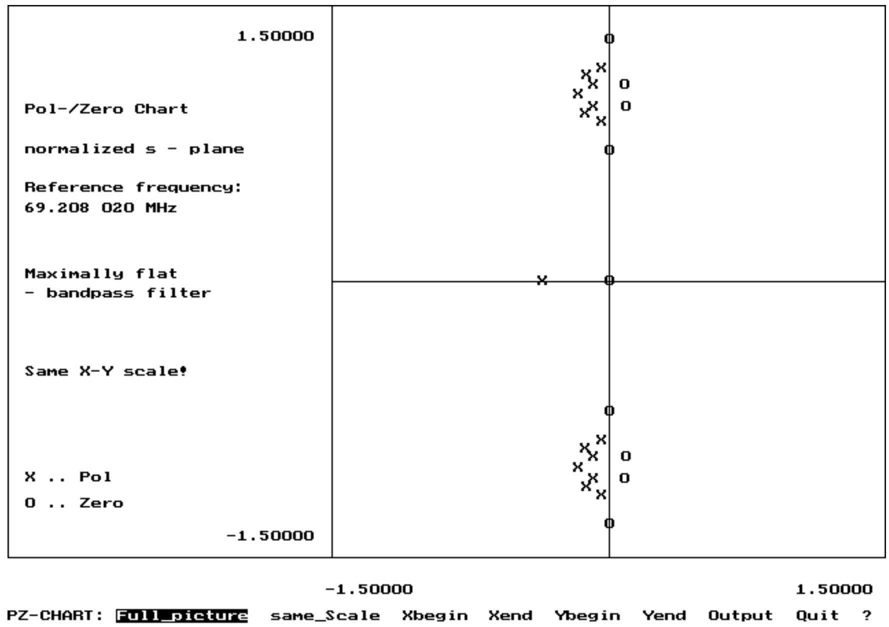
5.49 PZ-chart

(QUICKFIL:/Roots/PZ-chart)

In this menu, you can investigate the poles and zeros of the transfer function graphically. Poles and zeros are shown on the complex plane (x-axis: real part, y--axis: imaginary part). You can investigate special parts of the complex plain by zooming in.

.





Following menu items are available:

Full_picture

All the poles and zeros are shown on the screen. *QuickFil* will choose the best possible scales for the x-axis and the y-axis. Both scales of the x-axis and the y-axis are the same.

same_Scale

Here, you can force the x-scale and the y-scale to be the same. If both scales are the same, it is shown in the left part of the screen.

Xbegin, Xend

Here, you can define the beginning and the end of the x-axis. Alternatively, you can use the mouse and click directly to the input fields.

Ybegin, Yend

Here, you can define the beginning and the end of the y-axis. Alternatively, you can use the mouse and click directly to the input fields.

Output

You will jump to the menu OUTPUT GRAPHIC: where you can print, plot,.. the shown graphic.

For better investigation of the PZ-chart, you can zoom inside the diagram by using the mouse. Click the left mouse button at an edge of the interested field. Move the mouse and you will see a rectangle, which is part of the chart and will be shown next. By clicking the left mouse button a second time, you will zoom in the PZ-chart, as defined by that rectangle.

Note: If you have clicked the left mouse button at the wrong position, you can undo the zooming option by pressing the escape key.

To zoom outside of the diagram, use the right mouse button.

5.50 Macros

QuickFil allows you to record certain sequences of commands and data, and to replay them, whenever desired. Below, are some details on recording keystroke macros.

Generation of keystoke macros

The first step consists of opening a macro. Enter the MACRO menu directly from the main menu.

Depending on how you intend to invoke the macro, you can either:

- Save the macro under a filename (Record menu item); or
- Save the macro under a function keys (or a combination of function keys (function_Key menu item)).

In the latter case, *QuickFil* allows you to use a maximum of 48 different key combinations (depending on the keyboard):

[F1] to [F12]

[Shift+F1] to [Shift+F12]

[Ctrl]-[F1] to [Ctrl+F12]

[Alt+F1] to [Alt+F12]

After opening the macro, you will be in the program’s “record” mode. From this point forward, each keystroke will be recorded until you exit the record mode by pressing [Ctrl+z], or by exiting the program.

The main advantage of assigning macros to function keys, rather than storing them as files, is that function key macros can be started from any menu. To start a macro stored under a filename, you must first change to the MACRO menu.

While you are in the “record” mode, you have the following options:

Set an interruption

If you press [Ctrl+p] during the recording of a keystroke macro, the macro will stop at this point when replayed. This gives you the ability to adjust various entries, and change the result of the macro.

Practical example: When developing a parametric bandpass filter, the values of the components can be optimized by changing the parameters. In order to avoid having to re-enter all the data in each optimization step, we recommend using a keystroke macro.

Output a text file or message/provide a wait loop

If you press [Ctrl+u] during the recording of a keystroke macro, the RECORD menu will appear in the message line. Here, you can determine whether a text file is displayed, a message is displayed, or a wait loop is initiated when the keystroke macro is played back.

Invoking key macros

Keystroke macro files can be invoked in various ways:

To start a macro file from DOS prompt:

- Add the name of the respective macro file to the program name “QF” (separated by a blank). You do not have to add the extension .KDO.

For example (only for DOS application):

- QF approx[↵] Start *QuickFil* and immediately begin the macro APPROX.KDO.

Note: Macros can only be invoked in this way if they were saved to the same directory where the *QuickFil* program is stored (key macro files are not searched according to the path name).

To start a macro file from within the program:

For filename macros:

- Select the filename (with or without the path name) in the MACRO menu, under the Call menu function.

The execution of the macro begins immediately after confirming the entry in the field.

For function key macros:

- Press the appropriate key combination assigned to the macro.

The program automatically changes to the main menu where the macro commences.

The default extension of macro files is .KDO.

For more information on the macro file format, see the section on “Macro Files”.

The following “options” can be built into macros:

- Predefined interruptions in the automatic execution of the macro. This allows you to change the course of a macro.

To continue the macro, press [Ctrl+p]. Remember to return to the program section where the macro was interrupted!

- Time or wait loops, during which the contents on the screen remain displayed, a message or text file can also be displayed on the screen.

Changing the desired mode of execution

If the key combination [Ctrl+u] is pressed before invoking a macro, a menu will appear in the message line, which allows you to determine whether you intend to:

- Continue the execution after pressing the space bar instead of using the waiting periods (0 .. Waitmode);
- Take into account the waiting periods provided in the macro file (1 .. Timemode); or
- Process the macro file without considering waiting periods or interruptions (2 .. Fastmode).

Note: When creating macro files through the key combination [Ctrl+u], the RECORD menu is called up instead of this menu.

5.51 Example: Keystroke Macros

When developing a parametric bandpass filter, the component values can be optimized by changing the parameter field. To avoid having to enter all the specifications again in each optimization step, we can use the keystroke macro function.

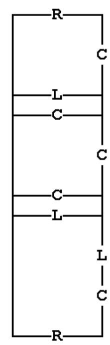
The desired structure of the circuit is predefined. This example illustrates how to enter your own circuit structure.

See the “Specification” and “Parametric Bandpasses” sections for more information.

The filter must fulfill the following requirements:

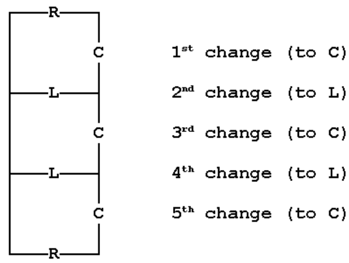
- Parametric equal ripple bandpass;
- Passband: 100 kHz to 120 kHz; and
- Passband loss: 1 dB.

Predefined structure of the circuit:



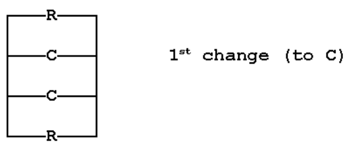
First step: Determine the number of transmission zeros at zero/infinity and pairs of transmission zeros at finite frequencies (you can proceed in accordance with the rules set out in the “Design with user-defined circuits” section):

Number of transmission zeros at zero:



Result: 5 transmission zeros at zero.

Number of transmission zeros at infinity:



Result: 1 transmission zero at infinity.

Number of pairs of transmission zeros at finite frequencies

There are no pairs of transmission zeros because the circuit does not include any resonating circuits.

The filter in question is a bandpass of the 6th degree (5 + 1).

Now, let's enter the data.

Opening the macro

From the main menu:

• [↵]	Activate the FILTERTYPE menu.
• [↵][B]	Set Bandpass.
• [A][↵]	Set equal ripple approximation.
• [Q][↵][S][N]	Activate SPECIFICATION menu, delete data (if any).
• [P][↵]	Select parametric type of bandpass.
• [A]100k[↓]	Lower passband edge frequency = 100 kHz
• 120k[↓]	Upper passband edge frequency = 120 kHz
• 1[↵]	Passband loss = 1 dB
• [G]5[↵]	Set 5 transmission zeros at zero.
• [D]	Set the default parameter.
• [Y]	Calculate the filter quality.

You will see the following screen:

```
            SPECIFICATIONS to :  Equal ripple - bandpass filter
(O)

     Kind of bandpass                  :                parametric
(M)  Parameter                         :            109.544 512 kHz

(A)  Lower passband edge frequency     :            100.000 000 kHz
(B)  Upper passband edge frequency     :            120.000 000 kHz

(E)  Passband bandedge loss            :               1.000 000 dB
(F)  Passband bandedge return loss     :                    6.87 dB
(R)  Passband reflection factor        :                   45.35 %

(G)  Transm. zeros at zero      :   5  | Fixed transm. zero pairs:   0
(H)  Transm. zeros at infinity  :   1  | Var. transm. zero pairs :   0

     Filter degree                     :                         6
     Lower 3dB edge frequency  (≈)     :             98.385 603 kHz
     Upper 3dB edge frequency  (≈)     :            122.959 982 kHz
     Filter quality                    :                     11.80

SPECIFICATION: M  A  B  E  F  R  G  H   qualitY   Conventional    Defaultparam.
               Stopbandopt. fiXtrans.zeros  cOmment  New  fiLe Printer  Quit  ?
```

We will develop the filter based on this data. The optimization of the component values is achieved by changing the parameter field in the SPECIFICATIONS menu. As the same entries have to be made repeatedly, we will use the keystroke macro function:

- [Q] Return to the main menu.
- [M] Activate the MACRO menu.
- [K] Select the function_Key menu item.
- [F10] Allocate the following macro to the function key [F10].
- [S] Activate the SPECIFICATION menu.

Insert an interruption here, so that the value of the parameter can be changed during the execution of the macro. The remainder of the calculations will be continued automatically.

- [Ctrl+P] Insert an interruption into the keystroke macro.

Now, let's go on to the circuit design.

- [Q][D] Activate the PASSIVE DESIGN menu.

The circuit proposed by *QuickFil* is not of interest. Delete it and replace it with your own circuit.

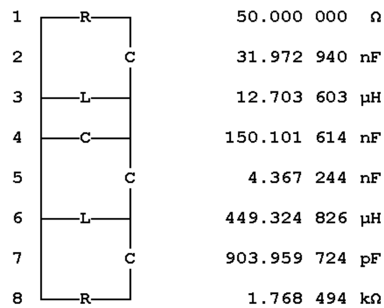
- [I][H] Activate the INPUT CIRCUIT menu,

delete the present circuit,

Now create the circuit, piece by piece.

- [B] Series capacitance
- [C] Parallel inductor
- [A] Parallel capacitance
- [B] Series capacitance
- [C] Parallel inductor
- [B] Series capacitance

The circuit should look like this:



The missing C is added through the Norton's transformation, which is necessary to achieve equal terminating resistances:

- [Q][M][N] Activate the NORTON’S TRANSFORMATION menu.
- [N] Switch to the next combination of elements, which can be applied to the Norton's transformation.
- [F]1[↵] Enter the value of the termination input resistance as the reference value of the component. The reference transformation factor (C) lies within the

limits and, therefore, the transformation can be carried out.

- [C] — Carry out the transformation in accordance with the reference factor.

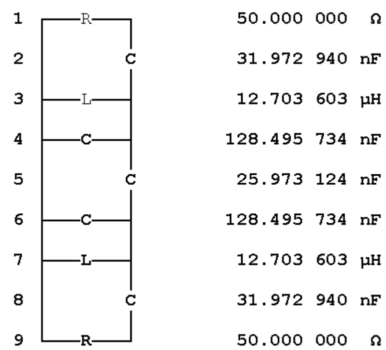
Now you have your circuit. Take a closer look at the values of the components.

- • [Q][O] — Output of the circuit.

Before investigating the circuit more closely, end the keystroke macro.

- [Ctrl+Z] — End the key macro.

You have calculated the following circuit:



Note: If your circuit shows other values, you might have forgotten to enter the value of 50 Ohms in the TERMINATION menu. You can enter this menu from the PASSIVE DESIGN menu.

You can scroll through the circuit with the arrow, [↓] and [↑], keys. You will see that all inductor values have the same value. However, we would like to have them at a more “nominal” value.

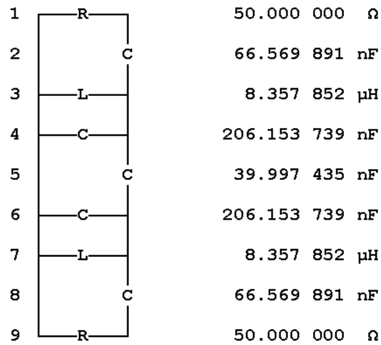
Try changing the parameter to change the inductor values to 10µH:

- • [F10] — Invoke the previously generated key macro.

If all entries were made correctly, the keystroke macro should stop in the SPECIFICATION menu. Then enter:

- [M]50k[↵] — Change the parameter to 50 kHz.
- [Ctrl+P] — Continue the macro.

After carrying out the macro, the following circuit will appear on the screen:



The values of the inductors reveal you have overdone the change slightly. The value of the parameter field was changed too much. The best value for the parameter actually lies between the default value (approx. 110 kHz) and 50 kHz. After several tries, you will find that the optimum value is: 77.38 kHz.

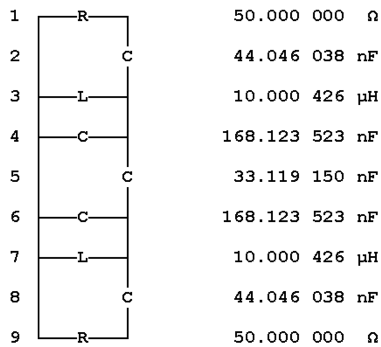
Again, execute the macro with this value.

- [F10] Start the macro.

When the interruption in the SPECIFICATION menu occurs, enter:

- [M]77.38k[↵] Change the parameter to 77.38 kHz.
- [Ctrl+P] Continue the macro.

After completing the macro you will have the following circuit:



That’s it! This completes the example.

- • [Q][Q][Y][Q] Return to the main menu.

6 ISSPICE Interface

6.1 QuickFil Output to ISSPICE

A special feature of the *QuickFil* program is its direct interface with the *ISSPICE* and *ICAPS* simulation system. Filters, designed with *QuickFil*, can be saved in one of two *SPICE* compatible netlists formats. As a stand-alone netlist, your filter can immediately be simulated in *ISSPICE,* or any other *SPICE* compatible simulator. This provides access to the powerful features of *ISSPICE*, such as time domain and temperature analyses. As a subcircuit netlist, your filter design can easily be integrated with the rest of your circuit, through the use of the *SPICENET* schematic entry module in *ICAPS*.

Component tolerances may also be included in either netlist format, allowing Monte Carlo statistical yield analyses to be conducted on your filter, when used in conjunction with Intusoft's *ICAPS* package.

6.2 Stand-alone Netlist

This example will show how to use *QuickFil*'s *ISSPICE* interface to create a stand-alone netlist. The filter specification is shown next.

- Center frequency: 70 MHz
- Bandwidth: 27 MHz
- Upper stopband limit frequency: 97 MHz
- Return loss: 20 dB
- Stopband attenuation: 25 dB
- Approximation: Chebyshev

After this information is entered, proceed to the *SPICE* menu.

- [D] Activate the PASSIVE DESIGN menu.
- [O] Activate the OUTPUT CIRCUIT menu.
- [S] Activate the SPICE menu.

The following screen will appear as:

```
                  Spice - Output: Standalone Analysis
 *
 * QuickFil  5.1           OMICRON electronics    2002-06-20  18:48:22
 * Chebychev - bandpass filter
 *
 * Lower passband edge frequency   :       66.500 000 MHz
 * Upper passband edge frequency   :       83.500 000 MHz
 * Lower stopband edge frequency   :       57.244 845 MHz
 * Upper stopband edge frequency   :       97.000 000 MHz
 * Passband bandedge loss          :         0.043 648 dB
 * Stopband loss                   :             25.92 dB
 * Filter degree                   :                  8
 * Case                            :                  b
 *
 L2         2   200    436.849N
 *  Q=100.00
 RL2      200     3    2.29191
 C3         3     4    10.4424P
           Move with [Cursor keys]
SPICE: Save  Standalone  Subcircuit  Name  Tol  R-tol  L-tol  C-tol    qualitY
       L-qual. C-qual.  fBegin fEnd  fDefault  Points  Lin Dec  Digits  Quit ?
```

Notice that the representation is already in a stand-alone format. If the format was previously changed, it will have to be changed back to stand-alone by typing:

- [S][↵] Activate the Standalone output format.

The *ISSPICE* analysis is controlled by the following parameters:

fBegin

This is the FSTART parameter in the .AC statement (the starting frequency).

fEnd

This is the FSTOP parameter in the .AC statement (the ending frequency).

fDefault

Invokes the *QuickFil* defaults for the fBegin and fEnd parameters.

Points

This controls the number of points taken by the AC analysis.

Lin, Dec

This specifies how the Points will be taken, either linearly over the entire frequency span, or per decade of frequency.

The syntax for the .AC statement is repeated for reference:

.AC [LIN, DEC, OCT] [Number of points] [fStart] [fStop]

Since the center frequency is 70MHz and the bandwidth is 27MHz, satisfactory start and stop frequencies for the .AC statement are 10MEG and 100MEG, respectively.

- [B] Edit the fBegin field.
- 10M[↵] Enter 10MEG as the start frequency.
- [E] Edit the fEnd field.
- 150M[↵] Enter 150MEG as the stop frequency.

We will collect data in this frequency range linearly, by entering LIN in the .AC statement.

- [L][L][L][↵] Enter LIN into the .AC statement.

Over this linear scale we will collect 200 points.

- [P] Edit the Points field.
- 200[↵] Enter 200 points.

At any point, the number of digits used, after the decimal point for the component values, can be altered. This is done by selecting the Digits option. We will now alter the netlist to use 4 significant digits for all values.

- [D][D][D][↵] Edit Digits field.
- 4[↵] Enter 4 as the number of significant digits.

At this point, we can save this circuit for simulation.

- [↵] Enter Save field.
- CIRCUIT1[↵] Enter a name for the file.

The file CIRCUIT1.CIR can now be simulated by *ISSPICE*.

6.3 Subcircuit Netlist

This example will demonstrate how to output the filter as an *ISSPICE* subcircuit. It may then be used with the *ICAPS* package.

In addition to demonstrating the subcircuit feature, we will use *QuickFil* to change the quality factor Q of the inductors in the filter.

We will use the filter from the previous circuit as a foundation.

At this point type:

- [S][S][↵] Activate the Subcircuit representation of the filter circuit.

The output will be shown in the standard *ISSPICE* subcircuit format, with three connections to the filter - input, output, and reference (normally ground). The netlist will also contain information about the particular filter in the form of the approximation, reference frequency, and input and output resistances. An example is shown below:

```
                   Spice - Output: Subcircuit
 .SUBCKT  FILTER     2   500   100
 * 2-Input  500-Output  100-Reference
 *
 *
 * QuickFil  5.1           OMICRON electronics    2002-06-20  18:40:51
 * Chebychev - bandpass filter
 *
 * Lower passband edge frequency   :      66.500 000 MHz
 * Upper passband edge frequency   :      83.500 000 MHz
 * Lower stopband edge frequency   :      57.244 845 MHz
 * Upper stopband edge frequency   :      97.000 000 MHz
 * Passband bandedge loss          :        0.043 648 dB
 * Stopband loss                   :            25.92 dB
 * Filter degree                   :                 8
 * Case                            :                 b
 *
 *  Rin  = 5.000000E+01
            Move with [Cursor keys]
SPICE: Save  Standalone  Subcircuit  Name  Tol  R-tol  L-tol  C-tol    qualitY
       L-qual. C-qual.  fBegin fEnd  fDefault  Points  Lin Dec  Digits  Quit ?
```

To investigate the filter further, the quality factor of a component can be altered from its ideal initial position. Quality factors can be added in one of two ways: either on a particular type of component (i.e. all inductors or all capacitors); or on an individual component. In this case, we want to investigate the effect of low inductor Q on the performance of the circuit.

- [L][L][↵] Edit the L-qual. field.
- 20[↵] Enter a quality factor of 20 to all inductors.

The subcircuit representation will change to show the effect of the new inductor Q. Series resistances will be added to all the inductors in the circuit. A comment line is also added to each inductor/resistor combination as a reminder of the Q value.

SUBCIRCUIT NETLIST

The subcircuit will now appear as:

```
                   Spice - Output: Subcircuit
 *  Rin  = 5.000000E+01
 *  Rout = 6.111111E+01
 *
 L2          2   200    436.849N
 *  Q=20.00
 RL2       200     3    11.4595
 C3          3     4    10.4424P
 L4          4   201    18.8520N
 *  Q=20.00
 RL4       201   100    494.531M
 C5          4   100    241.977P
 L6          4   202    739.376N
 *  Q=20.00
 RL6       202     5    19.3955
 C7          5   500    6.16974P
 L8        500   203    31.9074N
 *  Q=20.00
          Move with [Cursor keys]
SPICE: Save  Standalone  Subcircuit  Name  Tol  R-tol  L-tol  C-tol    qualitY
       L-qual. C-qual.  fBegin fEnd  fDefault  Points  Lin Dec  Digits  Quit ?
```

Now, the subcircuit can be saved.

- [↵] Activate the Save field.
- FILTER[↵] Enter the name FILTER.

The file will be saved using the default extension of .LIB.

This is how multiple filter designs are saved in a single library file. The library file will be in the format compatible with the *PRESPICE* program. However, in order to distinguish one filter from another in the library, it is necessary to change the name of the subcircuit using the Name option. Do this by typing:

- [N] Edit the Name field.
- FIL1[↵] Enter the name for the altered subcircuit.

Notice that the name in the .SUBCKT line changes. There can be any number of filter subcircuits in a library file. However, each must have its own unique name.

Interface With *The SPICENET Program*

The easiest way to use the filter subcircuit is with the *SPICENET* schematic entry program. A symbol called FILTER has been created for *SPICENET.* It will interface directly to the filters saved in the subcircuit format. The filters must be stored in a library file called FILTER.LIB because the symbol causes an *INCLUDE Filter.Lib to be entered into the final *ISSPICE* netlist.

To include a *QuickFil* designed filter in a *SPICENET* schematic:

- Create the filter and select the Subcircuit netlist format in the SPICE menu.
- Change the name of the filter to a unique name (FIL1, for example).
- Save the filter subcircuit netlist into a file called FILTER.LIB.

Make sure that the FILTER.LIB file is stored in your *SPICE* model library directory, normally C:\SPICE\PR.

- Enter *SPICENET* and place the symbol called FILTER.
- Select the FILTER symbol and display its label menu. Enter the unique name you gave the filter (FIL1, for example) in the value field. The name in the value field links the symbol to the unique name used in the .SUBCKT line.
- Enter “*INCLUDE FILTER.LIB” in the control menu.

Stand-alone Netlist With Monte Carlo Syntax

The *SPICENET* schematic entry program is included in the *ICAPS* simulation system.

The Get Symbol function can be used to get the Filter symbol.

The last example will deal with placing Monte Carlo tolerances on components using a stand-alone netlist. The method is easy and straight forward. We will use the same filter so there is no need to re-enter any information. However, we will have to remove the Q placed on the inductors in the previous example and switch the output to Standalone.

- [L][L][↵] Edit the L-qual. field.
- 0[↵] Enter a 0 Quality factor. This changes the inductor Q to infinite.
- [S][↵] Activate the Standalone netlist option.

Now that the screen is showing a stand-alone netlist, we will enter device tolerances for the inductors and capacitors. We will place a 5% tolerance on the inductors and a 10% tolerance on the capacitors.

- [L][↵] Edit the inductor tolerance field.
- 5[↵] Enter a 5% tolerance on all inductors.
- [C][↵] Edit the capacitor tolerance field.
- 10[↵] Enter a 10% tolerance on all capacitors.

At this point, the screen should look similar to the first example, with the addition of the TOL=% tolerance statements. This file can now be saved and used with the *ICAPS* simulation system to run an extensive Monte Carlo statistical yield analyses on this filter.

Note: Monte Carlo tolerances may be inserted in netlists that use the subcircuit format, as well as non-ideal quality factors.

For more information about the Monte Carlo analysis, please refer to the *SPICE* User's Guide.

Appendices

Appendix A - Comparison of Approximations

A comparison between all the approximations supported by *QuickFil* is shown below. The following settings were used for the example:

Lowpass

Passband: up to 1 kHz

Stopband loss:

from 1.15 kHz: 55 dB

from 1.3 kHz: 40 dB

(The lower stopband loss from 1.3 kHz can only be taken into account in general amplitude approximations).

Passband loss max. 3 dB

number of transmission zeros at infinity: 2

number of variable transmission zero pairs: 2

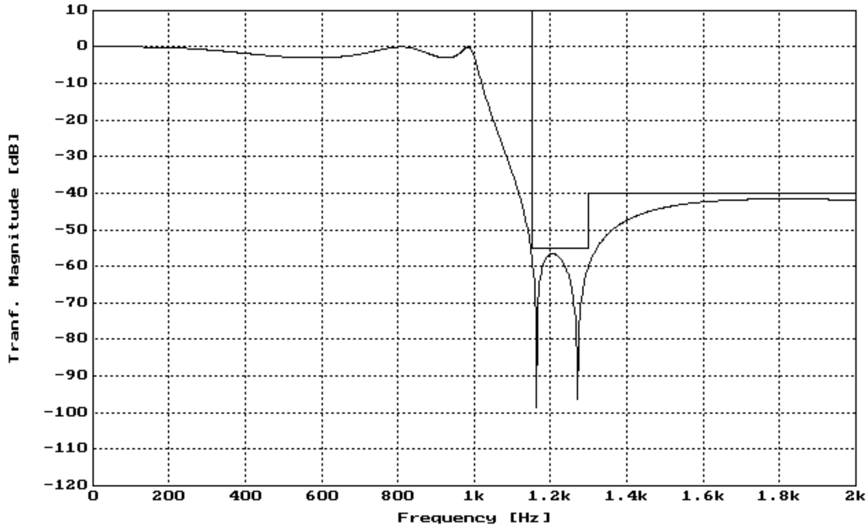
As an indicator of the amount of work required for the individual approximations, we have included the respective degree in each diagram.

General Amplitude Approximations

Equal Ripple Approximation

- Equal ripple in the passband.
- Loss curve in the stopband can be selected with either monotonic rise or optimized for the same minimum distance to the tolerance scheme defined by the user.
- Optional: setting of fixed blocking places.

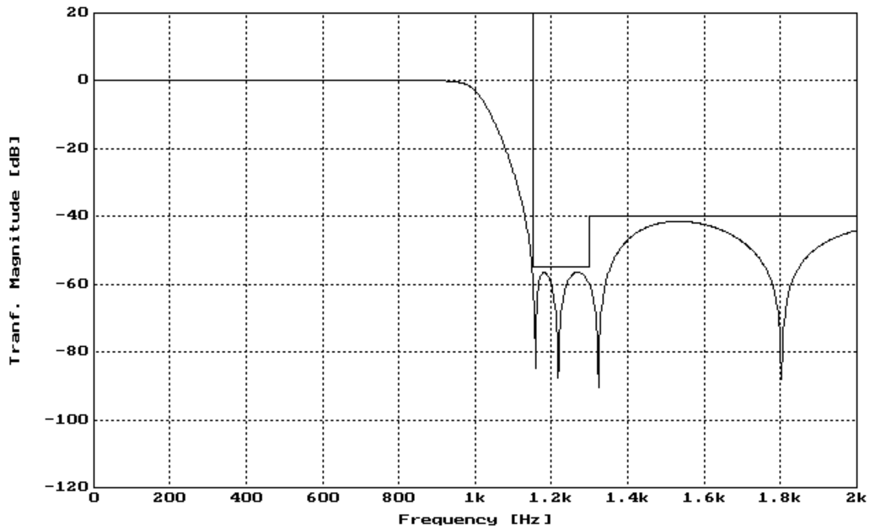
Example: Lowpass, Equal ripple approximation



Maximally Flat Approximation

- Maximally flat loss curve in the passband.
- Loss curve can be individually set in the stopband as in the equal ripple case.

Example: Lowpass, Maximally flat approximation (degree: 12)



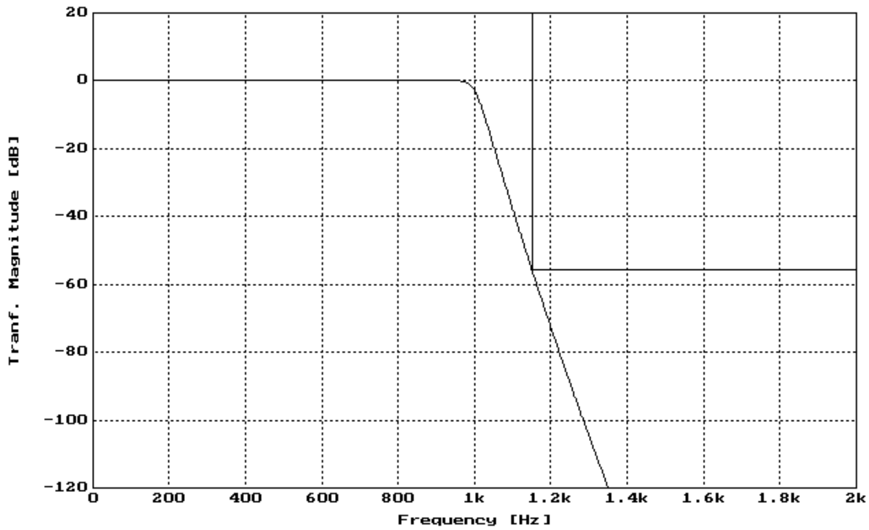
Standard Approximations

Listed below are special cases of the general amplitude approximations described above. The most essential difference, as compared to the general approximation, is the fact that in standard approximations only one specific stopband loss can be set for the whole stopband.

Butterworth Approximation

- Maximally flat loss curve in the passband.
- Monotonic rise of the loss in the stopband (special maximally flat approximation).

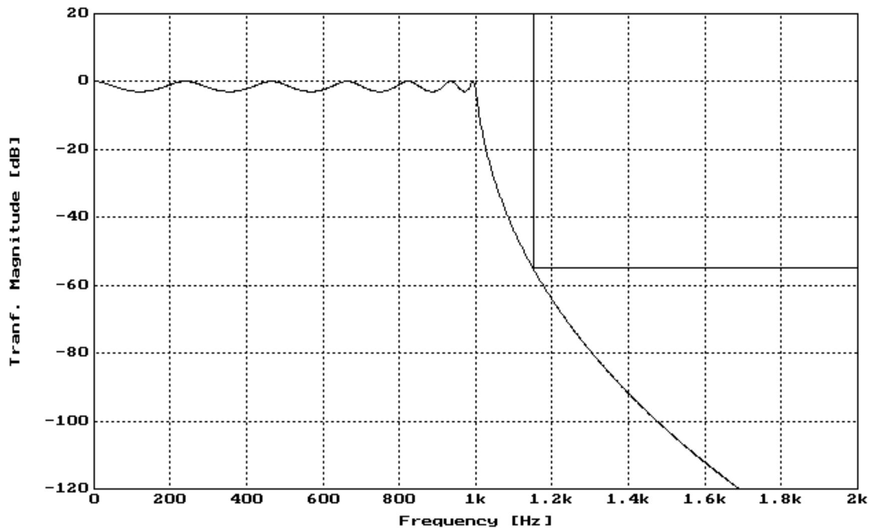
Example: Lowpass, Butterworth approximation (46th degree)



Chebyshev Approximation

- Equal ripple in the passband.
- Monotonic rise of the loss in the stopband (special equal ripple approximation).

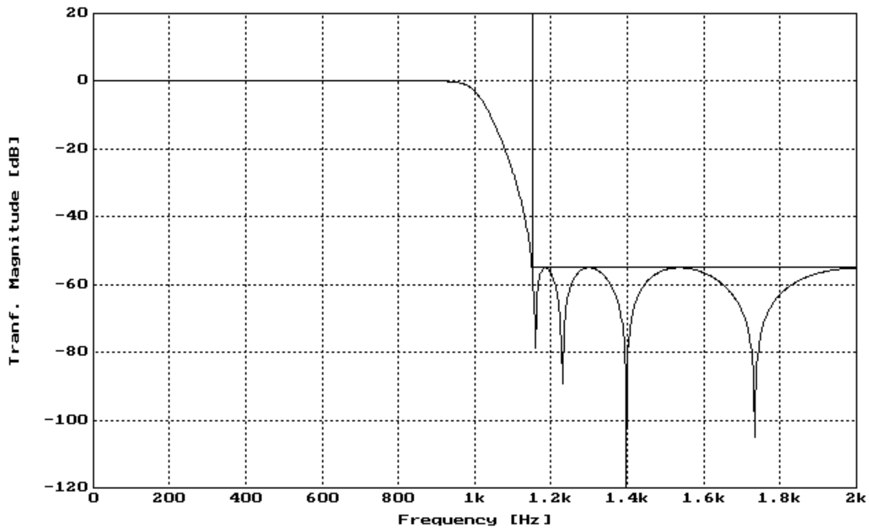
Example: Lowpass, Chebyshev approximation (13th degree):



Inverse Chebyshev Approximation

- Maximally flat loss curve in the passband.
- Loss “arcs” with the same minimum in the stopband (special maximally flat approximation).

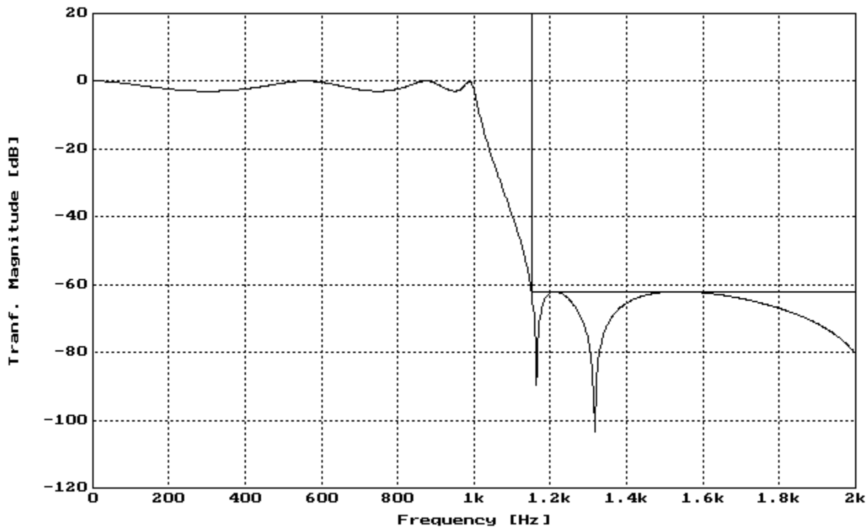
Example: Inverse Chebyshev approximation (13th degree):



Elliptic Approximation (Cauer filter)

- Equal ripple in the passband.
- Loss “arcs” with the same minimum in the stopband (special equal ripple approximation).

Example: Lowpass, Elliptic (7th degree):

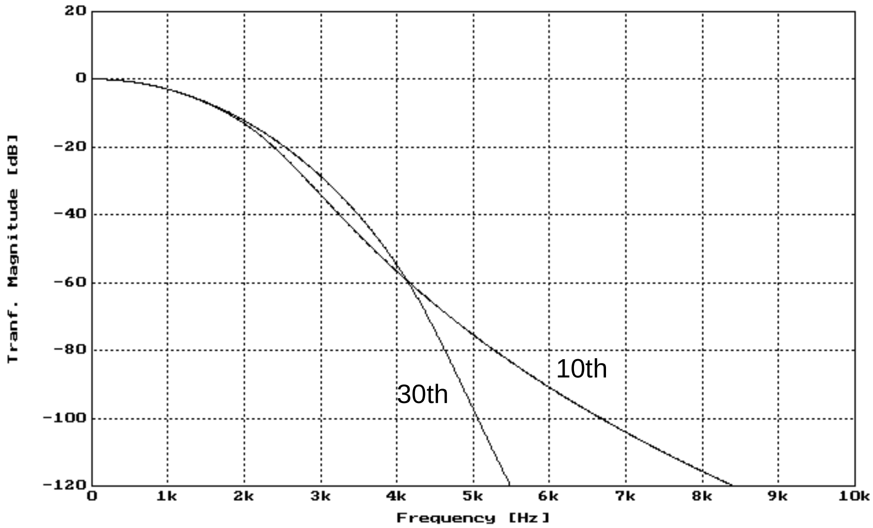


Bessel (Thomson) Approximation

This approximation (only for lowpass filters) is optimized to constant group delay (linear phase). In contrast to the other approximations described above, it is not possible to always achieve a desired steepness in the amplitude-frequency characteristic with the Bessel approximation.

The loss curve approaches a parabola with increasing order (see [1] “Literature” section).

Example Bessel 10th and 30th degree:



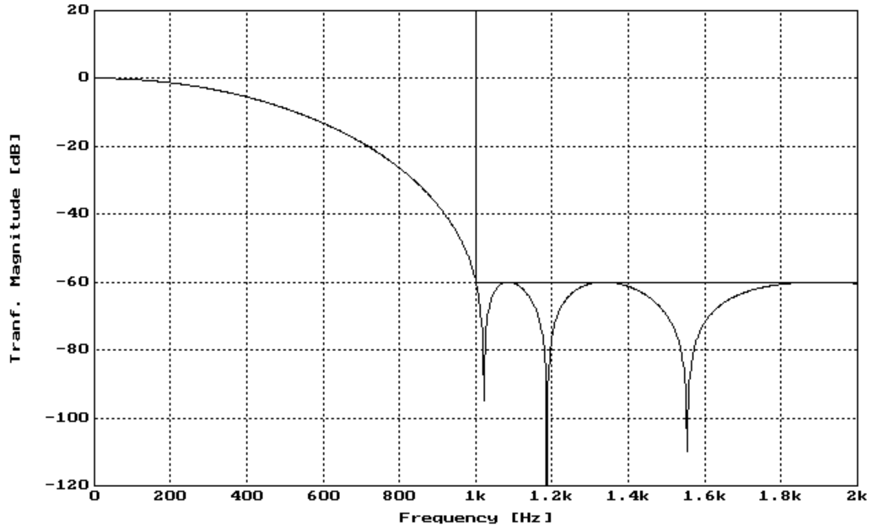
Generally, it is possible to transform a Bessel lowpass to a highpass, bandpass or bandstop filter using the usual transformations, but these new filter types do not have a linear phase response. Therefore, *QuickFil* does not allow these derived filters from the Bessel lowpass filter.

Modified Bessel Approximation

This approximation (only for lowpass filters) is optimized to constant group delay (linear phase). The difference to the Bessel lowpass filter is that the it has finite transmission zeros for increasing the stopband attenuation which have no effect to the group delay response.

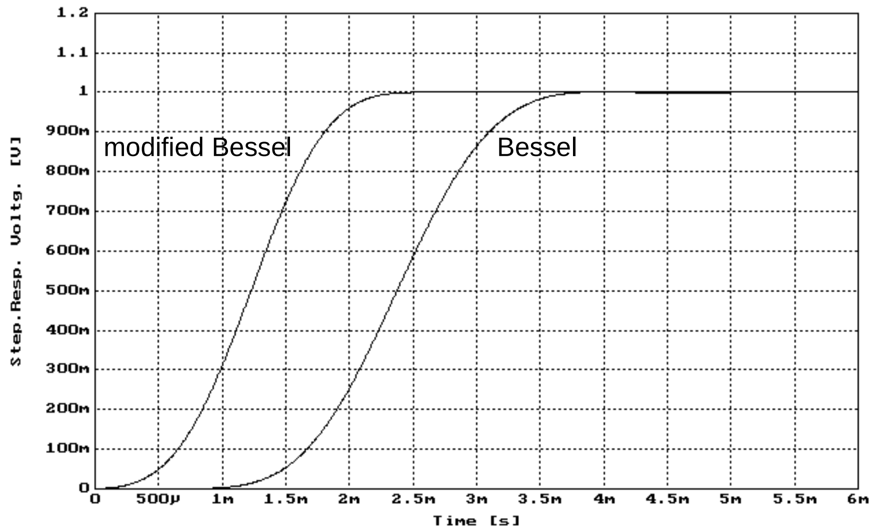
Using the Modified Bessel approximation, you can get a lowpass filter which has a very good step response and a good stopband performance. This approximation is very interesting for measurement filters, which should remove some disturbing frequencies and should have a low settle time too.

Example: Lowpass, Modified Bessel(10th degree, case b)



For the same stopband performance, you will get better step response than for Bessel approximation.

example:
stopband frequency: 1kHz, stopband loss: 60dB, degree: 10



The advantages of the design with the general amplitude approximation over that of the standard approximation are:

- More freedom in the stopband specification - you can enter a tolerance scheme with different loss values, set fixed transmission zeros (blocking places), select the number of transmission zeros at extreme frequencies, etc.
- Save on costs for the components by exact approximation to the required loss curve.
- Calculation of unsymmetrical bandpass filters.
- More freedom with regard to the ratio of the input/output resistance in lowpass, highpass and bandstop filters.

Appendix B - Case, Terminating Resistance Ratio

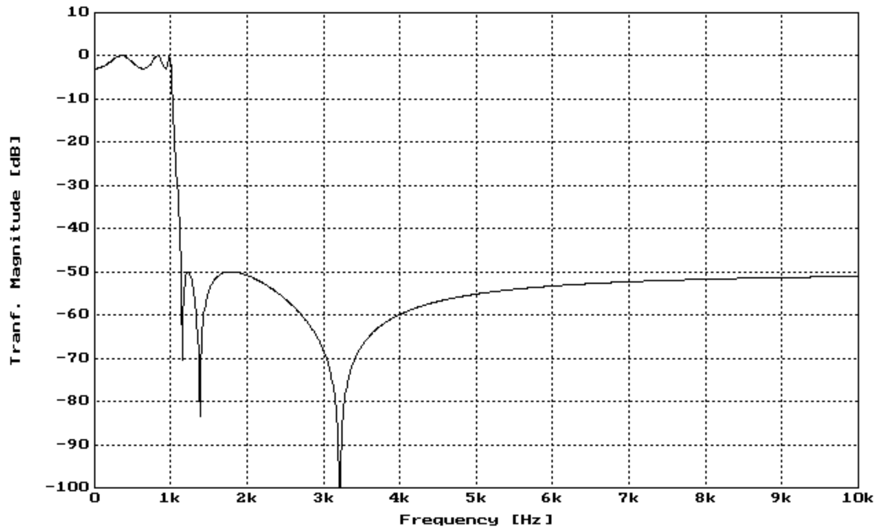
Standard Approximations

Case a

In filters with transmission zeros in the finite stopband, (Elliptic and Inverse Chebyshev filters) there exists an optimal approximation equal to the optimal use of the tolerance scheme in the passband with a maximally achievable minimum stopband loss. In commonly available literature, as well as in q,

this is termed “case a”.

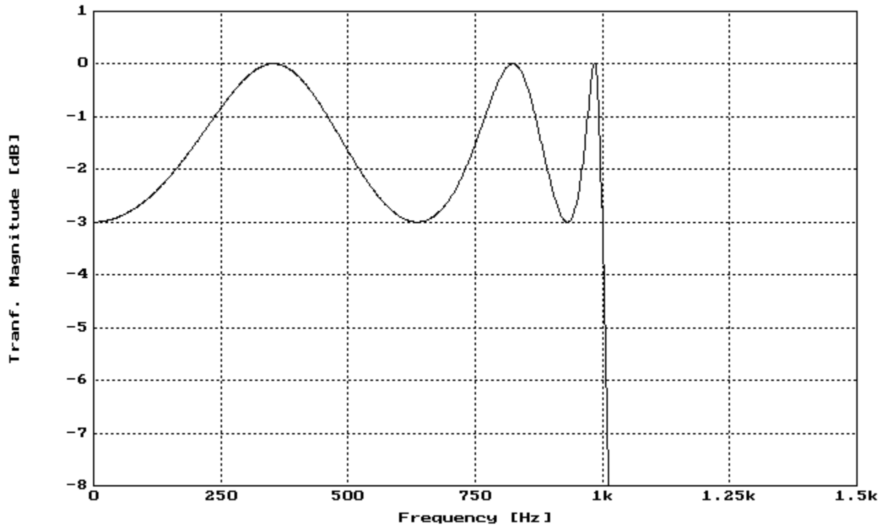
Example: Elliptic (Cauer) lowpass in case a



Note: Since the loss has a finite value when the frequency is infinite, this case can only be realized with active filters or with mutual inductances in passive filters, neither of which are supported by *QuickFil*.

With Case “a” type Elliptic lowpass filters we find that due to the position of the zeros in the passband, there is a loss at zero frequency equal to the value of the maximally allowed passband loss. This leads to different values of the terminating resistances (resistance ratio not equal 1) which does not disturb an active realization.

Example: Passband of a Cauer lowpass in case a.



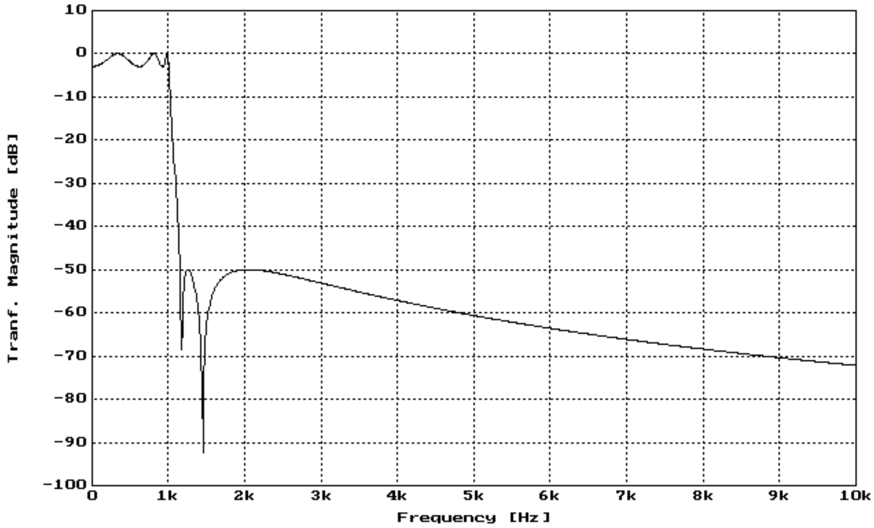
Case b

It is possible, by way of a transformation, to move the highest transmission zero position towards infinity in the approximations described previously, which allows the filter to be realized passively.

However, this leads to a slight "deterioration" in the achievable stopband loss when compared with case "a". This case is named "case b".

Note: The "initial loss" remains in zero frequency approximations with ripples in the passband. This results in differing terminating resistances for case b passive filters. If the resistors must be equivalent, then you will have to select case c.

Example: Cauer lowpass in case b:

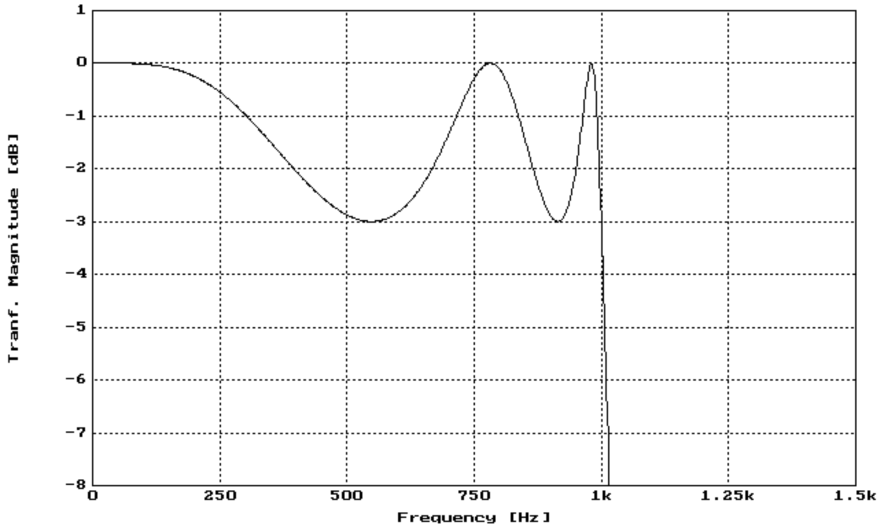


Case c

In filters with ripples in the passband (Chebyshev and Cauer), it is possible to displace, by way of a transformation, the first zero position in the passband to zero frequency. This causes the ratio of the input and the output resistances to become 1 (the two terminating resistances have the same value).

However, this leads to another slight deterioration in the filter data when compared to the filter in case b. This case is referred to as “case c”.

Example: Cauer lowpass in case c:



Summary:

Approximation	Possible Cases	Remarks
Butterworth	none	monotonic passband monotonic stopband
Chebyshev	b or c	equal ripple in the passband and monotonic stopband
Inv.-Chebyshev	a or b	monotonic passband and equal minima in the stopband
Elliptic (Cauer)	a, b or c	equal ripple in the passband and equal minima in the stopband
Bessel	none	monotonic passband monotonic stopband
Modified Bessel	a or b	monotonic passband and equal minima stopband

Note: The case studies only have an influence on the result of the calculation in lowpass and highpass filters of even degree, and in bandpass and bandstop filters whose degree can be divided by four.

Equal Ripple Approximation

As mentioned before, the position of the first zero position in the passband of a lowpass can be displaced by a transformation, which changes the “initial loss” at zero frequency.

This enables two extremes in the standard Chebyshev and Elliptic (Cauer) approximations:

Case b: Loss at zero frequency = maximum allowed passband loss

Case c: Loss at zero frequency = zero

The ratio between the input and output resistance is determined by the initial loss at zero frequency. For the standard approximation, this can either be a specific value (case b) or 1 (case c).

In the equal ripple approximation, the user has the ability to select the resistance ratio. *QuickFil* will then “displace” the respective zero position to the best position in the passband using a special transformation.

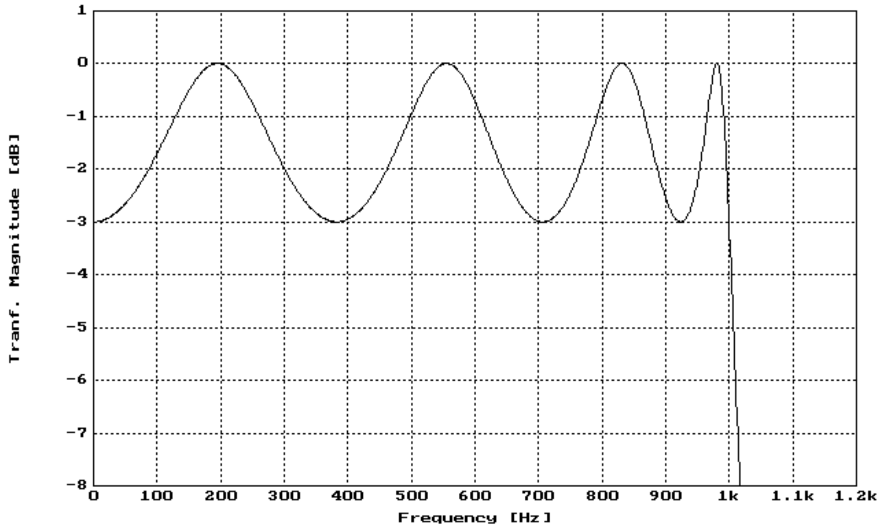
This can be performed in lowpass filters of an even degree, highpass filters of an even degree, and bandstop filters with a degree that can be divided by four.

Note: In lowpass and highpass filters with an odd degree and bandstop filters whose degree cannot be divided by four (because bandstop filters are transformed to lowpass filters with half degree), the resistance ratio is generally 1, i.e. the terminating resistances are equivalent.

It is not necessary to enter the resistance ratio in bandpass filters because it is possible to influence the values of the terminating resistances by using Norton’s transformation! The entry is made in the SPECIFICATION menu. The following options are available:

Loss at zero/infinity = passband bandedge loss

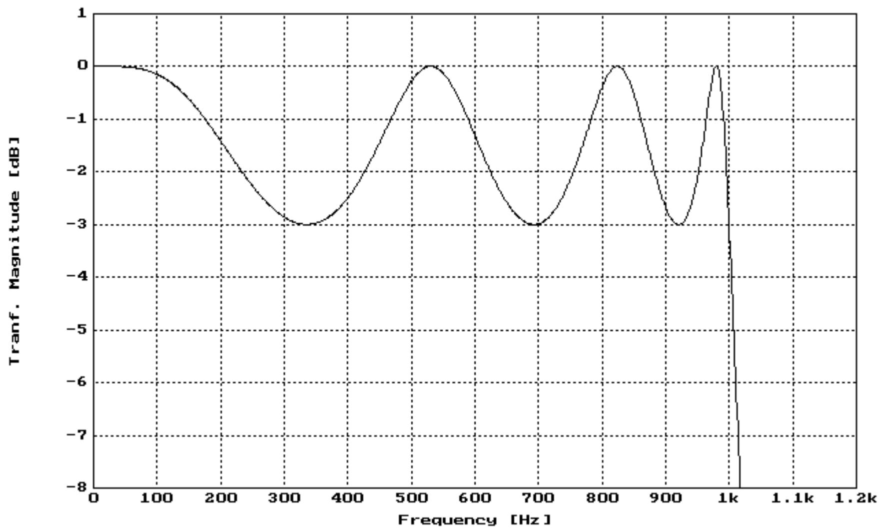
Results in the optimal equal ripple approximation by optimizing use of the tolerance scheme in the passband and optimal stopband loss at the given degree.



This is equivalent to standard approximations using cases a and b.

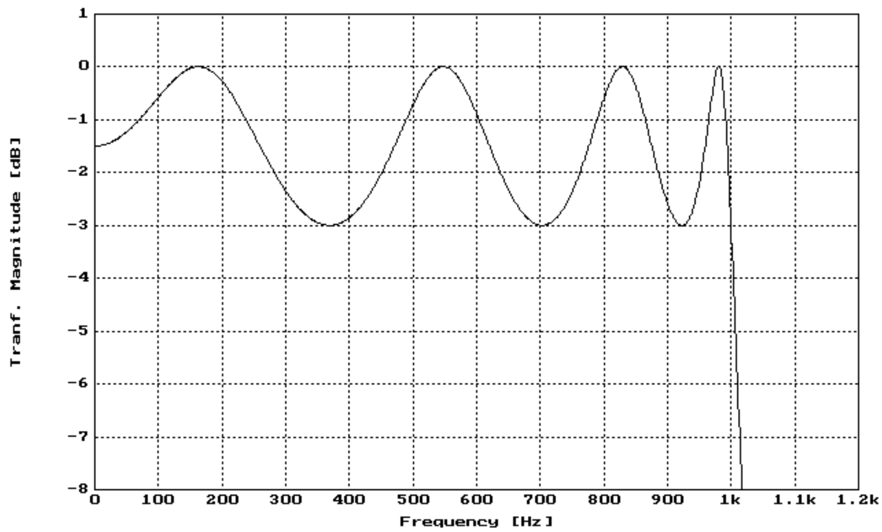
Note: We recommend accepting the proposal for the resistance ratio if you intend to realize the filter actively, because the resistance ratios of the terminations are not of importance for active realizations.

Loss at zero/infinity = 0

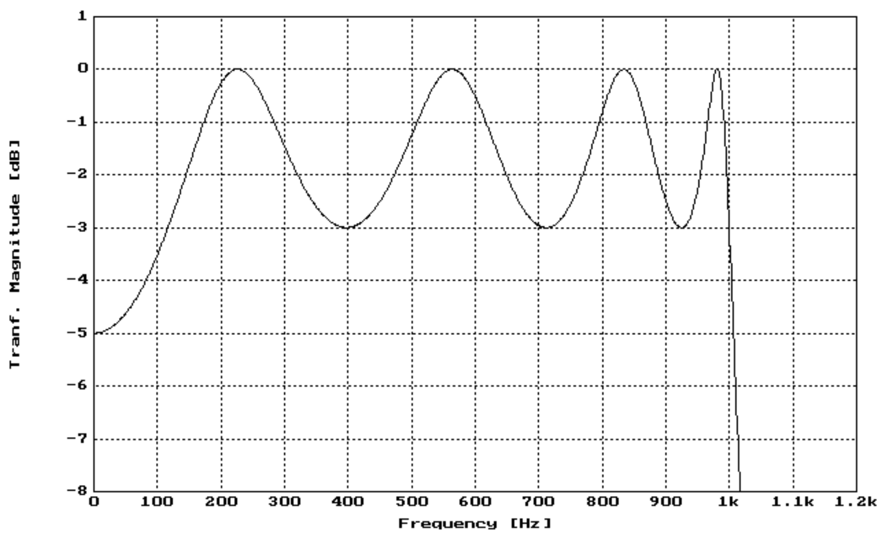


This is equivalent to case c in the standard approximations.

Loss at zero/infinity is between zero and passband bandedge loss



Loss at zero/infinity is higher than the passband bandedge loss



Note: The filter no longer fulfills the requirements! (This option is, however,provided for special cases).

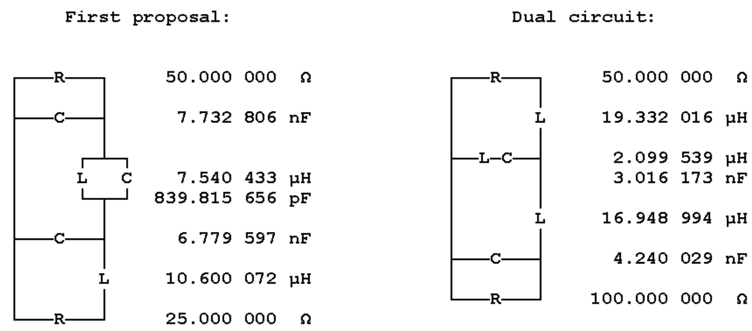
Definition of “resistance ratio”

The resistance ratio in the SPECIFICATION menu is the ratio between input and output resistance. The higher of the two resistance values is understood to be the numerator of this ratio. Therefore, the resistance ratio can be selected either higher than, or equal to 1.

The value set in the TERMINATION menu refers to the input.

If the calculated output termination is not in a direct inverse ratio to this value, the output termination can be changed to the desired value by selecting the dual circuit.

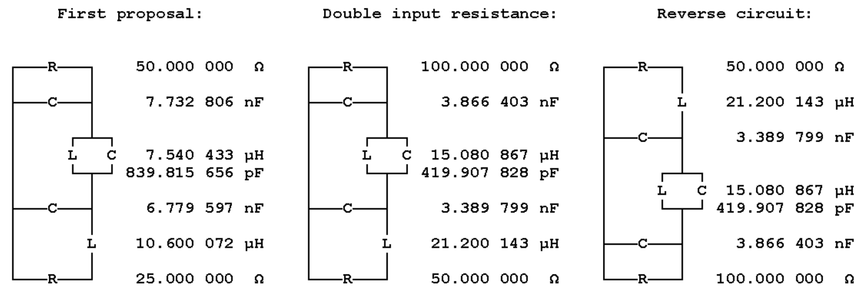
Example (resistance ratio = 2):



Alternatively:

- Set the terminating resistance in the TERMINATION menu to the value which is desired at the output of the circuit.
- Then, reverse the circuit in the MANIPULATION AND ANALYSIS menu.

For example:



Limitations

There are cases where the frequency transformation is not possible for certain terminating resistances. This can happen in lowpass filters: if there is no transmission zero at infinity and all the transmission zeros are located near the passband edge.

Highpass filters: if there is no transmission zero at zero and all the transmission zeros are located near the passband edge, and bandstops filters: if there are no transmission zeros near the center frequency and all the transmission zeros are located near the passband edge.

In such cases *QuickFil* will use the proposed resistance. This is indicated by a message. The resistance ratio is purely theoretical in such cases, because these approximations cannot be realized passively and the resistance ratio is unimportant for the active filter.

Appendix C - Conventional/Parametric Bandpass Filters

For bandpass filters of the general, equal ripple, and maximally flat approximations, *QuickFil* allows you to select between two different types of circuits:

- conventional
- parametric

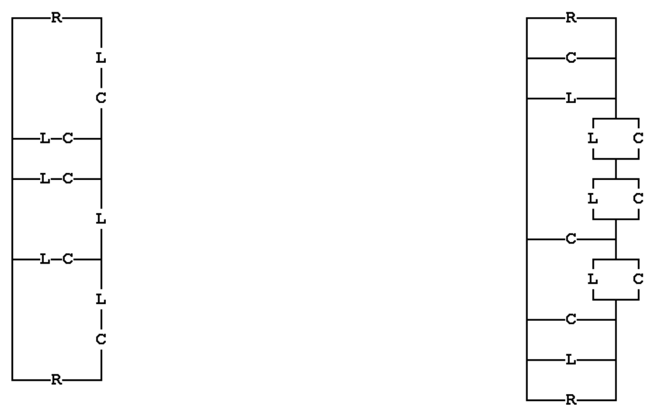
Conventional Bandpass Filters

This type of bandpass was the only one used until about thirty years ago (hence the name "conventional"). Conventional bandpass filters can only be developed with an even degree. (*QuickFil* automatically corrects the number of the transmission zeros at zero and infinity in such a way as to automatically achieve a filter of an even degree).

A significant feature of this type of filter is that its input impedance at zero and infinite frequency always has the same extreme value (zero or infinite). The same applies to the output impedance.

Examples of a conventional bandpass filter (8th degree):

Computer circuit: The pertinent dual circuit:



Impedance (input, output) at extreme frequencies (zero, infinity):

Infinite Zero

Parametric Bandpass Filters

The desire to obtain bandpass filters with fewer inductances led to the development of the parametric type of bandpass. Parametric filters can be designed with an even or odd degree.

However, it is necessary to have at least one transmission zero at zero and infinity.

Note: For passive filters, only parametric bandpass filters of an even degree are interesting.

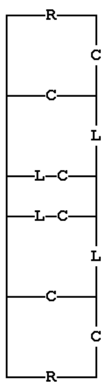
In even degree parametric filters, the value of the input or output impedance at infinite frequency assumes the opposite value (zero or infinite) of the impedance at zero frequency.

In odd degree parametric filters, only the input or output impedance assumes this behavior. If the output impedance assumes the behavior, then there is a negative sign in the parameter field.

The term "parametric" is used because this type of bandpass filter has an additional free parameter that is equal to the zero position of the polynomial f(s). Through this parameter (actually labeled "parameter" in the *QuickFil* SPECIFICATIONS menu) you can influence the values of the components during the design.

Examples of parametric bandpass filters:

Bandpass filter of even (8th) degree:



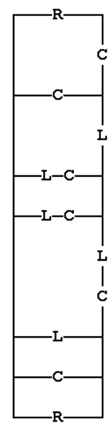
Input and output impedance at zero frequency: infinite

Input and output impedance at infinite frequency: zero

Note: The dual circuit is not of interest because it would yield a maximum number of inductors.

Bandpass filter of odd (9th) degree:

Positive parameter value:

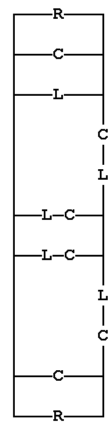


Input impedance at zero frequency: infinite

Input impedance at infinite frequency: zero

Output impedance: zero for each

Negative parameter value:



Output impedance at zero frequency: infinite

Parameter refers to the value of the Parameters field in the SPECIFICATIONS menu.

Output impedance at infinite frequency: zero

Input impedance: zero for each

Advantages of parametric bandpass filters over the conventional filters:

- Fewer inductors (when selecting the best circuit).
- Additional options for varying the values of the components through parameter variation. This option does not require additional components like the Norton's transformation.

Disadvantages:

- Less stopband loss with the same settings of a conventional filter.
- No extreme terminations possible in even degrees.

Appendix D - Design With User Defined Circuits

It is often necessary to realize a filter with a given circuit structure. In order to enter this circuit through the program section “Entry of circuits”, the number of transmission zeros/ pairs of transmission zeros must already be known when the filter is specified. These can be determined in the following way:

Lowpass, Highpass, and Bandpass Filters

A passive filter circuit can be regarded as the sum of series connected two-ports. You can distinguish between six different types of two-ports:

- Series inductor (ls)
- Parallel inductor (lp)
- Series capacitance (cs)
- Parallel capacitance (cp)
- Series-resonance circuit that is connected in parallel (se)
- Parallel-resonance circuit that is connected in series (pa)

By mentally “manipulating the circuit” you can easily determine the desired number of transmission zeros.

Determination of the number of transmission zeros at zero

All two-ports except cs and lp are left out of the circuit. The number of changes between cs and lp results in the number of transmission zeros at zero with the first element counting as a change.

Determination of the number of transmission zeros at infinity

All two-ports except cp and ls are left out of the circuit. The number of changes between cp and ls results in the number of transmission zeros at infinity with the first element counting as a change.

Number of the pairs of finite transmission zeros

This number is equivalent to the number of all pa and se two ports. In the equal ripple and maximally flat approximations, it can be set as the total of the number of fixed and variable pairs of transmission zeros.

Difference between conventional/parametric bandpass filters:

In bandpass filters you have the additional problem of determining whether the given circuit structure is a conventional or parametric filter.

Some hints on distinguishing between the two:

- If the degree is odd, the circuit represents a parametric filter (It is only possible to produce conventional bandpass filters with an even degree).
- If the proposed bandpass has an even degree, we recommend that you check the input impedance’s behavior:

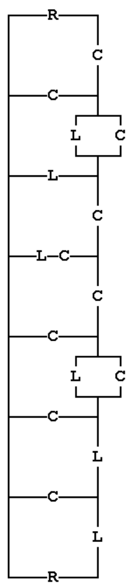
Conventional bandpass filters: The input impedance has the same value (zero or infinite) at both zero and infinite frequency.

Parametric bandpass filters: The value of the input and/ or output impedance has the opposite extreme value (zero or infinite) at infinite frequency than at zero frequency.

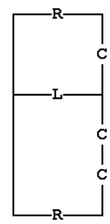
Important note concerning parametric filters:

If the proposed filter has the behavior described above at only the output, you can either enter a negative parameter for the design, or enter the circuit in reverse order and reverse it at the end of the dimensioning.

Example:
Given circuit structure:

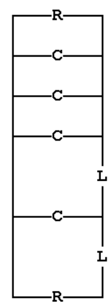


Number of transmission zeros at zero:



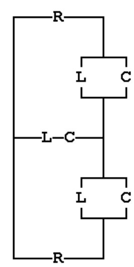
Result: 3 transmission zeros at 0.

Number of transmission zeros at infinity:



Result: 4 transmission zeros at infinity

Number of finite transmission zeros:



Summary:

Type of filter: parametric bandpass, positive parameter

Filter degree: 3 + 4 + (2 x 3) = 13

Bandstop Filters

Bandstop filters are slightly more complicated. The best way to proceed is to transform the symmetrical bandstop circuit into a lowpass circuit. The following “transformation” allows you to determine from this circuit, the bandstop’s number of pairs of transmission zeros.

Number of pairs of transmission zeros at the center frequency

This is equal to the number of transmission zeros at infinity in the lowpass.

Number of finite pairs of transmission zeros in the bandstop

This is equal to twice the number of finite pairs of transmission zeros in the lowpass.

Appendix E - Calculation Speed

One outstanding feature of *QuickFil* is its calculation speed. A computing algorithm, written in assembly code, with a very high internal accuracy which allows you to design filters up to the 50th order. *QuickFil* avoids iterative procedures where possible, preferring to use analytically-derived formula.

Appendix F - Calculating Component Values

The procedure below for calculating component values has proved itself for loss-free reactance filters (see [1]):

1. Choose a circuit design capable of creating the desired transmission zeros (depending on the transfer function).
2. The input reactance, with no load or a short across the filter output, is calculated from the transmission function H(s) and characteristic function K(s) (for definitions, see the following pages).
3. The first component value is calculated based on the reactance function and then the reactance of the remainder of the circuit is calculated. In resonance circuits, the transmission zero value is used for the resonant frequency. This step is repeated until all the component values are known.
4. Reverse the circuit (mirror-image) and calculate the component values in the same way.
5. The two theoretical circuits are compared and the output terminating resistance calculated. This comparison may give rise to numerical problems. With single-side terminated filters, only one circuit is designed. Therefore, these numerical problems cannot arise.

Determination of Variable Transmission Zero Pairs

This is how *QuickFil* finds the proposal for the number of variable transmission zeros:

Assessment in lowpass filters

A mean stopband loss is determined from the entered tolerance scheme. This is equivalent to the mean value of the stopband loss when displayed in the gamma plane.

$$\gamma = \frac{1}{2}\ln\left(1 - \left(\frac{f_g}{f}\right)^2\right)$$

Then, the degree of a reference filter that has the same stopband edge frequency and uses the previously calculated mean stopband loss as the stopband loss, is determined.

Type of reference filter:

- For equal ripple lowpass filters: Elliptic lowpass.
- For maximally flat lowpass filters: Inverse Chebyshev lowpass.

Half of the degree number of this reference filter is then entered in the program as the proposal for the number of pairs of transmission zeros.

Assessment in highpass filters

The tolerance scheme is first transformed to that of a lowpass and, as described above, the number of variable transmission zeros is determined.

Assessment in bandpass filters

Here, the number of transmission zeros in the upper and lower stopband is determined separately. First, the lower tolerance scheme is transformed to a tolerance scheme of the lowpass and, as already described for the lowpass, the number of variable transmission zeros is determined. It is presumed that the bandpass filter is symmetrical. Hence, the same procedure is used to determine the number of transmission zeros in the upper stopband.

Assessment in bandstop filters

The tolerance scheme is transformed to a tolerance scheme of the lowpass. However, due to ambiguities in the lowpass bandstop transformation, this generally leads to two lowpass tolerance schemes. From these two schemes a single tolerance scheme is formed which is equivalent to the respective maximum of each of the two tolerance schemes. As was described for the lowpass, the number of variable transmission zeros is determined. The double values of these transmission zeros are used as the proposal for the bandstop.

Initialization of the variable transmission zero pairs

This is how *QuickFil* finds the best initial values of the transmission zero for the subsequent optimization. In lowpass, highpass, and bandstop filters, the tolerance scheme is transformed to a lowpass. The variable transmission zeros must be located in the stopband which results in the range [gammas, 0] in the gamma plane. The variable transmission zeros are then favorably distributed in this range in accordance with certain rules.

In bandpass filters, the transmission zeros in the upper and lower stopband are initialized separately. In the gamma plane, the upper stopband is equivalent to the range [gammas1, 0] and the lower stopband to the range [ln(1/a), gammas2].

The variable transmission zeros are initialized after entering their number or the tolerance scheme. The transmission zero frequencies are only used as the initial values for the following optimization. The arrangement of the transmission zero frequencies takes place in such a way that, referring to the transformed frequency variable gamma γ, they are evenly distributed in the stopband.

for lowpass:

$$\gamma = \frac{1}{2}\ln\left(1-\left(\frac{f_g}{f}\right)^2\right)$$

fg ... passband edge frequency

for bandpass:

$$\gamma = \frac{1}{2}\ln\left(\frac{f^2-f_{gl}^{\ 2}}{f^2-f_{gu}^2}\right)$$

fgl ... lower passband edge frequency
fgu... upper passband edge frequency

Appendix G - Output Roots

In the menu OUTPUT ROOTS: the characteristics of the passive filter are specified. For passive filters, the scatter parameters or transmission parameters S_{11}, S_{12}, S_{21}, and S_{22} are used for the description of the performance.

Here is a short list of expressions used in the theory of passive filters:

s	is the complex radian frequency.
n	is the degree of the filter, degree of the polynomial e(s).
n_p	is the number of finite transmission zeros.
n_f	is the number of finite reflection zeros.
S_{ek}	are the normalized zeros of the polynomial e(s), poles of G(s), called natural modes.
S_{pk}	are the normalized zeros of the polynomial p(s), zeros of G(s), called transmission zeros.
S_{fk}	are the normalized zeros of the polynomial f(s), zeros of ρ(s), called refelection zeros.
ω_o	is the reference radian frequency, used for normalization.

Transfer function:

$$G(s) = \frac{1}{H(s)} = S_{21}(s) = S_{12}(s) = \frac{p(s)}{e(s)}$$

$$G(s) = \frac{1}{C_{ep}} \frac{\prod_{k=1}^{n_p}\left(\frac{s}{\omega_0} - s_{pk}\right)}{\prod_{k=1}^{n}\left(\frac{s}{\omega_0} - s_{ek}\right)}$$

Note: The terms for the "transfer function" in the book “Modern Filter Theory and Design” [1] is different. In filter theory, normally the expression

H(s) is called "transfer function". In most other technical areas, the reciprocal value G(s) is called "transfer function".

Characteristic function:

$$K(s) = \frac{S_{11}(s)}{S_{21}(s)} = \frac{f(s)}{p(s)}$$

$$K(s) = C_{fp} \frac{\prod_{k=1}^{n_f} \left(\frac{s}{\omega_0} - S_{fk} \right)}{\prod_{k=1}^{n_p} \left(\frac{s}{\omega_0} - S_{pk} \right)}$$

Reflection Factor:

$$\rho(s) = S_{11}(s) = \frac{f(s)}{e(s)} = \frac{K(s)}{H(s)}$$

$$\rho(s) = C_{fe} \frac{\prod_{k=1}^{n_f} \left(\frac{s}{\omega_0} - S_{fk} \right)}{\prod_{k=1}^{n_p} \left(\frac{s}{\omega_0} - S_{pk} \right)}$$

The relationship between the constants is given by the equation:

$$C_{fp} = C_{ep} \cdot C_{fe}$$

The following relationships is one of the most important equations in filter theory. It is called "Feldtkeller equation" [1].

$$H(s) \cdot H(-s) = \frac{1}{G(s) \cdot G(-s)} = 1 + K(s) \cdot K(-s)$$

For the polynomials e(s), p(s), f(s) you will get:

$$e(s) \cdot e(-s) = p(s) \cdot p(-s) + f(s) \cdot f(-s)$$

Due to this equation you will get following inequalities:

$$|\rho(s)|\big|_{s = j\omega} \leq 1$$

$$|G(s)|\big|_{s = j\omega} \leq 1$$

Appendix H - Dual Circuits

In *QuickFil*, you can create alternative circuits using the dual transformation. The reference resistance is used as the duality constant.

The formula is:

$$Z_A \cdot Z_B = R_B^{\ 2}$$

where,

Z_A is the impedance of original circuit.

Z_B is the impedance of dual circuit.

R_B is the reference resistance. This is based
with single-termination: on the input resistance
with double-termination: on the resistance given.

If you design a dual circuit from the MANIPULATION AND ANALYSIS menu, you can select your own reference resistance.

If you design a dual circuit from the PASSIVE DESIGN menu, the program will use the specified terminating impedance.

Dualities:

$$\frac{L_1}{C_2} = R_B^{\ 2}$$

$$\frac{L_2}{C_1} = R_B^{\ 2}$$

$$\frac{L_1}{C_2} = R_B^{\ 2} \qquad \frac{L_2}{C_1} = R_B^{\ 2}$$

Appendix I - Norton's Transformation

Norton´s transformation enables you to transform certain component combinations within a circuit so that all subsequent components are multiplied (L´s and R´s) or divided (C´s) by a given factor. This does not affect the transmission characteristics. The circuit before and after Norton's transformation will have the same S-parameters.

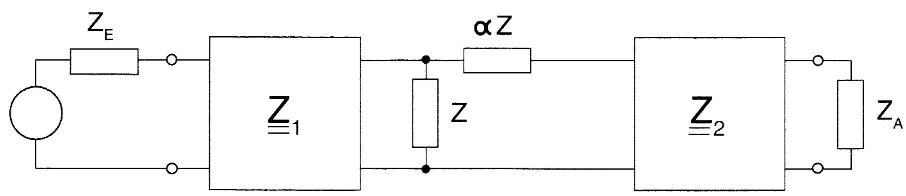
In *QuickFil*, it can therefore be used to:

- Achieve a desired termination.
- Change problematic component values.
- Achieve equal component values.

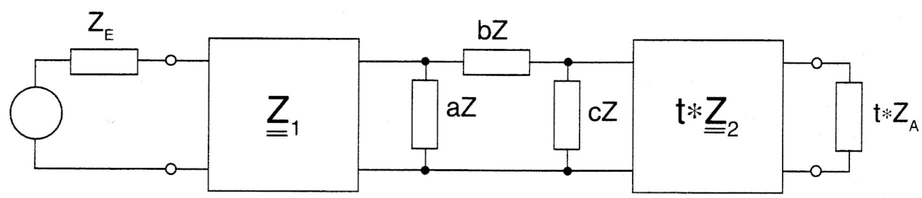
To perform a Norton's transformation of a filter, you need a special structure of the circuit. You need the same kind of elements in parallel and series. The elements can be capacitors, inductors, parallel or serial resonant circuits. This feature is only possible for bandpass filters.

Principle

before transformation:



after transformation:



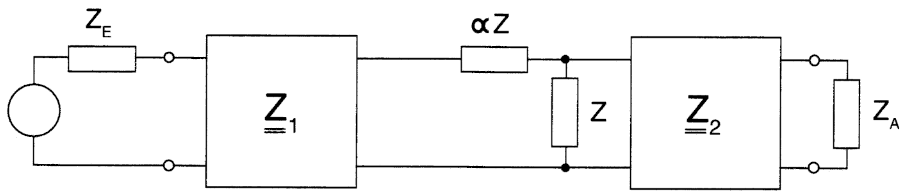
t is the transformation factor and a, b and c are dependent on t.

Z1 is the impedance matrix of the two-port 1

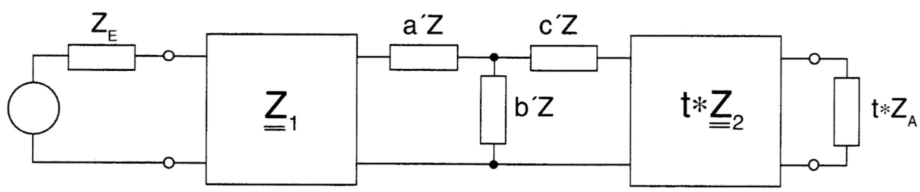
Z2 is the impedance matrix of the two-port 2

The second possibility is shown below:

before transformation:

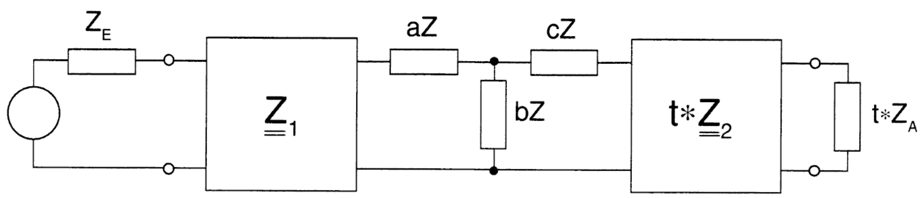


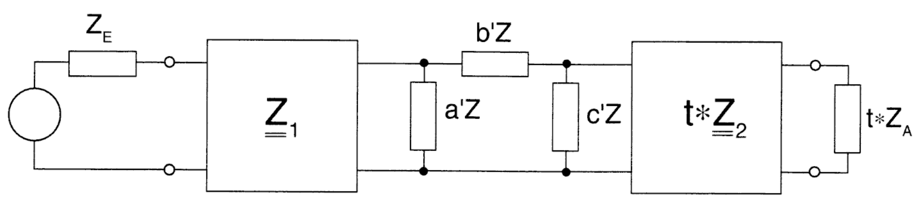
after transformation:



t is the transformation factor and a', b' and c' are dependent on t.

The result of Norton's transformation is a PI or a TEE-circuit part, which are equivalent. Using *QuickFil* you can choose between PI or TEE circuit. If you have chosen the wrong, you can convert the PI to TEE circuit in a special menu of MANIPULATION_ANALYSIS menu.

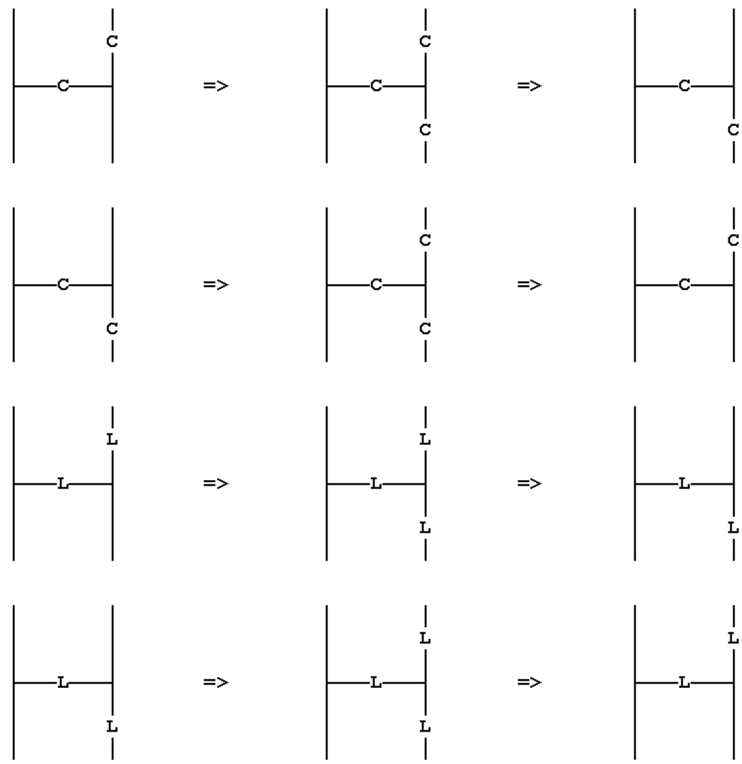




Depending on the transformation factor t, the resulting elements of the PI or TEE circuit can get zero or negative too. In the case that c or a' will get zero, we say the Norton's transformation is maximum or minimum. If the transformation factor is outside the limits of maximum and minimum, you will get negative element values which are not realizable.

Here are examples of simple Norton's transformations:

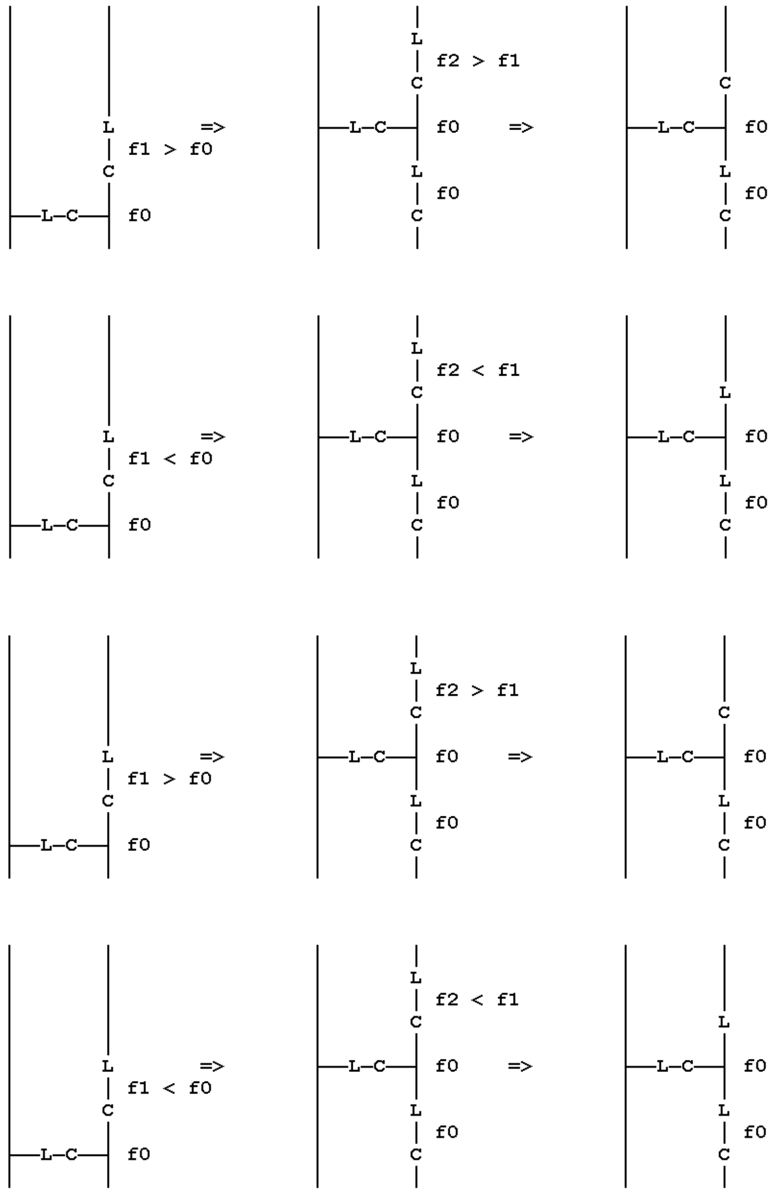
before transformation | after normal transformation | after max, min transformation



A bit more complicated, are Norton's transformations for resonant circuits. You can choose series or parallel resonant circuits and perform Norton's transformation too.

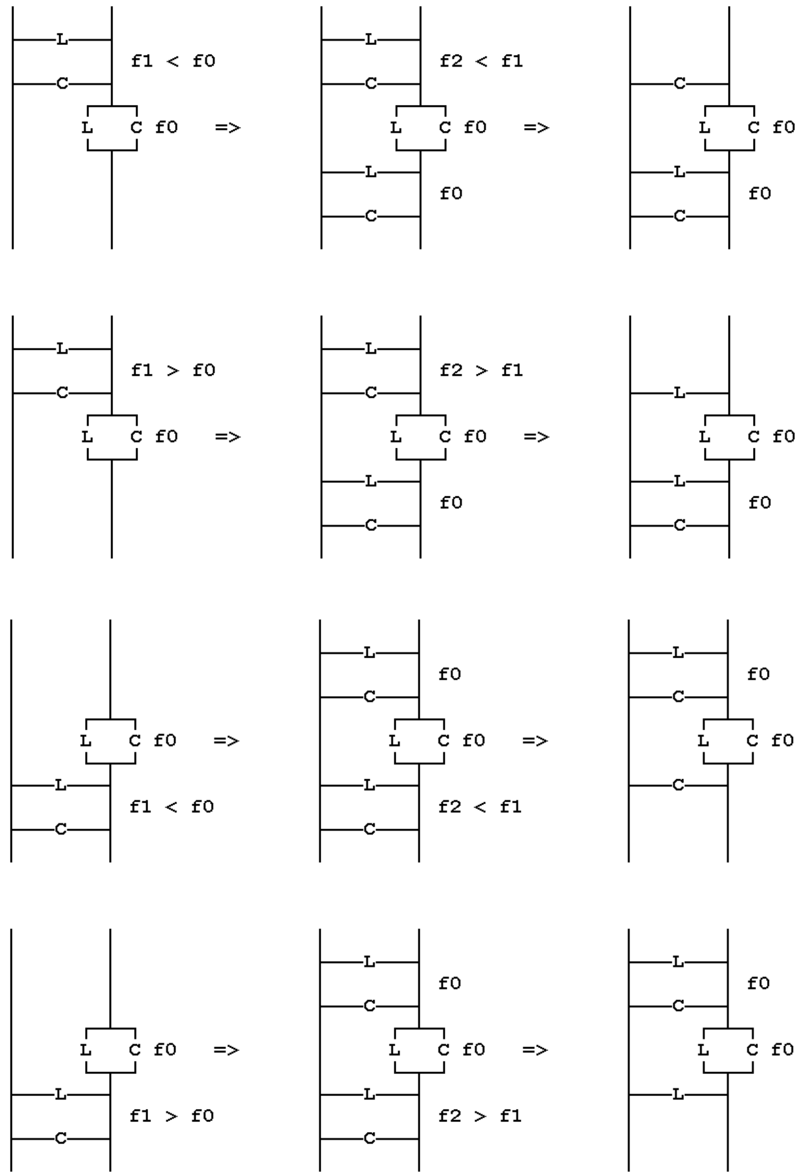
Here are some examples:

before transformation	after normal transformation	after max, min transformation



Resonance frequencies must be equal, before a transformation can be carried out. This is achieved by splitting off either a C or an L.

before transformation after normal transformation after max, min transformation

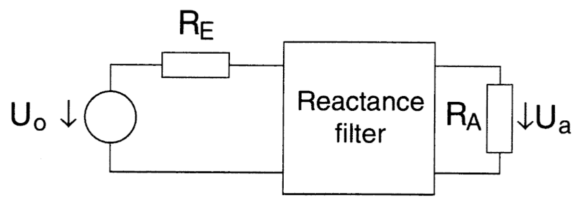


Normally, these special transformations are interesting when you have no other possibility, but normally you will prefere the simple Norton’s transformation, because you need less elements.

Appendix J - Circuit Analysis

In the circuit analysis you can investigate the performance of the filter. There are some expressions which will be explained here more precisely.

Transfer function of a doubly terminated filter:



$$G(s) = S_{21} = \frac{2 \cdot U_A}{U_0}\sqrt{\frac{R_E}{R_A}}$$

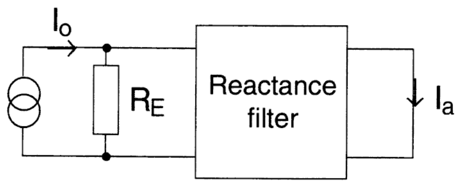
The transfer function is a measure for the power, which is transferred by the filter compared to the maximum possible power:

$$\frac{P_a}{P_{amax}} = |G(j\omega)|^2$$

Transfer function of a singly terminated filter:

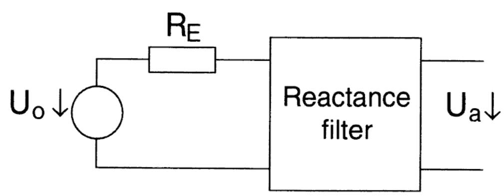
The filter has only one termination resistor, the second port of the filter is open or has a short circuit. Therefore you cannot transfer any power. Now, the transfer function for singly terminated filters is defined as the ratio of the output to input voltage for open output, or as the ratio of the output to input current for short circuit output.

- Short circuit at the output:



$$G(s) = \frac{I_A}{I_0}$$

- Open output:

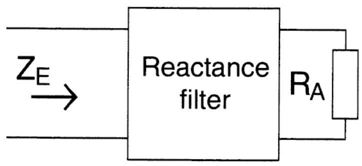


$$G(s) = \frac{U_A}{U_0}$$

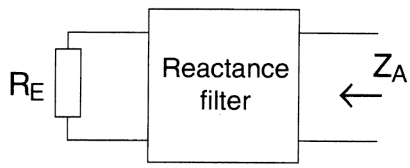
Input and Output impedance:

You can analyze the impedance at the input and output port of the filter if the opposite port is terminated by the specified termination. The termination can be a resistor, open or short circuit, as defined by the synthesis.

Input impedance:



Output impedance:



Input reflection factor:

The input reflection factor is defined as the input impedance and the input termination:

$$\rho_E(s) = \frac{Z_E(s) - R_E}{Z_E(s) + R_E}$$

Output reflection factor:

The output reflection factor is defined as the output impedance and the output termination:

$$\rho_A(s) = \frac{Z_A(s) - R_A}{Z_A(s) + R_A}$$

Appendix K - Circuits with positive elements

At the synthesis of passive ladder circuits there is a fundamental problem that not all transfer functions are realizable because there could be negative element values. Negative capacitors and negative inductors are not realizable and the calculated circuit has little practical meaning.

For the same transfer function, there is a large amount of different ladder circuits possible, and some of the circuits have only positive element values and are realizable, while other circuits are not realizable. Sometimes there is only one circuit realizable and it is a little bit difficult to find. On the other side, you will find the different circuits which are realizable, choose the best one for you.

The calculation of the component values

Here, you will find a short summary of the algorithm for calculating the element values of passive filters which many books cover.

First, the input impedance of the filter is calculated, assuming that the output of the filter is open or there is a short circuit at the output. Since all elements of the filter are reactances, there is no power loss and therefore the input impedance function is a reactance function.

Now different two-ports can be extracted from this impedance function which will reduce the degree of the reactance function. These two-ports represent sub circuits of the filter, the reduced impedance function (remainder impedance function) represents the rest of the filter. The condition for a realizable filter is that the element values of the sub circuit are positive (realizable) and the reduced impedance function is a reactance function. These conditions are checked for the different sub circuits and the realization process is performed recursively until the remainder impedance function vanishes.

For the manual search you can choose the sub circuit for realizing the filter partly. *QuickFil* will calculate the impedance function of the rest of the filter and will offer the possible sub circuits.

In the case that there is no sub circuit possible, which has positive elements only, the only solution is to remove the last sub circuit. By doing this, you can choose another sub circuit and perhaps you will find a solution.

The whole algorithm can be represented in a tree which has branches and each branch has branches, etc. Some of the branches are broken (no sub circuit with positive elements) and all branches of that branch aren't available. Now, you must search the whole tree for a path to the last branch and if you have found such a path, you will get a realizable ladder circuit.

There is an automatic search available (default) which will search circuits with positive elements systematically.

You can combine manual search and automatic search so you can improve the probability to find a solution very fast. Since the searching time depends on the degree of the filter in an exponential context, this feature is an advantage for high order filters. On the other hand, using the search algorithm, you can find all possible solutions. If the automatic algorithm states that there is no filter structure available, you can say, that this transfer function is not realizable by ladder circuits.

A singly terminated filter which has the termination at the output is reversed for the search of filters with positive elements otherwise, there is no impedance available.

Since the dual circuit is a further possible circuit for the same transfer function, but different input impedance, you can choose one of the two possible circuits.

Appendix L - Group delay correction

Method of the Approximation

For the approximation of the group delay the least p-th / Taylor optimization is used. First, the target response for the group delay is defined. Further, a transfer function for the allpass filter is specified using the transmission zeros as parameters. An error function is defined, which is the difference of the target response to the actual response of the allpass filter. For this error function, the least p-th norm is used to define a criterion for the optimum.

$$\|f(x)\|_p = \left[\frac{1}{x_B - x_A}\int_{x_A}^{x_B} |f(x)|^p dx\right]^{\frac{1}{p}}$$

The norm is defined in the interval $[x_A, x_B]$ and has a further parameter, the exponent p. If the parameter p tends to infinity, the norm will converge to the max norm.

$$\|f(x)\|_\infty = max_{x_A \le x \le x_B}|f(x)|$$

Since the calculation of the integral is difficult or impossible normally, the integral is approximated by a sum.

For the group delay optimization following error norm is used:

$$E = \left[\frac{1}{n}\sum_{k=1}^{n} |t_g(f_k) - t_{gt}(f_k)|^p\right]$$

t_g ... group delay of the actual filter

t_{gt} .. target group delay

f_k ... discrete frequencies

n ... number of points

p ... exponent

The discrete frequencies are linear spaced between the minimum and the maximum frequency which can be specified in the group delay menu.

For the optimization the p-th / Taylor optimization is used. The target of the optimization is to reach the minimum of the error Norm E. The parameters of the optimization are the locations of the transmission zero of the allpass filter and further, you can choose the main delay time as a further parameter. For better convergence, the optimization includes a damping property, which guarantees, that the error norm will decrease in each step. But the optimization algorithm can converge to a local minimum, which can't be checked anyway. The starting values of the optimization determine the result of the optimization.

Much effort was made for getting good starting values for the transmission zeros of the allpass filter. First, the target phase response of the allpass filter is calculated and, using this function, a good estimate for the transmission zeros is made by a special algorithm.

The optimization algorithm will stop, if the error norm will decrease by less than 1ppm (1E-6) at one iteration, or the user will interrupt it using the escape key. If the algorithms will stop automatically the message "optimized" will be shown. If there are more than 10 iterations and the algorithm is stopped by the user, the allpass is considered as optimized too.

Circuits of allpass filters

The design of allpass filters is very easy because you can cascade allpass filters of first and second order, without any influence of each other since every allpass filter has the same wave impedance.

For distinction of different allpass circuits for complex transmission zeros, there will be defined a phase angle for the transmission zero:

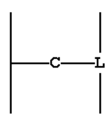
$$\phi = \arg(S)$$

S ... transmission zero (complex number)

Φ ... phase angle of the transmission zero

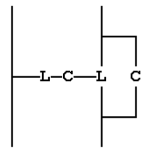
There are four different circuits available:

- one real transmission zero (allpass degree = 1):
  There is only one circuit available for a real transmission zero which includes a mutual inductance.

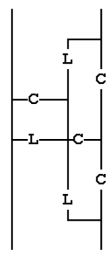


- two conjugated complex transmission zeros, circuit 1:
  This special circuit for a pair of complex conjugated transmission zeros

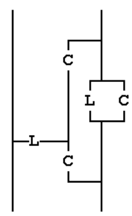
includes a mutual inductance and will be used, only if it is necessary.
usable for: $0 < \Phi < 90°$
used for : $0 < \Phi < 45°$ (*QuickFil* automatic choice)



- two conjugate complex transmission zeros, circuit 2:
  This is a special circuit, which has no mutual inductances but has relatively many elements.
  usable for: $45° < \Phi < 90°$
  used for : $45° < \Phi < 60°$ (*QuickFil* automatic choice)



- two conjugate complex transmission zeros, circuit 3
  This circuits is used if the quality factor of the transmission zero is relatively high.
  usable for: $60° < \Phi < 90°$
  used for : $60° < \Phi < 90°$ (*QuickFil* automatic choice)



Allpass Filters and Spice files and Touchstone files

For Spice files, the quality factor and the tolerances of components of allpass filters cannot be specified individually, but you can specify the same quality factor and the same tolerance for all inductors and for all capacitors.

If there are mutual inductors necessary, the coupling factor has to be one. For Spice there will be troubles if you choose a quality factor of one. Therefore, the quality factor of 0.999999 = (1 - 1E-6) is used.

For Touchstone, there are no coupled inductors used. The circuit in the Touchstone file will include negative inductors, which are equivalent to mutual inductors.

Appendix M - Key Macro Function

QuickFil allows you to save command and data sequences and retrieve them automatically at a later time (creating and recalling macros). This can save a great deal of time in many areas, e.g. in comparative analysis or calculations at the design stage. (Example: selecting the most appropriate approximation for realizing a given specification.) Below are some notes on using macros.

Creating Macros

The first step is to open the macro. This is done via the MACRO menu and Record option. Once you have entered and confirmed the file name, you will be in the program's create mode.

Each keystroke will be recorded and saved until you exit this mode using [Ctrl+z] or the program.

Reactance filter

In Record mode, you can:

Set a break

If you press [Ctrl + p] while creating a macro and then recall the macro at a later date, the sequence will break at this point, enabling you to change inputs/ settings etc. at that point.

Example:

Let's assume you are looking for the best approximation for a given filter. Before running the first approximation type, you open a macro which you can interrupt before you decide on the approximation. Now continue with the macro and complete the calculation sequence. For the next approximation type, you can call up the macro and switch to the new approximation when the break occurs.

Set a text/message output or delay loop

Pressing the key combination [Ctrl+u] makes the RECORD menu appear in the message line. At this point, you can use this to specify whether you want to:

- Display a file
- Issue a message
- Insert a break

at this point when running the macro at a later date.

Note: When inserting delay loops in macros, it may be worth setting the macromode to "timemode" in the macro itself (via the OPTIONS or RECORD menu). Otherwise, the automatic sequence does not run without a break as desired (e.g. if "waitmode" is still set as the result of running a previous macro).

Running Macros

To call a program from DOS;

- At the DOS prompt type the name of the appropriate command file after QF. The .KDO extension is not necessary.
- For example, at the DOS prompt type: QF bandp1 [↵]
- This will start *QuickFil* and call the BANDP1.KDO macro.

Restriction: Macros can only be called up in this way if they are saved in the same directory as the *QuickFil* program and *QuickFil* is called up from that directory (the system does not look for macro files on a path).

To call a program from within *QuickFil****;***

- Select the Call option from the CONTROL menu and enter the name of the macro file. The macro will start running as soon as you confirm the input field entry.

The following options can be built into macros:

- Preset breaks in the automatic sequence, where you can change inputs, analyze curves or tables and much more. Continue the sequence with [Ctrl+p], but first, remember to go back to the same section of the program where you interrupted the sequence.
- Time loops or delay points during which the screen display is ‘frozen’, or a message or text appears on the screen.

Changing the Macro Mode

Pressing the key combination [Ctrl+u] will make the message line display a menu indicating whether you want the automatic sequence to

- Observe the wait times specified in the command file (1 ... Timemode)
- Instead of having a wait time, continue only once the spacebar is pressed (0 ... Waitmode)
- Or, whether the command file should be run without wait points or times (2 ... Fastmode).

When creating command files, you can use the key combination [Ctrl+u] to call up the RECORD menu instead.

([Ctrl+U] while recording a macro)

RECORD: 0 .. Filename, 1 .. Message, 2 .. Timedelay, 3 .. Macromode

This is used to:

- Show the contents of a text file.
- Display a message.
- Bring the macro to a temporary stop.
- Alter the macro sequence at this point during macro execution.

The following options are available:

0 .. Filename

This menu option lets you integrate a macro pause to display the contents of a text file. When you activate this option, the program will prompt you for the name of the text file you want the macro to display on the screen. You will also have to specify how long you want the text to remain on the screen.

1 .. Message

When you activate this option, the program will prompt you for the desired message and the time (number of seconds) during which the inserted message is to be displayed.

2 .. Delay

Prompts you for the number of seconds you wish the macro to pause during execution.

3 .. Macromode

Select this command to enter a submenu which offers the following special macro options:

0 .. Waitmode

Instead of stopping for a subsequently specified pause duration, the macro pauses until the spacebar is pressed. This allows the user to determine execution speed.

1 .. Timemode

Macro pauses for the subsequently specified duration.

2 .. Fastmode

The specified pauses are ignored, the macro is executed without a break. Caution is advised when selecting “Waitmode” or “Fastmode”, as all subsequent pauses will be ignored. If you want to use this for a particular section of your macro, remember to switch back to “Timemode” when defining the macro.

Application:

Is used if you want to invoke a macro while executing another macro, but don’t want the wait loops or wait locations which the second macro contains for this particular application. Before invoking the second macro file, switch to “fastmode” while creating the primary macro, and switch back to “Timemode” after the secondary has been terminated. The macro stops again for all subsequent pauses.

Appendix N - Data Filenames

Most files saved in *QuickFil* are stored in a text format. Because of this, any word processing package can be used to view them. Listed below are the extensions corresponding to text files created by *QuickFil*.

Default Extension File Content

.KDO	Macros
.SPZ	Filter specifications (protocol form)
.FDT	Polynomial zeros (protocol form)
.SCH	Circuits (circuit designs with component values, protocol form)
.CIR	Circuit descriptions in a form readable by the *ISSPICE* circuit analysis program.
.CKT	Circuit descriptions in a form readable by the Touchstone analysis program.
.LIB	A Collection of filter subcircuits in an *ISSPICE* format.
.PLT	Graphics in HPGL format.
.BAK	Backup file, which *QuickFil* automatically makes if an existing file is overwritten.

Some files are designed solely for use within *QuickFil* and are not in a text format. The extensions for these files are:

.QF	Stored filter characteristics.
.SH	Stored circuit description.
.GDF	Graphics (diagrams, curves)

Note: The program adds these extensions unless otherwise specified. If you specify another extension, *QuickFil* will of course use this when saving the file.

The QF.DEF file is used to store *QuickFil* preferences. Every time the program is started this file is read for program defaults.

Appendix O - Macro File Format (.KDO)

(QUICKFIL:\Macro:\Record)

Keyboard inputs are saved as ASCII characters, together with remarks enabling them to be edited. Command files can also be created or changed in other programs (word processors).

The remarks, which are separated from the commands themselves by spaces, do not need to be input.

Example:

```
f    MAIN: Filtertype
b
t    FILTERTYPE: Type
b    TYPE: Bandpass
a    FILTERTYPE: Approximation
e
#13  []     APPROXIMATION: Elliptic
q    FILTERTYPE: Quit
s    MAIN: Specification
a    SPECIFICATION: A
100k
#24  Ctrl-X  (CuDn)
120k
#24  Ctrl-X  (CuDn)
80k
#24  Ctrl-X  (CuDn)
150k
#24  Ctrl-X  (CuDn)
0.2
#24  Ctrl-X  (CuDn)
13.47
#24  Ctrl-X  (CuDn)
21.21
#24  Ctrl-X  (CuDn)
60
#24  Ctrl-X  (CuDn)
8
#13  []
i    SPECIFICATION: I
c
#13  []
q    SPECIFICATION: Quit
a    QUICKFIL: polynomial_Analysis
#32  [Spacebar]
#32  [Spacebar]
#13  []     POLY. ANALYSIS: Property
p    PROPERTY: Pulse_response
q    PROPERTY: Quit
r    POLY. ANALYSIS: Repres.
v    REPRESENTATION: Voltage[V]
q    REPRESENTATION: Quit
#13  []     POLY. ANALYSIS: Graph
i    X-Y: optImize
```

Appendix P - Filter Specification Protocol (.SPZ)

(QUICKFIL:\Specification:\File)

The contents of the SPECIFICATION menu are held in an open text format.

For example:

```
Elliptic (Cauer) - bandpass filter


Lower passband edge frequency   :     100.000 000 kHz
Upper passband edge frequency   :     120.000 000 kHz
Lower stopband edge frequency   :      80.000 000 kHz
Upper stopband edge frequency   :     150.000 000 kHz

Passband bandedge loss          :        0.200 000 dB
Passband bandedge return loss   :            13.47 dB
Passband reflection factor      :            21.21 %
Stopband loss                   :            60.17 dB

Filter degree                   :                   8
Case                            :                   c

Lower 3dB edge frequency        :      98.441 053 kHz
Upper 3dB edge frequency        :     121.900 362 kHz

Filter quality                  :               22.16
```

Appendix Q - Polynomial Pole/Zero Protocol (.FDT)

(QUICKFIL:\Roots:\File)

Polynomial zeros are saved in an ASCII text format. Natural modes are the roots of the denominator e(s) of the transfer function shown in Appendix D. Transmission zeros are the roots of the numerator p(s) of the transfer function.

For example:

```
Elliptic (Cauer) - bandpass filter


Reference frequency      :       109.544 512 kHz
Reference quality factor :             5.477 226


Normalized natural modes
------------------------

   -0.060 400 102 832  +-j    0.959 304 425 752
   -0.065 374 221 149  +-j    1.038 305 842 840
   -0.024 913 546 144  +-j    1.103 927 100 460
   -0.020 433 069 874  +-j    0.905 395 781 437



Normalized transmission zeros
-----------------------------

    0.000 000 000 000
    0.000 000 000 000
    0.000 000 000 000  +-j    0.709 492 119 176
    0.000 000 000 000  +-j    1.409 458 925 579



Normalized reflection zeros
---------------------------

   -0.000 000 000 000  +-j    0.920 129 257 947
   -0.000 000 000 000  +-j    1.000 000 000 000
   -0.000 000 000 000  +-j    1.000 000 000 000
   -0.000 000 000 000  +-j    1.086 803 828 226
```

```
Constants of transfer function and characteristic function

Cep =         54.575 938 dB
Cfe =          0.000 000 dB
```

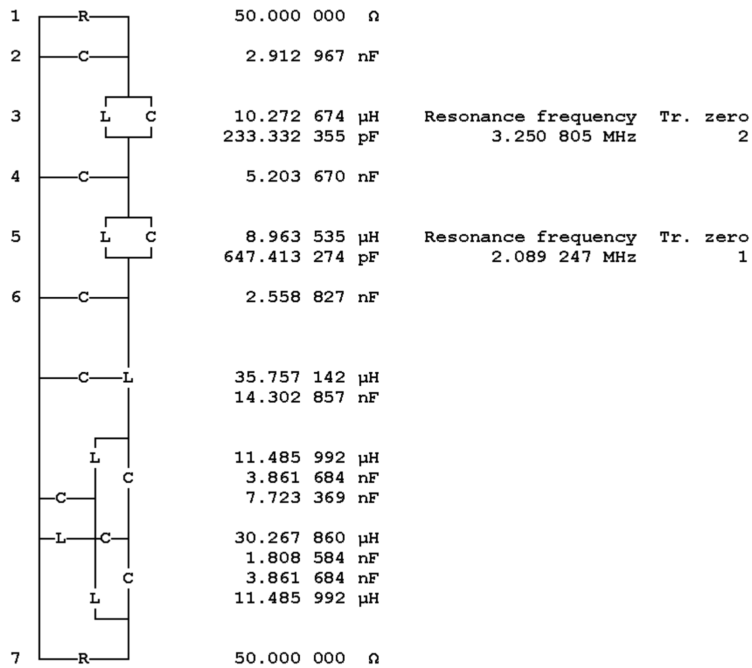
Appendix R - Circuit Protocol (.SCH)

(QUICKFIL:\Passive Design:\Output Circuit:\File) or

(QUICKFIL:\Passive Design:\Manipulation and Analysis:\Output Circuit:\File)

Circuits are saved in an open text format.

For example:



Appendix S - Internal Format(.QFT)

(QUICKFIL:\Roots:\Filterdata(File))

Using this format, you can store data which can be read by other programs (for example DPA 1000). The transfer function is described by the zeros of the polynomials e(s), p(s) and f(s).

Example:

```
BEGIN QUICKFIL-FILTERDATA
  REM
  FILTERTYPE    BANDPASS
  APPROXIMATION MAXIMALLYFLAT
  OMEGA-0    4.34846812935856E+0008
  QUALITY-B  3.29561998888086E+0000
  LN(CE/CP)  4.49763012936068E+0000
  LN(CF/CE)  0.00000000000000E+0000
  BEGIN POLYNOM-E
    COMPLEX-NUMBER  1 -1.70512680075770E-0001  1.01433646235328E+0000
    COMPLEX-NUMBER  2 -1.28820623915199E-0001  9.08973458805813E-0001
    COMPLEX-NUMBER  3 -1.25762679983376E-0001  1.11625558978868E+0000
    COMPLEX-NUMBER  4 -4.27524489411997E-0002  8.63910978936925E-0001
    COMPLEX-NUMBER  5 -4.17321656663104E-0002  1.15910149081302E+0000
    COMPLEX-NUMBER  6 -8.79603647435484E-0002  9.50884795309847E-0001
    COMPLEX-NUMBER  7 -8.75294078569296E-0002  1.07058173393246E+0000
    REAL-NUMBER     1 -3.61644716246129E-0001
    REAL-NUMBER     2 -3.60816734676719E-0001
    POLYNOM E 16  7  2
  END POLYNOM-E
  BEGIN POLYNOM-P
    COMPLEX-NUMBER  1  0.00000000000000E+0000  7.08010432393843E-0001
    COMPLEX-NUMBER  2  0.00000000000000E+0000  1.31487651730285E+0000
    COMPLEX-NUMBER  3  8.79603647435484E-0002  9.50884795309847E-0001
    COMPLEX-NUMBER  4  8.75294078569296E-0002  1.07058173393246E+0000
    REAL-NUMBER     1  0.00000000000000E+0000
    REAL-NUMBER     2  0.00000000000000E+0000
    REAL-NUMBER     3  0.00000000000000E+0000
    POLYNOM P 11  4  3
  END POLYNOM-P
  BEGIN POLYNOM-F
    COMPLEX-NUMBER  1 -0.00000000000000E+0000  1.01251572229799E+0000
    COMPLEX-NUMBER  2 -0.00000000000000E+0000  1.01251572229799E+0000
    COMPLEX-NUMBER  3 -0.00000000000000E+0000  1.01251572229799E+0000
    COMPLEX-NUMBER  4 -0.00000000000000E+0000  1.01251572229799E+0000
    COMPLEX-NUMBER  5 -0.00000000000000E+0000  1.01251572229799E+0000
    REAL-NUMBER     1 -3.61229812445838E-0001
    REAL-NUMBER     2  3.61229812445838E-0001
    POLYNOM F 12  5  2
  END POLYNOM-F
END QUICKFIL-FILTERDATA
```

Appendix T - SPICE Format(.CIR)

(QUICKFIL:\Passive Design:\Output Circuit:\Spice) or

(QUICKFIL:\Passive Design:\Manipulation and Analysis:\Output Circuit:\Spice)

This format is used to save the filter circuit for use with the *ISSPICE* circuit analysis program, or any other SPICE compatible simulator. Circuits can be saved using standalone (ready to simulate) or subcircuit (*ISSPICE* Library) formats. Monte Carlo component tolerance syntax can also be activated on any component.

Below, is an example of *ISSPICE* output generated using a standalone netlist format with component tolerance statements set up for a Monte Carlo analysis.

Example:

In the standalone format, a source is added to the input. Its AC value, 1 or 2, will depend on the type of termination. 2 when input and output terminations exist, and 1 for other orientations.

```
*
* QuickFil 5.1           OMICRON electronics    2002-06-04  13:55:52
* Elliptic (Cauer) - bandpass filter
*
* Lower passband edge frequency   :     100.000 000 kHz
* Upper passband edge frequency   :     120.000 000 kHz
* Lower stopband edge frequency   :      80.000 000 kHz
* Upper stopband edge frequency   :     150.000 000 kHz
* Passband bandedge loss          :        0.200 000 dB
* Stopband loss                   :            60.17 dB
* Filter degree                   :                8
* Case                            :                c
*
L2         2     3    405.987U
C3         3     4    5.19932N
L4       200     0    57.1678U
C4         4   200    73.3521N
L5       201     0    28.7771U
C5         4   201    36.9239N
L6         4     5    581.217U
C7         5   500    3.63179N
RH7        5   500    1.00000T
L8       500     0    12.4309U
C9       500     0    169.807N
VIN        1     0    AC    2    0
R1         1     2    50.0000
R10      500     0    50.0000
.AC  DEC  100   10.0000K   1.00000MEG
.OPTIONS LIMPTS=201
.PRINT  AC  V(500)  VP(500)
.END
```

Note: In the standalone format a source is added to the input. Its AC value, 1 or 2, depends on the type of termination. 2V is used, if the filter is a doubly terminated filter, 1V is used for singly terminated filters.

Appendix U - Touchstone Format(.CKT)

(QUICKFIL:\Passive Design:\Output Circuit:\Touchstone) or

(QUICKFIL:\Passive Design:\Manipulation and Analysis:\OutputCircuit:\Touchstone)

This format is used to save the filter circuit for use with the Touchstone circuit analysis program.

Here is an example:

```
!
! QuickFil 5.1           OMICRON electronics    2002-06-04  14:05:09
! Elliptic (Cauer) - bandpass filter
!
! Lower passband edge frequency   :     100.000 000 kHz
! Upper passband edge frequency   :     120.000 000 kHz
! Lower stopband edge frequency   :      80.000 000 kHz
! Upper stopband edge frequency   :     150.000 000 kHz
! Passband bandedge loss          :        0.200 000 dB
! Stopband loss                   :            60.17 dB
! Filter degree                   :                   8
! Case                            :                   c
!

DIM
     FREQ     HZ
     RES      OH
     IND      H
     CAP      F
     TIME     SEC

CKT
     SLC     1     2  L = 4.0599E-04  C = 5.1993E-09
     SLC     2     0  L = 5.7168E-05  C = 7.3352E-08
     SLC     2     0  L = 2.8777E-05  C = 3.6924E-08
     SLC     2     3  L = 5.8122E-04  C = 3.6318E-09
     PLC     3     0  L = 1.2431E-05  C = 1.6981E-07
     DEF2P   1     3  FILTER

TERM
     Z0 = 5.0000E+01
     FILTER  0.0000E+00  0    0.0000E+00  0

OUT
     FILTER  DB[S21] GR1

FREQ
     SWEEP   1.0000E+04    1.0000E+06    1.0000E+04
```

Appendix V - Graphics Data Format(.PLT)

(...\X-Y:\Output Graphic:\File)

This stores graphics displayed on the screen in the X-Y menu, in HPGL format, under the name entered. This data can be used in DOS to output a diagram to a plotter or to a DTP package.

Here is an example of a file:

```
SC0.00,639.00,0.00,479.00;
SP10;
PU63.90,123.99;
PD619.83,123.99;
PD619.83,470.23;
PD63.90,470.23;
PD63.90,123.99;
SP14;
PU243.46,82.75;
PD243.46,104.27;
PD252.83,104.27;
PU243.46,94.70;
PD251.27,94.70;
PU259.08,82.75;
PD259.08,97.09;
PU259.08,92.31;
PD262.20,97.09;
PD265.33,97.09;
PU271.58,89.92;
PD279.39,89.92;
PD279.39,92.31;
PD277.82,97.09;
PD273.14,97.09;
PD271.58,92.31;
PD271.58,87.53;
PD273.14,82.75;
PD277.82,82.75;
PD279.39,85.14;
PU293.44,77.97;
PD293.44,97.09;
PU293.44,94.70;
PD291.88,97.09;
PD287.20,97.09;
PD285.63,92.31;
PD285.63,87.53;
PD287.20,82.75;
PD291.88,82.75;
...
...
```

Appendix W - Definition File Format (QF.DEF)

This file is used to save program parameters (e.g. printer and plotter interface, extensions, etc.). This file is read every time *QuickFil* is started. Any attribute changes made within *QuickFil* will be added to this file when *QuickFil* is exited. These attributes include:

- Graphics attributes (linestyle, color, height, width etc.)
- Default extensions
- Default limits

The definition file is divided into subsections, which will be explained in detail:

Programmdefaults:

The program defaults describe the interface parameters, which can be changed in the OPTION menu.

Here is an example of the Block:

```
BEGIN PROGRAMDEFAULTS
  DECIMALSIGN    .
  GROUPING       ON
  PRINTINTERFACE LPT1:
  IBMPRINTER     FALSE
  POSTSCRIPT     FALSE
  PRINTERINIT
  PRINTBORDER    8
  PRINTLINES     60
  CHARPERLINE    80
  PLOTINTERFACE  LPT2:
  SERIALPORT     COM1: DISABLED
  SERIALPORT     COM2: DISABLED
  SERIALPORT     COM3: DISABLED
  SERIALPORT     COM4: DISABLED
END PROGRAMDEFAULTS
```

Some brief notes on the individual entries:

- REM last update: Contains details of when QF.DEF was last overwritten or otherwise amended.
- DECIMALSIGN "." OPTIONS: Decimal_point
  "," OPTIONS: Decimal_comma
- GROUPING OFF OPTIONS: Grouping_off,
  ON OPTIONS: Grouping_on
- PRINTINTERFACE LPT1(2,3): Printer interface: LPT1(2,3):
  COM1(2): Printer interface: COM1(2):
- IBMPRINTER Here, you can specify if the printer can print IBM characters. If the printer cannot print the IBM characters, substitutions of the characters are made.
- POSTSCRIPT Here, you can specify if the printer is a postscript printer.
- PRINTERINIT Here, you can specify an initialization sequence at the beginning of each printing (see OPTION menu).
- PRINTBORDER Here, you can specify the number of blanks at the beginning of each line.
- PRINTLINES Here, you can specify the number of lines per page.
- CHARPERLINE Here, you can specify the number of characters per line, which can be printed by the printer.
- PLOTINTERFACE LPT1(2,3): Plotter interface: LPT1(2,3):
  COM1(2): Plotter interface: COM1(2):
- SERIALPORT Specifies interface parameters for COM1:, COM2: COM3: and COM4: For Windows operation system it is recommended to disable all serial ports. You can specify the parameters in the OPTION menu of *QuickFil*.

FILENUMBERS:

This block is used by the library which is used by *QuickFil*. However, the program itself does not need these items.

```
BEGIN FILENUMBERS
  FILENUMBER 1,   1
  FILENUMBER 2,   1
  FILENUMBER 3,   1
  FILENUMBER 4,   1
  FILENUMBER 5,   1
  FILENUMBER 6,   1
  FILENUMBER 7,   1
  FILENUMBER 8,   1
  FILENUMBER 9,   1
END FILENUMBERS
```

GRAPHICDEFAULTS:

Here, you will find all the necessary information for the graphic of *QuickFil*.

Here is an example:

```
BEGIN GRAPHICDEFAULTS
  DISPLAYTYPE    Autodetect (Last used: 640 x 480 VGA)
  FONTFILE       FONT1.FON
  MAXPEN         4
  FASTTEXT       FALSE
  REM Format for LineAttr: Id,Color,LineStyle,PrPen,PlPen
  REM Format for TextAttr: Id,Height,Width,Angle,Color,PrPen,PlPen
  TEXTATTR AXISNUMBERX,3.0000,1.1000,0.0000,WHITE,16,16
  TEXTATTR AXISNUMBERY,2.0000,1.7000,90.0000,WHITE,15,15
  TEXTATTR AXISTEXTX,6.0000,2.2000,0.0000,WHITE,14,14
  TEXTATTR AXISTEXTY,4.0000,3.4000,90.0000,WHITE,13,13
  LINEATTR GRIDPOINT,LIGHTGRAY,DOT,12,12
  LINEATTR GRIDLINE,LIGHTGRAY,DASH,11,11
  LINEATTR GRIDBORDER,WHITE,FULL,10,10
  LINEATTR LOGOLINE,LIGHTCYAN,FULL,9,9
  LINEATTR CURVE1,YELLOW,FULL,1,1
  LINEATTR CURVE2,LIGHTGREEN,FULL,2,2
  LINEATTR CURVE3,LIGHTCYAN,FULL,3,3
  LINEATTR CURVE4,LIGHTRED,FULL,4,4
  LINEATTR CURVE5,LIGHTMAGENTA,FULL,5,5
  LINEATTR CURVE6,BROWN,FULL,6,6
  LINEATTR CURVE7,GREEN,FULL,7,7
  LINEATTR CURVE8,LIGHTBLUE,FULL,8,8
  LINEATTR AXISLINE,LIGHTGRAY,FULL,11,11
  LINEATTR UNITCIRCLE,LIGHTGRAY,FULL,11,11
  TEXTATTR POLTEXT,2.0000,1.1000,0.0000,YELLOW,1,1
  TEXTATTR ZEROTEXT,2.0000,1.1000,0.0000,LIGHTRED,4,4
  TEXTATTR KOORDTEXT,2.0000,1.1000,0.0000,YELLOW,1,1
  TEXTATTR PNHEADERTEXT,2.0000,1.1000,0.0000,WHITE,15,15
END GRAPHICDEFAULTS
```

Note: GraphicDefaults contains parameters that cannot be changed in the program, but the user may alter them in special cases.

The entry in the first line, “DISPLAYTYPE”, tells the program which monitor to use. This normally contains the entry “Autodetect”. This means that the program automatically detects what graphics card is installed. If it fails to do so, the appropriate monitor data can be entered manually at this point, requiring the program to use the proper resolution.

The “DEF” file can also be used to change line and text attributes, which govern the appearance of lines and text on the screen and when printed/plotted. The following diagram components can be changed: Curve 1 through Curve 8, Logoline, Gridborder, Gridpoint, AxistextX, Axistexty, AxisnumberX, and AxisNumberY.

Listed below are some hints for possible settings:

LINEATTR Id, Color, LineStyle, Pen

"Id" can be:
CURVE1 to CURVE8, GRIDBORDER, GRIDLINE, GRIDPOINT, LOGOLINE

“Color” can be set to:
BLACK, BLUE, GREEN, CYAN, RED, MAGENTA, BROWN, LIGHTGRAY, DARKGRAY, LIGHTBLUE, LIGHTGREEN, LIGHTCYAN, LIGHTRED, LIGHTMAGENTA, YELLOW, WHITE

“LineStyle” can be set to:
FULL, DOT, DASH, DASHDOT

“Pen (number of pen in plotter output) can be set to: 1 to 16

TEXTATTR Id, High, Width, Angle, Color

"Id" can be:
AXISTEXTX, AXISTEXTY, AXISNUMBERX, AXISNUMBERY

“Height” and “width” are stated (approximately) as a percentage of the screen width/height. We recommend that you try this out for yourself until you find the settings you want.

The “angle” is reckoned counter-clockwise (0 degrees is horizontal). “Color” is like LINEATTR.

Practical Example:

If you would like to double the font size in the graphic output;

• Change the lines
“TEXTATTR AXISTEXTY 4.0 3.4 90.0 WHITE, 13”
“TEXTATTR AXISTEXTX 6.0 2.2 0.0 WHITE, 14”
Id is the identity of the attribute and must NOT be changed. Id is the identity of the attribute and must NOT be altered.

RASTERIZER

In this section, *QuickFil* saves all printer parameters for prints of the graphical output which is set in the PRINTER OPTIONS menu.

Here is an example:

```
BEGIN RASTERIZER
  DRIVER     HP LaserJet
  RESOLUTION 2
  PAPER      198,270
  COPIES     1
  INVERSE    FALSE
  ROTATE     0
  SCALE      1.000,1.000
  OFFSET     0.000,0.000
  PEN        1,4,9
  PEN        2,4,8
  PEN        3,4,7
  PEN        4,4,6
  PEN        5,4,5
  PEN        6,4,4
  PEN        7,4,3
  PEN        8,4,2
  PEN        9,6,8
  PEN        10,1,9
  PEN        11,1,9
  PEN        12,1,9
  PEN        13,5,9
  PEN        14,5,9
  PEN        15,3,9
  PEN        16,3,9
  PEN        17,1,9
  PEN        18,1,9
  PEN        19,1,9
  PEN        20,1,9
END RASTERIZER
```

QuickFil

In this section, you can change the extensions which *QuickFil* automatically adds to a filename if no extension is specified. Also, it is possible to enter an initialization resistance for the passive design part.

```
BEGIN QuickFil
  EXTENSION-SPICE CIR
  EXTENSION-TOUCHSTONE CKT
  EXTENSION-CIRCUIT-PICTURE SCH
  EXTENSION-CIRCUIT SH
  EXTENSION-ZEROS FDT
  EXTENSION-AFIL QFT
  EXTENSION-QFINIT QF
  EXTENSION-SPEZ SPZ
  EXTENSION-PASSIVPARAM PP
  EXTENSION-POLYPARAM PN
  EXTENSION-GRAFIK GDF
END QuickFil
```

BTICFILT

This part contains constants which control the input limits for Butterworth, Chebyshev, inverse Chebyshev and Elliptic filter calculations. For example (with default values):

```
BEGIN BTICFILT
  FMIN 1.000000m
  FMAX 100.000000G
  KPMIN 0.001000
  KPMAX 1.000000
  DSMIN 5.000000
  DSMAX 200.000000
  KFMIN 0.010000
  KFMAX 0.999900
  QMAX 1000.000000
END BTICFILT
```

Meaning:

- FMIN — Lowest edge frequency to be entered. This is the lower limit for passband edge frequencies and stopband edge frequencies.
- FMAX — Maximum edge frequency as in FMIN.The calculated values can either exceed or fall below the values mentioned above. However, it is not possible to enter values outside of these limits. This also applies to the entry of the bandwidth or relative bandwidth, if the results are to be "retransformed" to frequency values.
- KPMIN — Minimum value of the factor kp Typical value: 0.001 The factor kp depends on the passband loss, return loss, the reflection factor, and the proposed resistance ratio. A value of 0.001 is

equivalent to a maximum return loss of 60 dB and a minimum reflection factor of 0.1%. The minimum must be higher than zero.

- KPMAX — Maximum value of the factor kp. Typical value 1.000. The maximum must be less than or equal 1, so that the 3 dB edge frequencies are calculated correctly. If you drop the 3 dB limits, you can increase the factor as much you like. The value KPMAX = 1 is equivalent to a maximum passband loss of 3.0103 dB.
- DSMIN — Minimum stopband loss
- DSMAX — Maximum stopband loss When calculating stopband losses it is possible for your values to exceed or fall below the above limits. However, it is not possible to make entries outside of these limits.
- KFMIN — Minimum frequency factor (0 < kfmin < kfmax < 1). This factor determines the minimum ratio between the passband edge frequency and the stopband edge frequency in a lowpass. In the other types of filters, the factor refers to the transformed lowpass filter.
- KFMAX — Maximum frequency factor (0 < kfmin < kfmax < 1). The value is slightly less than 1, and should not be selected too high to avoid "runtime errors". This factor determines the maximum ratio between the passband edge frequency and the stopband edge frequency in a lowpass. In the other types of filters, the factor refers to the transformed lowpass filter.
- QMAX — Determines the maximum operating quality for the bandpass and bandstop.

$$Q = \frac{\sqrt{f_{do} \cdot f_{du}}}{f_{do} - f_{du}}$$

ERMFFILT

This part contains constants which control the entry limits for equal ripple and maximally flat filters calculations.

Example:

```
BEGIN ERMFFILT
  AUSG3DB TRUE
  FMIN 1.000000m
  FMAX 100.000000G
  KPMIN 0.001000
  KPMAX 1.000000
  DSMAX 200.000000
  KFMIN 0.000010
  KFMAX 0.999900
  PAMAX 1.000000k
  D0MAX5.000000
  MINOPT 0.010000
END ERMFFILT
```

Meaning:

- AUSG3dB — (TRUE/FALSE) The automatic calculation of the 3dB edge frequency can be switched off.The 3dB edge frequency is only calculated in combination with the filter quality. This parameter can always be changed.
- FMIN — Minimum frequency of the passband edges
- FMAX — Maximum frequency of the passband edges
- KPMIN — Minimum value of the factor kp. The factor kp depends on the passband loss, return loss, the reflection factor and the proposed resistance ratio. A value of 0.001 is equivalent to a maximum return loss of 60 dB and a minimum reflection factor of 0.1 %. The minimum must be higher than zero.
- KPMAX — Maximum value of the factor kpThe maximum must be less than or equal 1, so that the 3dB edge frequencies are calculated correctly. If you drop the 3 dB edges, you can increase the factor as much you like. The value KPMAX = 1 is equivalent to a maximum passband loss of 3.0103dB.
- DSMAX — Maximum stopband loss. This limit is used as the maximum when the tolerance scheme of the stopband loss is entered. KFMIN Minimum frequency factor (0 < kfmin < kfmax < 1). This factor serves as the limit when entering the transmission zero frequencies. It limits the transmission zero frequencies that are farthest

away from the passband edge frequency. This also applies when entering the tolerance scheme.

- KFMAX — Maximum frequency factor(0 < kfmin < kfmax < 1). This factor sets the limitations for transmission zero frequencies closest to the passband edge frequency. This also applies to the tolerance scheme, the minimum distance of the passband edge frequencies of the bandpass, and the bandstop.
- PAMAX — Maximum ratio of the parameter to the default parameter in parametric bandpass filters. The minimum ratio is equal to 1 / PAMAX.
- D0MAX — Attenuation at zero or infinity for in lowpass, highpass filters, whose degree is even and for bandstops filters, whose degree is divisible by four. The minimum attenuation is zero. This parameter is only important for equal ripple approximation.
- MINOPT — Minimum terminating edge for the optimization. This parameter also influences the calculation precision when determining the extreme values of the loss function in the stopband. The precision lies around 1/4 of MINOPT. If MINOPT is reduced considerably, the calculating time increases during the optimization.

BESSELFILTER

This section contains constants controlling the entry limits for Bessel lowpass filter calculations.

Example:

```
BEGIN BESSELFILT
  FMIN 1.000000m
  FMAX 100.000000G
END BESSELFILT
```

Meaning:

- FMIN — Minimum passband edge frequency
- FMAX — Maximum passband edge frequency. Calculated frequencies might exceed or fall below these values. However, it is not possible to make entries outside of these limits.

PASSIV

In this block you can specify the initialization resistance for passive filter realization. Further, you can specify that all circuit elements be displayed as graphical symbols (European symbols).

```
BEGIN PASSIV
  RINIT 50.000000
  GRAPHCIRCUIT FALSE
END PASSIV
```

Appendix X - Frequently Asked Questions

Up to what frequency range can** QuickFil **be used?

Estimates: no practical upper limit; for circuit design, only as far as designs are feasible with discrete components (L's and C's).

Does the circuit synthesis consider parasitic components?

No. Synthesis assumes that components are ideal. In analysis, the quality of inductors and capacitors can be taken into account when assessing the expected filter characteristics.

What language is the program written in?

Mostly Turbo Pascal, with some parts in assembler.

Does the program support the numeric coprocessor?

Partially. But not for calculations since the coprocessor is not accurate enough. This is why we have written our own mathematical routines in the assembler. The coprocessor is used mainly for graphics.

Appendix Y - Literature

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[3] R. Saal and E. Ulbrich, “On the design of filters by synthesis"IRE Trans. Circuit Theory, vol. CT-5, pp. 284-327, December 1958

[4] R. Saal und W. Entenmann, “Handbook of Filter Design” Elitera Verlag, Berlin, 1979

[5] H.J. Orchard and G.C. Temes, “Filter design using transformed variables”, IEEE Trans. Circuit Theory, vol. CT-15, pp. 385-408, December 1968

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[7] F. Kaufmann, "Synthese von Reaktanzfiltern", ISBN 3-486-22884-6, Oldenbourg Verlag, 1994

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