= 1 . Likewise , the nominal or characteristic impedance of the filter is set to R ' = 1 ? .

In principle , any non @-@ zero frequency point on the filter response could be used as a reference for the prototype design . For example , for filters with ripple in the passband , the corner frequency is usually defined as the highest frequency at maximum ripple rather than 3 dB . Another case is in image parameter filters (an older design method than the more modern network synthesis filters) which use the cut @-@ off frequency rather than the 3 dB point since cut @-@ off is a well @-@ defined point in this type of filter .

The prototype filter can only be used to produce other filters of the same class and order . For instance , a fifth @-@ order Bessel filter prototype can be converted into any other fifth @-@ order Bessel filter , but it cannot be transformed into a third @-@ order Bessel filter or a fifth @-@ order Tchebyscheff filter .

= = Frequency scaling = =

The prototype filter is scaled to the frequency required with the following transformation :

<formula>

where ?c ' is the value of the frequency parameter (e.g. cut @-@ off frequency) for the prototype and ?c is the desired value . So if ?c ' = 1 then the transfer function of the filter is transformed as :

<formula>

It can readily be seen that to achieve this , the non @-@ resistive components of the filter must be transformed by :

<formula> and , <formula>

= = Impedance scaling = =

Impedance scaling is invariably a scaling to a fixed resistance . This is because the terminations of the filter , at least nominally , are taken to be a fixed resistance . To carry out this scaling to a nominal impedance R , each impedance element of the filter is transformed by :

<formula>

It may be more convenient on some elements to scale the admittance instead :

<formula>

It can readily be seen that to achieve this , the non @-@ resistive components of the filter must be scaled as :

<formula> and , <formula>

Impedance scaling by itself has no effect on the transfer function of the filter (providing that the terminating impedances have the same scaling applied to them) . However , it is usual to combine the frequency and impedance scaling into a single step :

<formula> and , <formula>

= = Bandform transformation = =

In general , the bandform of a filter is transformed by replacing i? where it occurs in the transfer function with a function of i? . This in turn leads to the transformation of the impedance components of the filter into some other component (s) . The frequency scaling above is a trivial case of bandform transformation corresponding to a lowpass to lowpass transformation .

= = = Lowpass to highpass = = =

The frequency transformation required in this case is :

<formula>

where ?c is the point on the highpass filter corresponding to ?c ' on the prototype . The transfer function then transforms as :

<formula>
Inductors are transformed into capacitors according to ,
<formula>
and capacitors are transformed into inductors ,
<formula>
the primed quantities being the component value in the prototype .

= = = Lowpass to bandpass = = =

In this case , the required frequency transformation is :
<formula>
where Q is the Q @-@ factor and is equal to the inverse of the fractional bandwidth :
<formula>
If ?1 and ?2 are the lower and upper frequency points (respectively) of the bandpass response corresponding to ?c ' of the prototype , then ,
<formula> and <formula>
?? is the absolute bandwidth , and ?0 is the resonant frequency of the resonators in the filter . Note that frequency scaling the prototype prior to lowpass to bandpass transformation does not affect the resonant frequency , but instead affects the final bandwidth of the filter .
The transfer function of the filter is transformed according to :
<formula>
Inductors are transformed into series resonators ,
<formula>
and capacitors are transformed into parallel resonators ,
<formula>

= = = Lowpass to bandstop = = =

The required frequency transformation for lowpass to bandstop is :
<formula>
Inductors are transformed into parallel resonators ,
<formula>
and capacitors are transformed into series resonators ,
<formula>

= = = Lowpass to multi @-@ band = = =

Filters with multiple passbands may be obtained by applying the general transformation :
<formula>
The number of resonators in the expression corresponds to the number of passbands required . Lowpass and highpass filters can be viewed as special cases of the resonator expression with one or the other of the terms becoming zero as appropriate . Bandstop filters can be regarded as a combination of a lowpass and a highpass filter . Multiple bandstop filters can always be expressed in terms of a multiple bandpass filter . In this way it , can be seen that this transformation represents the general case for any bandform , and all the other transformations are to be viewed as special cases of it .
The same response can equivalently be obtained , sometimes with a more convenient component topology , by transforming to multiple stopbands instead of multiple passbands . The required transformation in those cases is :
<formula>

= = Alternative prototype = =

In his treatment of image filters , Zobel provided an alternative basis for constructing a prototype which is not based in the frequency domain . The Zobel prototypes do not , therefore , correspond to any particular bandform , but they can be transformed into any of them . Not giving special significance to any one bandform makes the method more mathematically pleasing ; however , it is not in common use .

The Zobel prototype considers filter sections , rather than components . That is , the transformation is carried out on a two @-@ port network rather than a two @-@ terminal inductor or capacitor . The transfer function is expressed in terms of the product of the series impedance , Z , and the shunt admittance Y of a filter half @-@ section . See the article Image impedance for a description of half @-@ sections . This quantity is nondimensional , adding to the prototype 's generality . Generally , ZY is a complex quantity ,

<formula> and as U and V are both , in general , functions of ? we should properly write ,

<formula>

With image filters , it is possible to obtain filters of different classes from the constant k filter prototype by means of a different kind of transformation (see composite image filter) , constant k being those filters for which Z / Y is a constant . For this reason , filters of all classes are given in terms of U (?) for a constant k , which is notated as ,

<formula>

In the case of dissipationless networks , i.e. no resistors , the quantity V (?) is zero and only U (?) need be considered . Uk (?) ranges from 0 at the centre of the passband to -1 at the cut @-@ off frequency and then continues to increase negatively into the stopband regardless of the bandform of the filter being designed . To obtain the required bandform , the following transforms are used :

For a lowpass constant k prototype that is scaled :

<formula>

the independent variable of the response plot is ,

<formula>

The bandform transformations from this prototype are ,

for lowpass , <formula>

for highpass , <formula>

and for bandpass , <formula>