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SEMINAR NOTES: FILTER, THEORY, DESIGN AND APPLICATIONS,(U)
JAN 75 R W STEER, R L ALTMAN, S M RUSCIO

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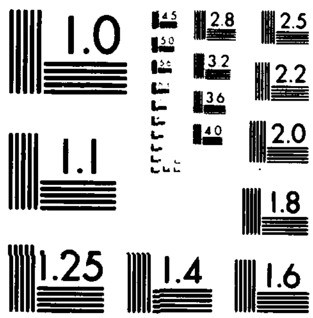

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Frequency Devices INC. 25 Locust Street, Haverhill, Massachusetts 01830 617/374-0761

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SEMINAR NOTES:

FILTER, THEORY, DESIGN AND APPLICATIONS,

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Dr. Robert W. Steer Jr.
Ronald L. Altman
Steven M. Ruscio

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NOTES

A BASIC REVIEW OF FILTER THEORY

Ronald L. Altman

Director of Engineering
Frequency Devices, Inc.

IDEAL FILTER

There are two ways of viewing the characteristics of an ideal filter. The first and most common is from the point of view of the frequency domain. This first class of filters would pass without attenuation all frequencies inside certain frequency limits and provide infinite attenuation for all other frequencies. The second point of view is from the time domain. A designer may only be interested in the time response of a filter. This class of filters would have an output that is identical to its input except be delayed by some time τ_0.

This set of notes will present in a basic manner some of the characteristics of mathematical models which have been derived to approximate these two types of filters.

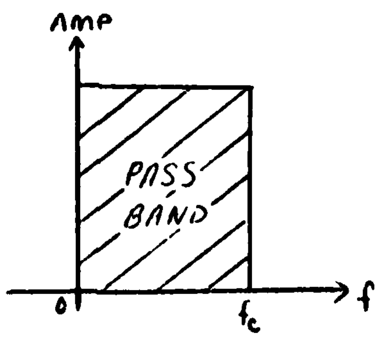
The first class of filters to be considered are those whose desired response is to pass or attenuate specific frequency ranges. We will discuss four basic types of filters in this class. These are Low-Pass, Band Pass, High Pass, and Bandreject.

Based on the definition of the ideal filter the following filter types can be defined.

Low-Pass Filter

Passes signals from zero frequency up to a certain cutoff frequency and rejects all signals whose frequency is beyond the cutoff frequency.

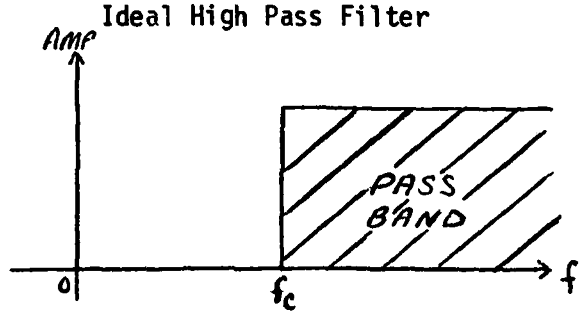
Frequency Domain
Representation of
Ideal Low-Pass Filter



High Pass Filter

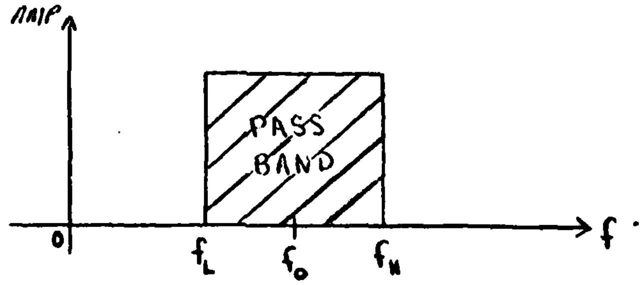
Rejects all signals up to f_c and passes all frequencies greater than f_c with no attenuation.

Frequency Domain
Representation of
Ideal High Pass Filter



Band Pass Filter

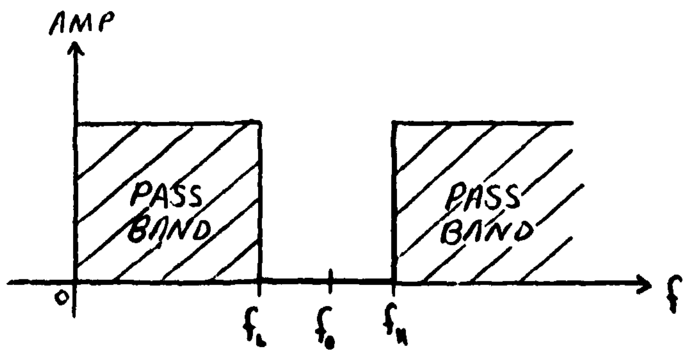
Passes only those signals that are within a specific frequency band and rejects all others.



$$f_o = \sqrt{f_L \times f_H}$$

Bandreject Filter

Rejects those signals within a specific frequency band and passes all others.



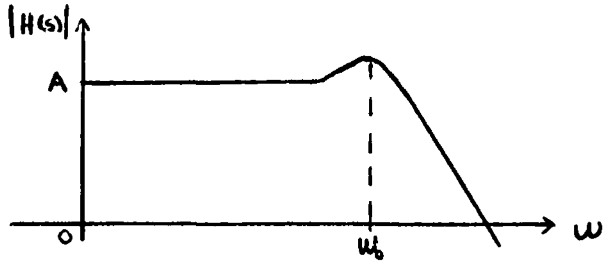
The shapes of these filters must be approximated by mathematical models which can be physically realized before a circuit can be built which will behave like the desired filter.

These mathematical models are based on the ratio of two polynomials in S. In order to develop a feel for the shape of the amplitude versus frequency curve of various polynomials in S let us consider one such function. We will discuss a second order function in order to keep the description simple and still deal with all of the basic parameters of an Nth order function. The function we will consider is Equation 1.

$$(1) \qquad H(s) = \frac{H_o}{S^2 + \frac{\omega_o S}{Q} + \omega_o^2}$$

Figure 1 shows the magnitude of H(s) as a function of the real frequency ($S = j\omega$) W.

SECOND ORDER TRANSFER FUNCTION



In general you can select values for the parameters H_o, ω_o, and Q and you will have slightly different amplitude vs frequency plots. There is, however, one characteristic which will not change. That is, as the frequency ω gets far away from ω_o the roll off (slope of the curve) becomes -N*20dB/decade where N is the order of the filter. In this case N = 2 and the slope is -40dB/decade.

It is generally desired to have the gain at $\omega = 0$ (i.e. A) be set to 1. Let's look at the result of this criteria.

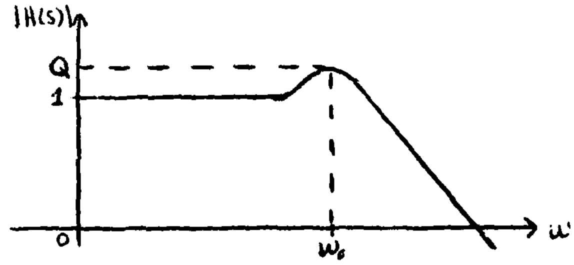
$$A = H(j0) = \frac{H_0}{(j0)^2 + \frac{\omega_0}{Q}(j0) + \omega_0^2} = \frac{H_0}{\omega_0^2}$$

$$A = 1 = \frac{H_0}{\omega_0^2}$$

Solve for H_0

$$H_0 = \omega_0^2$$

The result is the following general transfer function.



We call ω_0 the corner frequency. Let's find the magnitude of $H(s)$ at $\omega = \omega_0$

$$H(j\omega_0) = \frac{\omega_0^2}{(j\omega_0)^2 + \frac{\omega_0}{Q}(j\omega_0) + \omega_0^2} = \frac{\omega_0^2}{-\omega_0^2 + j\frac{\omega_0^2}{Q} + \omega_0^2}$$

$$H(j\omega_0) = \frac{\omega_0^2}{j\,\frac{\omega_0^2}{Q}} = Q \angle{-90^0}$$

Thus we see that the magnitude of the transfer function at $\omega_0 = Q$.

NOTICE: A potential hardware implementation problem can come about with high Q circuits.

Example:

Requirement: DC gain = 1 (i.e. $|H(j\omega)|$ at $\omega = 0$ is 1)

$\omega_o = 1$ KHz

$Q = 10$

Max. Input Voltage Swing = $\pm$ 10 volts

NOTICE: At ω_o(1 KHz) the filter gain will be 10 (Q).

This requires that the output be capable of swinging $\pm$ 100 volts.

You can get around this by making $H_o = \frac{\omega_o^2}{Q}$

In actuality a customer would not specify a Q for the filter but instead he will have a desired shape for the transfer function. The shape is controlled by the location of the poles and zeros of the transfer function.

The poles of the function are the values of S ($j\omega$) which make the denominator zero. The zeros of the function are the values of S($j\omega$) which make the numerator zero.

For the 2 pole low pass transfer function there are 2 poles (obviously).

$$S_1 = -\frac{\omega_o}{2Q} + j\omega_o\sqrt{1 - \frac{1}{4Q^2}}$$

$$S_2 = -\frac{\omega_o}{2Q} - j\omega_o\sqrt{1 - \frac{1}{4Q^2}}$$

and two zeros

$S = +\infty$

$S = -\infty$

By changing the location of the poles of this function you can change the shape of the function.

Let's say you wanted to stretch out the function i.e. move ω_o. To do this all you have to do is to multiply all the poles by a constant K where K is the ratio of the new corner frequency ω to the original frequency ω_o (i.e. $K = \frac{\omega}{\omega_o}$).

EXAMPLE: Given the general pole values

$$S_{P1} = -\alpha + jB$$

$$S_{P2} = -\alpha - jB$$

The frequency shifted poles would be

$$S^*_{P1} = -K\alpha + jKB$$

$$S^*_{P2} = -K\alpha - jKB$$

If you multiply out the poles and compare them with the original denominator of H(s) you will find that your scaled transfer function is now

$$H(\overset{*}{s}) = \frac{\omega_o^{\ 2}}{S^2 + \frac{K\omega_o S}{Q} + K^2\omega_o^{\ 2}}$$

Changing the poles by changing α and/or B will effect the shape of the transfer function. Let's look at the general transfer function and compare it to the expansion of the two general pole equations.

$$S_1 = -\alpha + jB \qquad S_2 = -\alpha - jB$$

multiplying these poles together

$$(S + \alpha - jB)\ (S + \alpha + jB) = S^2 + 2\alpha S + (\alpha^2 + B^2)$$

Comparing co-efficients of S with the original transfer function we see that

$$\omega_o = \sqrt{\alpha^2 + B^2}$$

and

$$Q = \frac{\omega_o}{2\alpha}$$

We can thus see that by changing α and/or B we can change ω_o and/or Q. It is not immediately clear from these equations how the shape of the function will change by a change in pole location.

There are equations which will tell us where to put the poles of a transfer function in order to achieve a certain shape. That is where Tchebyscheff, Butterworth and Bessel and numerous other mathematicians come into the act.

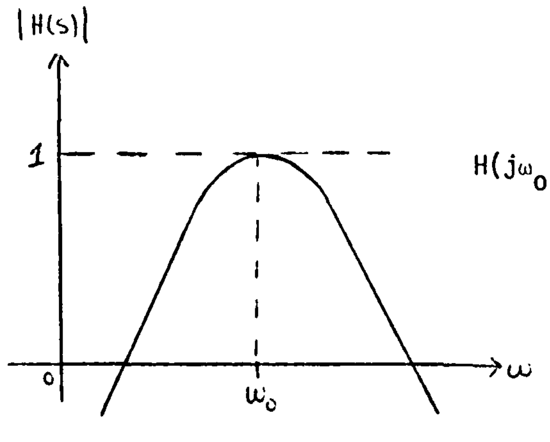
Before we get to them let's take a second to note what the location of the zeros of the functions do to its shape.

Lets move one of those zeros that is out at ∞ and bring it in to ω=o. The result is:

$$H(S) = \frac{SH_o}{S^2 + \frac{\omega_o}{Q} S + \omega_o^2}$$

Note at S=0 the function is 0
and at S=∞ the function is 0
and we can show by taking a derivative
that the function is a maximum at
$S=j\omega_o$

The result is a shape as shown below.



Let's solve for $|H(j\omega_o)|$

$$H(j\omega_o) = \frac{H_o\, j\omega_o}{(j\omega_o)^2 + \frac{\omega_o}{Q}(j\omega_o) + \omega_o^2}$$

$$= \frac{j\, H_o\, \omega_o}{j\, \frac{\omega_o^2}{Q}}$$

$$= \frac{H_o \omega_o\, Q}{\omega_o^2}$$

$$\left| H(j\omega_o) \right| = \frac{H_o Q}{\omega_o}$$

If we set the gain at the center = 1
we have

$$H_o = \frac{\omega_o}{Q}$$

We thus have the general transfer function for a second order Bandpass filter

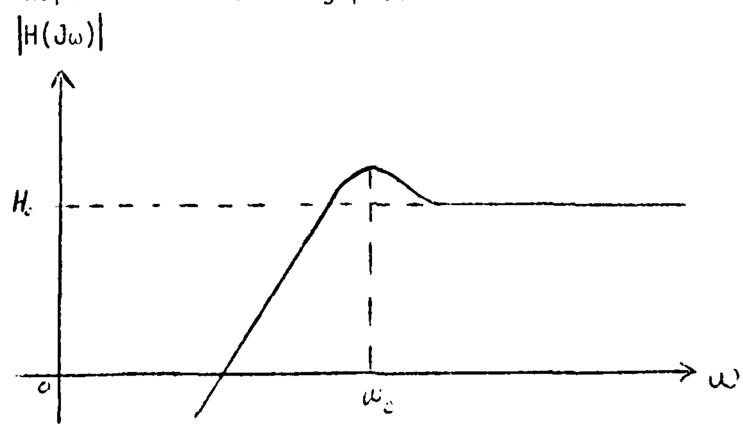
$$H(S) = \frac{H_o\, S}{S^2 + \frac{\omega_o}{Q} S + \omega_o^2}$$

If we move both zeros from ∞ to 0 we have a transfer function as follows

$$H(S) = \frac{H_o S^2}{S^2 + \frac{\omega_o}{Q} S + \omega_o^2}$$

Notice: The value at S=0 is 0
The value at $S=\infty$ is H_o
(divide numerator and denominator by S^2 and let $S \to \infty$)

The shape of this is a highpass function



Thus by moving the zeros of the transfer function we can generate lowpass, highpass, bandpass or even band reject filters if we place both zeros at the frequency we want to reject.

Since we can derive any type filter depending only on zero locations we will now talk only about the lowpass filter poles realizing that the same general characteristics will hold true for the other filter types.

One desirable transfer function shape is one which is maximally flat in the passband.

A Butterworth filter has this characteristic. The pole locations for a Butterworth filter are given by the following equations.

Butterworth Formulas

N = Degree of filter = # of poles

Note: All filters in Butterworth family have 3dB at ω=1.

Attenuation ratio

$$A = \sqrt{1 + \omega^{2N}}$$

$$\alpha = 20 \text{ Log } A$$

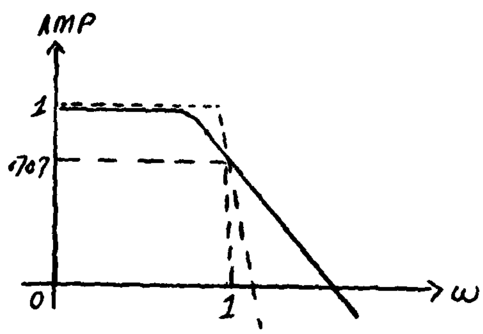
Pole location

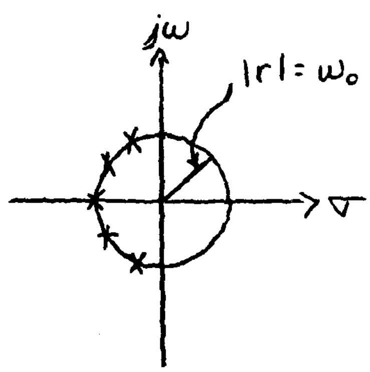
Poles from K=1 to N

$$P_K = -\text{Sin } \Theta_K + j \text{ Cos } \Theta_K$$

$$\Theta_K = \frac{(2K-1)\pi}{2N} = (2K-1) \frac{90^\circ}{N}$$

Note P_{N-K+1} is Complex conjugate of P_K





When you think of a filter most often you think of the ideal case with an infinitely steep roll off. Because of certain physical constraints we cannot achieve $-\infty$dB/decade but a Tchebyscheff type filter tends to give maximum roll off with a small sacrifice in the flatness in the passband.

TCHEBYSCHEFF -- Very fast initial roll off
final roll off -N*20dB/decade
-N*6dB/octave

Attenuation Ratio

$$A(\omega) = \sqrt{1 + \varepsilon^2 C^2_N(\omega)}$$

Given α_o in dB ← Passband Ripple

$$\varepsilon = \sqrt{10^{.1\alpha_o} - 1}$$

$C_N(\omega)$ - Tchebyscheff Polynomials
for $\omega>1$ i.e. stop band

* $C_N(\omega) = \text{Cosh}\ (N\ \text{Cosh}^{-1}\ \omega)$

or solving for ω

$$\omega = \text{Cosh}\left(\frac{1}{N}\ \text{Cosh}^{-1}\ C_N(\omega)\right)$$

for $\omega<1$ in passband

* $C_N(\omega) = \text{Cos}\ (N\ \text{Cos}^{-1}\ \omega)$

or

$$\omega = \text{Cos}\left(\frac{1}{N}\ \text{Cos}^{-1}\ C_N(\omega)\right)$$

Tchebyscheff Polynomials $C_N(\omega)$

N	$C_N(\omega)$
1	ω
2	$2\omega^2-1$
3	$4\omega^3-3\omega$
4	$8\omega^4-8\omega^2+1$
5	$16\omega^5-20\omega^3+5\omega$

Tchebyscheff Poles

for $K=1 \rightarrow N$ poles

$$P_K = -\sin\Theta_K \sinh\beta + j\cos\Theta_K \cosh\beta$$

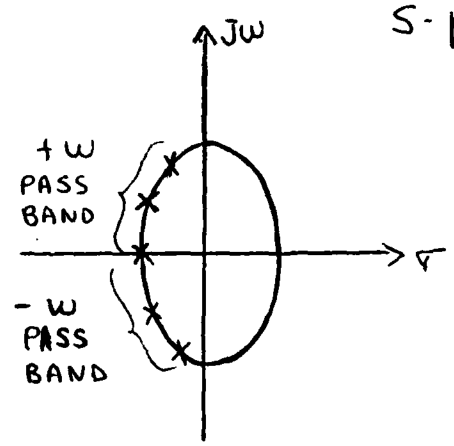
where $\Theta_K = (2K-1)\frac{\pi}{2N} = (2K-1)\frac{90°}{N}$

$$\beta = \frac{1}{N}\sinh^{-1}\left(\frac{1}{e}\right)$$

Note: P_{N-K+1} & P_K are complex conjugates
When N is odd there is one real pole

$$P_{\frac{N+1}{2}} = -\sinh\beta$$

S-plane plot



Refresher Equations

$\sinh X = \cancel{\sqrt{e^X - e^{-X}}} = (e^x - e^{-x})/2$

$\cosh X = \cancel{\sqrt{e^X + e^{-X}}} = (e^x + e^{-x})/2$

$\sinh^{-1} X = \ln\left(X + \sqrt{X^2 + 1}\right)$

$\cosh^{-1} X = \ln\left(X + \sqrt{X^2 - 1}\right)$

Useful identity to generate table

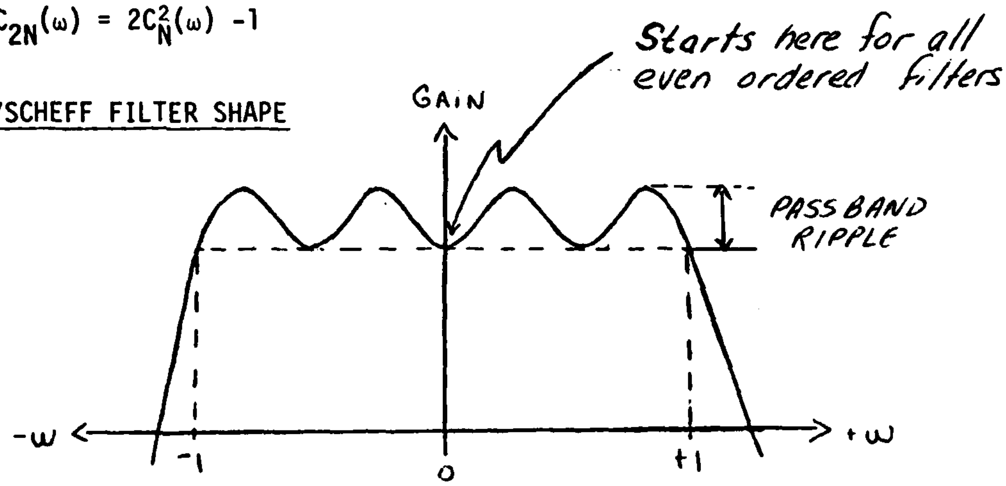
$$C_N(\omega) = 2\omega\, C_{N-1}(\omega) - C_{N-2}(\omega)$$

Useful identity when evaluating $A(\omega)$

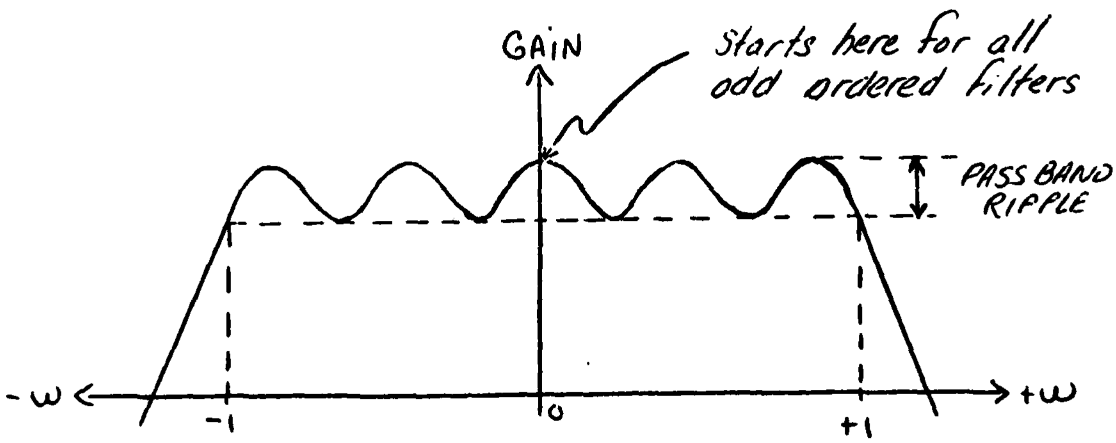
$$C_N^2(\omega) = \sqrt{C_{2N}(\omega) + 1}$$

$$C_{2N}(\omega) = 2C_N^2(\omega) - 1$$

TCHEBYSCHEFF FILTER SHAPE



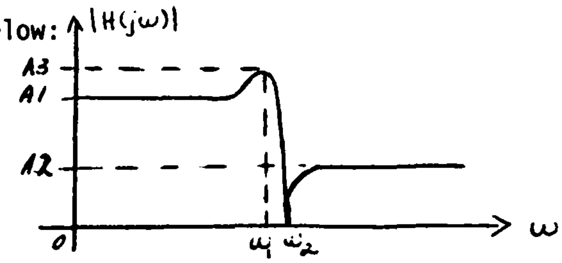
N=4
(4 peaks)



N=5
(5 peaks)

A third filter type which sacrifices attenuation in the stopband for rapid roll off is the Elliptic filter. Its transfer function is formed by placing the zeros near the poles of the function rather than at the two extremes of the frequency spectrum. A typical transfer function is shown below:

$$H(S) = \frac{H_o\,(S^2 + a)}{S^2 + \frac{\omega_o}{Q} S + \omega_o^2}$$



A very steep roll off can be acheived with this type filter as long as the resultant floor of attenuation A2 can be tolerated. Notice that as you bring the zero in from ∞ the roll off increases and so does the floor. When the zero is located at the corner frequency (assuming originally we had a Butterworth transfer function) the result is a Band Reject filter.

Let's now consider the class of filters that attempt to satisfy the time domain condition of pure time delay. Mathematically the transfer function must be as follows:

$e\,(t) \rightarrow$ Filter $\rightarrow e(t-\tau_o)$

INPUT OUTPUT

Stating this in equation form

$$e_{out}\,(t) = e_{in}\,(t - \tau_o)$$

We must now see what this equation looks like in the frequency domain (S).

Lets take the laplace transform of both sides of this equation.

Remember:

$$\mathcal{L}\left[f(t)\right] = \int_0^\infty f(t)e^{-st}dt = F(S)$$

Starting with $e_{out}(t)$

$$\mathcal{L}\left[e_{out}(t)\right] = \int_0^\infty e_{out}(t)\, e^{-st}dt = E_o(S)$$

Now with the Rt side

$$\mathcal{L}\left[e_{in}(t - \tau_o)\right] = \int_0^\infty e_{in}(t - \tau_o)\, e^{-st}dt$$

make a variable substitution $t^* = t - \tau_o$

then $dt^* = dt$ & $t = t^* + \tau_o$

$$= \int_o^\infty e_{in}(t^*) e^{-s(t^* + \tau_o)} dt^*$$

$$= \int_o^\infty e_{in}(t^*) e^{-st^*} . e^{-S\tau_o} dt^*$$

Note $e^{-S\tau_o}$ is a constant over the integral so

$$= e^{-S\tau_o} \int_o^\infty e_{in}(t^*) e^{-st^*} dt^*$$ let $t = t^*$ and we see

$$\mathcal{L}\left[e_{in}(t-\tau_o)\right] = e^{-S\tau_o} E_{in}(S)$$

Now forming the original equation we have

$$E_{out}(S) = e^{-S\tau_o} E_{in}(S)$$

$$\frac{E_{out}(S)}{E_{in}(S)} = e^{-S\tau_o}$$

$$H(S) = e^{-S\tau_o}$$

H(S) is the frequency domain equivalent of the original time domain transfer function.

There are two important characteristics of H(S)*

1) Its amplitude is 1 i.e. independent of frequency
2) Its phase is proportional to frequency with proportionality constant τ_o.

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* Recall $S = j\omega$ Therefore $H(j\omega) = e^{-j\omega\tau_o}$; from Eulers identity $e^{-j\omega\tau_o} = \cos\omega\tau_o - j \sin \omega \tau_o$. The amplitude of this complex number is $\sqrt{(\cos \omega \tau_o)^2 + (\sin \omega \tau_o)^2} = \sqrt{1} = 1$. Its phase angle is

$$\text{arc tan } \frac{-\sin\omega\tau_o}{\cos \omega\tau_o} = \text{arc tan } (-\tan \omega\tau_o) = -\omega\tau_o$$

A Bessel type filter approximates this desired response for input signals within a specified passband.

$$H(S) = \frac{H_o}{B_n(S)}$$

where

$$B_n(S) = a_0 + a_1 S + \ldots\ldots + a_{n-1} S^{n-1} + S^n$$

where the coefficients a_i are computed from the following equation

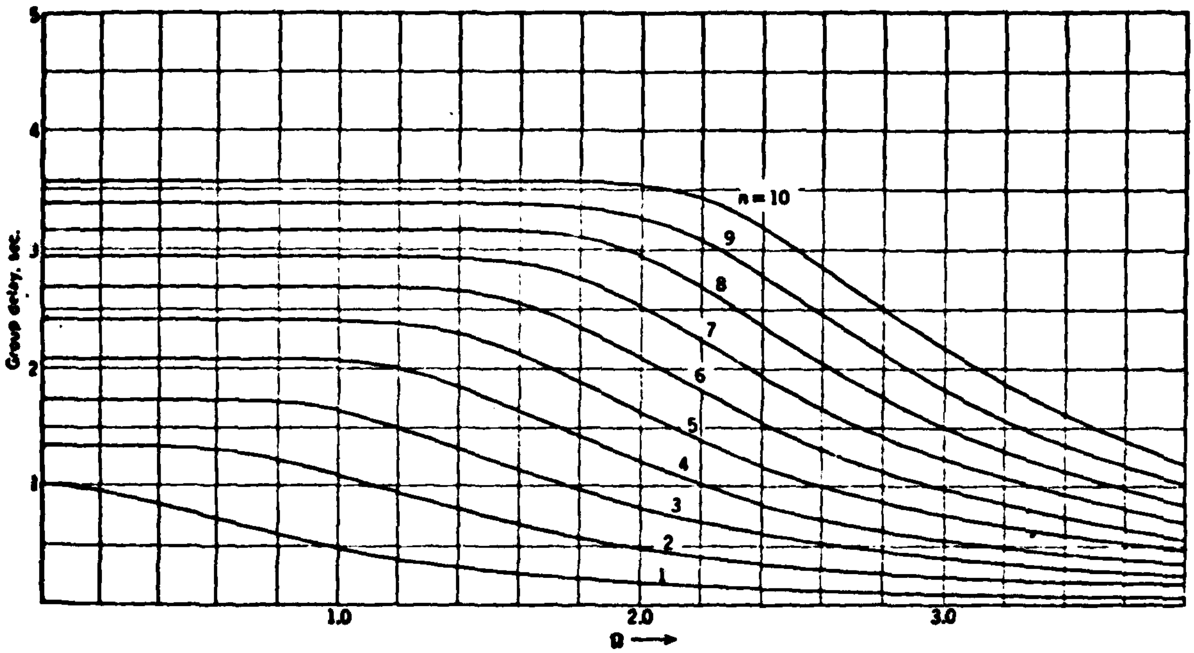
$$a_i = \frac{(2n-i)!}{i!(n-i)!} \quad (2\tau_0)^{i-n} \text{ for } i=0,1,2,\ldots n-1$$

n = the order of the filter

τ_0 = delay time

The corner of the Bessel filter is generally specified as $\omega = \frac{1}{\tau_0}$. F.D.I. specifies it as the -3dB point in order to keep all of our specifications consistent. Note the F.D.I. corner is just $K \cdot \frac{1}{\tau_0}$. A good measure of the linearity of the phase across the passband is the term called the Group Delay. The Group Delay is the derivative of the phase. If the group delay is a constant then the phase will be linear. The Bessel coefficients are derived by establishing a criteria of maximally flat group delay.

Over the passband the phase increases linearly from 0 to $n\pi/4$ radians at the cut off frequency.



Group-delay characteristics for maximally flat delay (Bessel) filters.

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NOTES

IMPLEMENTATION OF THE FILTER THEORY

Dr. Robert W. Steer

President
Frequency Devices, Inc.

IMPLEMENTATION OF FILTER THEORY

Topical Outline

1. Two types of filter synthesis commonly used to realize filter polynominal functions

   A. Passive RLC ladder synthesis techniques

   B. Active RC and Amplifier techniques

2. Design techniques that simply design procedures for passive and active filters

   A. Frequency scaling

   B. Impedance scaling

   C. Lowpass to Bandpass and other transformation

   $$\text{LP - BP} \qquad S \Longrightarrow \frac{S^2 + \omega_o^{\;2}}{S}$$

   gives Geometric Symmetry

   $$\text{LP - HP} \qquad S \longrightarrow \frac{1}{S}$$

3. Design Lowpass filters and transform up to ω_o to get Bandpass filters

4. Circuit Configurations

   A. Stagger tuned

   B. Leap frog

   C. Primary resonator block

   D. Analog computer circuit

5. Some common active filter circuits

   A. Sallen and key resonator

   B. Multiple feedback

   C. State variable - dual integrator

6. Sensitivities of various circuit configurations

7. Practical limitations for active filters

   A. Q is limited by available gain

   B. DC offset and noise of amplifiers

   C. Frequency limitations (QA product)

NOTES

FILTER SPECIFICATIONS

Steven M. Ruscio

Sales/Application Engineer
Frequency Devices, Inc.

ABSTRACT

The Active Filter is now available in what is commonly referred to as component form. It is an item the system designer may order off the shelf in any of a variety of configurations. As is the case with any component there are performance limitations to be considered and benefits to be derived by proper selection and application.

The presentation will outline system factors which relate directly to filter selection and review the state of the art from a performance point of view. Individual filter specifications will be discussed. The importance and method defining the filtering requirements before selecting a filter type will be emphasized.

FILTER SPECIFICATIONS

(typical @ 25°C and ±15V unless otherwise specified)

	REQUIRED	DESIRED	TYPICAL
Center/Cutoff Frequency			.001 to 50kHz
Tuning Range			20:1 to 1000:1
Drift			50 to 100 ppm/°C
Tolerance (Fixed Frequency)			to ±.1%
Passband Gain			0 to 60dB
Amplitude Match			Match is a Function
Frequency Range			of the sensitivity &
Phase Match			initial tolerance of
Frequency Range			the design to be used
Input			
Impedance			10^3 to $10^9 \Omega$
Source			typ <600Ω
Bias Current			1-10nA
Voltage Range			±10V at ±15V V_s
Safe Input Voltage			V_s
Output			
Full Power Response			50kHz
Noise			10 to 50μV
Rated Output			2-5mA
Offset Voltage			1mV-10mV
Drift			10 to 20μV/°C
Resistance			1Ω to 10Ω
Power Requirements			
Quiescent Current (mA)			Depends on Device
Voltage, Rated Specs (V)			±15V
Voltage, Operating			±5 to ±18V
Temperature (°C)			
Commercial Operating			0 to 70
Storage			-25 to +85 ~~(-55 to +125)~~
Military Operating			-25 to +85 (-55 TO +125)
Storage			-65 to 150
Standard Mechanical Configurations			
Length			1.0" to 3.00"
Width			1.0" to 2.00"
Height			0.4" to 1.00"

DEFINITION OF VARIABLES:

FIGURE 1 & 2

NOISE:	Maximum acceptable level should be indicated.
RIPPLE:	Acceptable variation in passband gain
GAIN:	Desired passband gain @ f_o

FIGURE 1:

f_{R1}, f_{R2} :	Limits of ripple band
f_1, f_2 :	-3dB Attenuation frequencies
$f_o = \sqrt{f_1 f_2}$:	Center frequency
A_{min} :	Minimum stopband attenuation, if monotonic roll off is required this must be specified.
f_{A1}, f_{A2} :	*Frequencies by which a given* attenuation A_{min} must be achieved.

FIGURE 2:

f_{R1}, f_{R2} :	Limits of the ripple band
f_{max} :	Maximum frequency at which full power response is required
$f_o = \sqrt{f_1 f_2}$	Center frequency
f_{A1}, f_{A2} :	Limits of Attenuation Band (stopband)
A_{min} :	Required Stopband Attenuation to be maintained $f > f_{A1}$ and/or $f < f_{A2}$

For Lowpass:

$f_1 = f_c$ = -3dB or cutoff frequency

A_{min}, f_{A1}, as described above

For Highpass:

$f_2 = f_c$ = -3dB or cutoff frequency

A_{min}, f_{A2} as described above

FIGURE 1: Bandpass

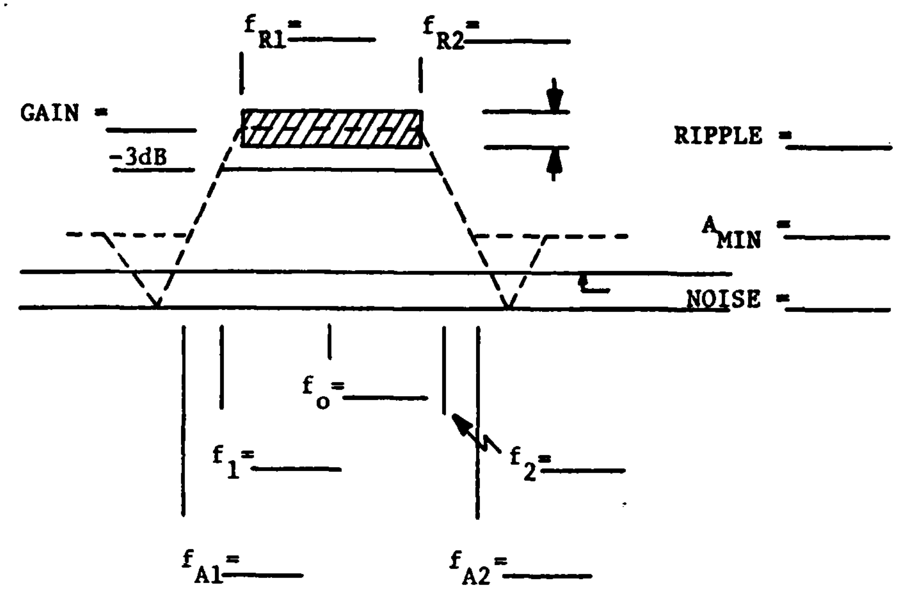
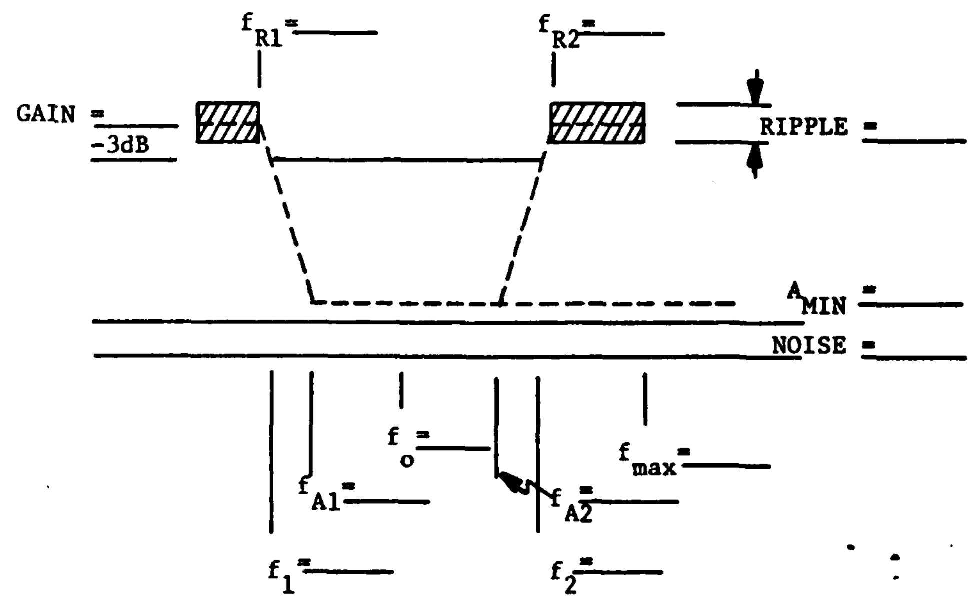


FIGURE 2: Band Reject, Highpass, Lowpass



FILTER SPECIFICATIONS

(typical @ 25°C and ±15V unless otherwise specified)

	REQUIRED	DESIRED	TYPICAL
Center/Cutoff Frequency			.001 to 50kHz
Tuning Range			20:1 to 1000:1
Drift			50 to 100 ppm/°C
Tolerance (Fixed Frequency)			to ±.1%
Passband Gain			0 to 60dB
Amplitude Match			Match is a Function
Frequency Range			of the sensitivity &
Phase Match			initial tolerance of
Frequency Range			the design to be used
Input			
Impedance			10^3 to $10^9\Omega$
Source			typ <600Ω
Bias Current			1-10nA
Voltage Range			±10V at ±15V V_s
Safe Input Voltage			V_s
Output			
Full Power Response			50kHz
Noise			10 to 50μV
Rated Output			2-5mA
Offset Voltage			1mV-10mV
Drift			10 to 20μV/°C
Resistance			1Ω to 10Ω
Power Requirements			
Quiescent Current (mA)			Depends on Device
Voltage, Rated Specs (V)			±15V
Voltage, Operating			±5 to ±18V
Temperature (°C)			
Commercial Operating			0 to 70
Storage			-25 to +85 ~~(-55 to +125)~~
Military Operating			-25 to +85 (-55 to +1
Storage			-65 to 150
Standard Mechanical Configurations			
Length			1.0" to 3.00"
Width			1.0" to 2.00"
Height			0.4" to 1.00"

DEFINITION OF VARIABLES:

FIGURE 1 & 2

NOISE:	Maximum acceptable level should be indicated.
RIPPLE:	Acceptable variation in passband gain
GAIN:	Desired passbard gain @ f_o

FIGURE 1:

f_{R1}, f_{R2} :	Limits of ripple band
f_1, f_2 :	-3dB Attenuation frequencies
$f_o = \sqrt{f_1 f_2}$:	Center frequency
A_{min} :	Minimum stopband attenuation, if monotonic roll off is required this must be specified.
f_{A1}, f_{A2} :	Frequencies by which a given attenuation A_{min} must be achieved.

FIGURE 2:

f_{R1}, f_{R2} :	Limits of the ripple band
f_{max} :	Maximum frequency at which full power response is required
$f_o = \sqrt{f_1 f_2}$	Center frequency
f_{A1}, f_{A2} :	Limits of Attenuation Band (stopband)
A_{min} :	Required Stopband Attenuation to be maintained $f > f_{A1}$ and/or $f < f_{A2}$

For Lowpass:

$f_1 = f_c$ = -3dB or cutoff frequency

A_{min}, f_{A1}, as described above

For Highpass:

$f_2 = f_c$ = -3dB or cutoff frequency

A_{min}, f_{A2} as described above

FIGURE 1: Bandpass

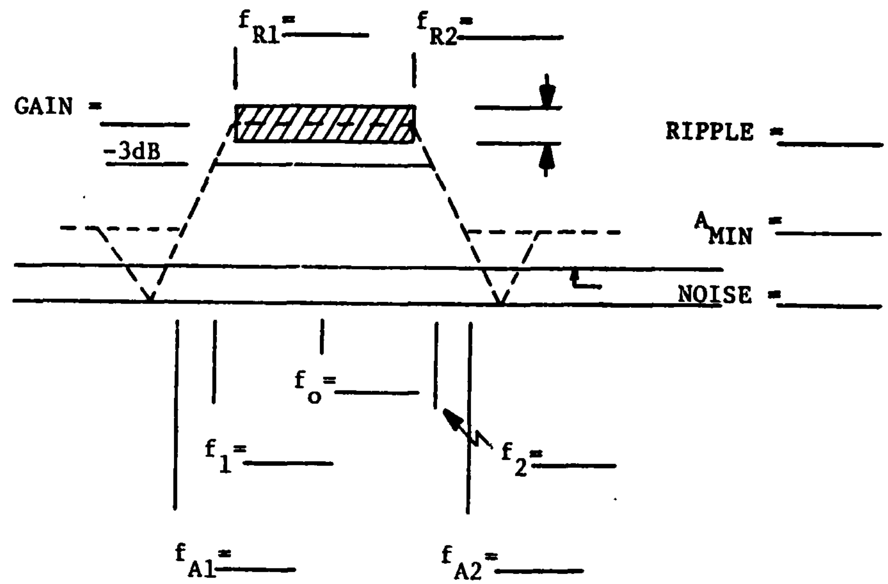
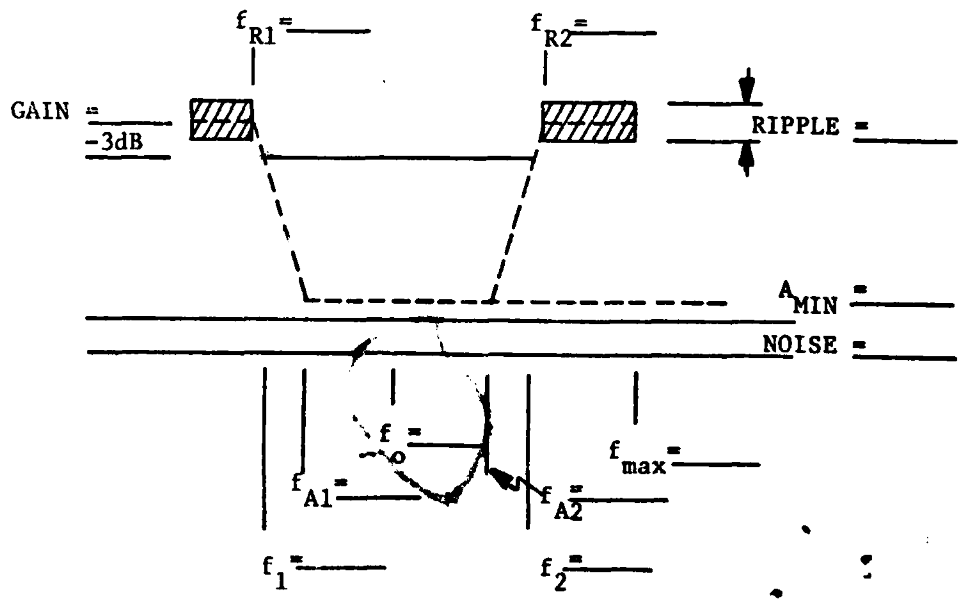


FIGURE 2: Band Reject, Highpass, Lowpass



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