Abstract:
"All-Pass Phase-Filtering in Analog Active Band-Pass Circuits"

We present an experimental study
of original novel design Analog Phase-Filters
emphasizing the "ALL-PASS" filter
and comprised of :

(a) Parallel-Channel-Differential Filters
(b) All-Pass Filters
(c) Differentially derived Dual-Notch filter

This project provides near DSP narrow filtering characteristics,
narrow band-width,   without computer processing.

This project's modules include :

(a) Pre-Amp with OV limiter
(b) Triad-Roofing-Differential Filters
(c) Active Log-Limiter
(d) All-Pass Filters,
(e) Phase-Filtered Non-Resonant Dual-Notch stage
(f) Delyannis Narrow band-pass filters,

This project has applications which include

(a) Narrow Passband Audio Filters for Radio Operations
(b) Laboratory Analog Instrument filters "stand-alone".

This "AFC" project was developed to explore **All-Pass Filters**
and is the contunuation of the author's previous project "AFX"
and produced via the PartSIM.com browser based ngSpice Simulator.
See "RGN" presentation of "AFX" at :
https://www.researchgate.net/post/Are-there-any-Analog-Active-Audio-Filters-that-match-any-Digital-Signal-Processing-filters

This paper presents an experimental project
produced via the PartSIM.com browser based ngSPICE Simulator,
in our lab as posted on our website, and as posted on ResearchGate.Net,
during 2014 thru 2021.

Tutorial analysis for modules in "AFC" are written in our previous "AFX" project.

All concepts presented herein are common to Active BandPass Filters as presented in academic course-work assigned to Electronic Engineering students.

ONLY the SYNTHESIS of common ideas into radically new approaches is different.

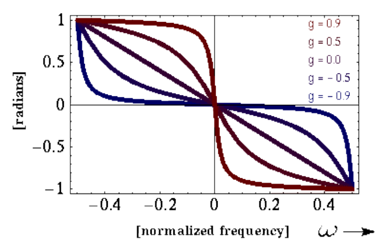
Some All-Pass Filter Theory
from common Electrical Engineering textbooks :

Defined: An *all-pass filter*
is a signal processing *filter* that *passes all* frequencies equally in gain, but changes the phase relationship among various frequencies. Wiki

To transmit a signal with minimum phase distortion, an all-pass filter must have a constant group delay across the specified frequency band. The group delay is the time by which the all pass filter delays each frequency within that band.

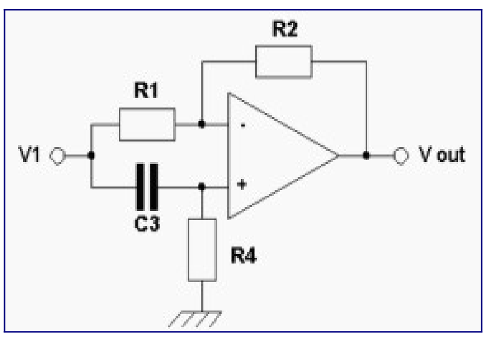
First-Order Allpass Phase Response $(\text{let } g = re^{j\theta})$

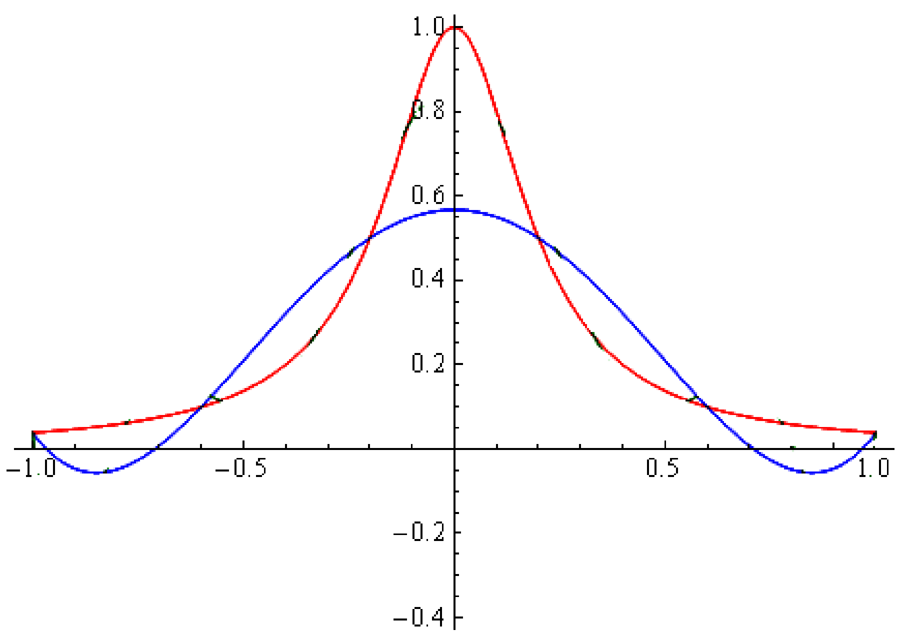
$$\Theta(\omega) \triangleq \angle H\left(e^{j\omega}\right) = -\omega - 2\tan^{-1}\left[\frac{r\sin(\omega-\theta)}{1 - r\cos(\omega-\theta)}\right]$$



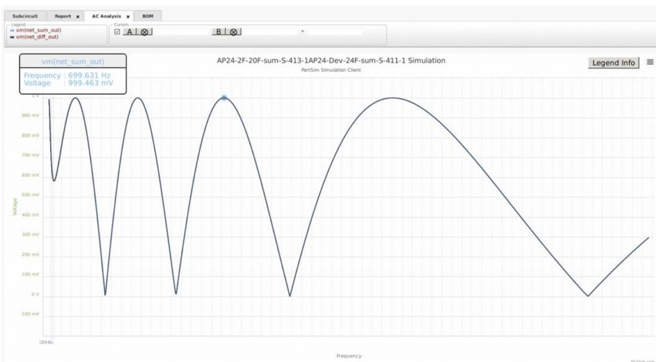
All-Pass Filter via LaPlace Transform : H(s) = [(n1(s)/d1(s)] + [n2(s)/d2(s)]

**“The AFX / AFC ”Differential Dual Notch”
can be developed
at the points
where these
two waveforms
cross-over.**

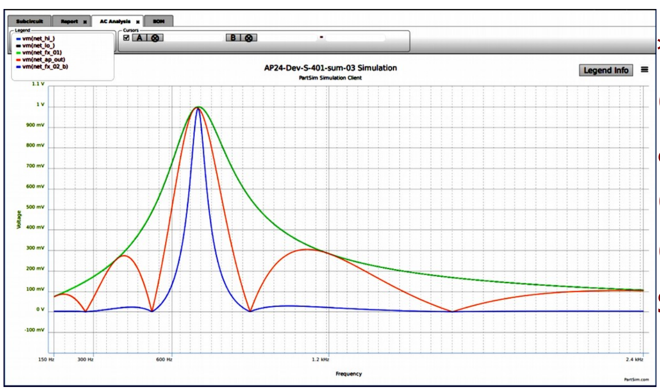




*** We have developed
working **Dual-Notch Band-Pass** circuits
which emphasize the **All-Pass Filter** elements
in “LoPass and HiPass combined“ circuit configurations .



*** Raw All-Pass Bode***
All-Pass Filters in a cascade network.
(example of All-Pass Bode
as used in the AFC circuits)



(example of “AFC” circuit.
Major Vout signals)

*** Bode Plot ***
(1) Input signals (**green)**
are driven through All-Pass .
(2) All-Pass signals (**red**)
(3) are driven through
several stages of
Resonant MFB filters
(**blue** trace).

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*****   AFC Project Begins   ***************************

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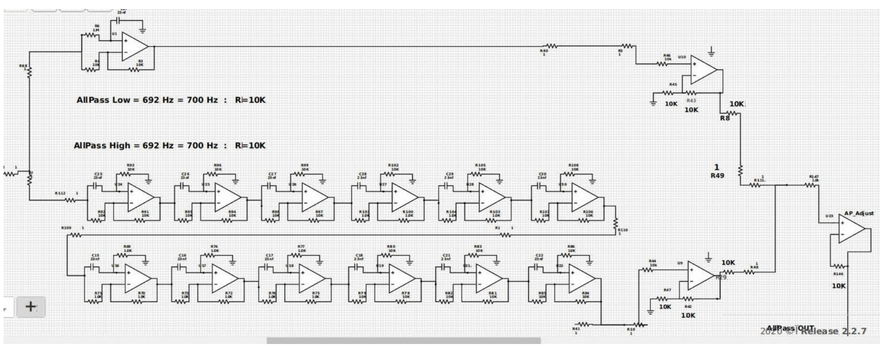
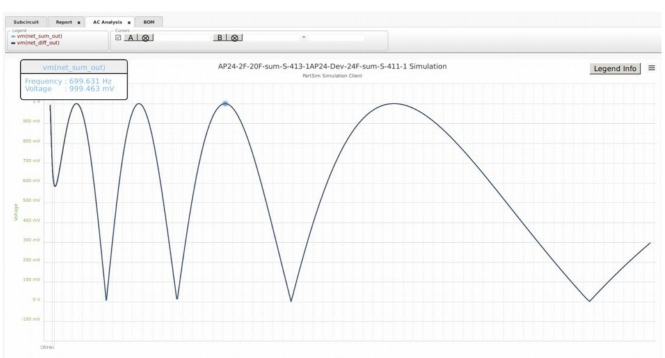
The "All-Pass" Band-Pass Filter

We examine this All-Pass Phase-Filter circuit :



Circuit #1 : AFC lattice core   1 Lo-Pass and 12 Hi-Pass

The All-Pass filter has **frequency responses**
which **must be zero at w=0 and at w=pi**." wiki



All-Pass Filter via LaPlace Transform : H(s) = [(n1(s)/d1(s)] + [n2(s)/d2(s)]

The principle is ... adding signals with the same amplitude and different phase. For a phase shift of 180 ° (+ 360 °; + 720 °; ... etc.) we obtain a minimum of transmission.   (Dr. Josef Puncochar, VŠB-Technical University of Ostrava)

Miller theorem (applied to Q1's rbe resistance) applies to the OPA R(input) for each OpAmp in Schematic. These Standard OPA are R(in=1M Ohm).

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This paper presents the experimental development of
two unique original circuits emphasizing All-Pass-Filters .

Presented here are
the CURRENTLY PREFERRED versions of this type of Band-Pass Filter.
Certainly we have developed other topologies
but these are two good examples.

Circuit #1 is a narrow Band-Pass Audio Filter

AFC-1RL-AP1-AP12-F2Q

Most simple of this set
which includes filter designs with these characteristics:
... Based on the All-Pass Filter group
... All-Pass Filter stages for mulitple w/0 signal lobes
... f(0) Revised to f(600) in order to align the ‘zero’ frequencies closer together.
f = 600 Hz R = 12.1 K Ohm. C = 22 nF .
... Summing **Phase-Filter** for Dual-Notch at +/- 200 Hz of f(0)
... driving Final Hi-Q MFB filters

Circuit #2 is a narrow Band-Pass Audio Filter

AFC_3R-2F-8A-Dif

most complex of this set : includes filter designs with these characteristics:
... Based on the Basic ‘AFX’ design
... All-Pass Filter stages for mulitple w/0 signal lobes
... f(0) Revised to f(677) in order to align the ‘zero’ frequencies closer together.
f = 677 Hz R = 11.1 K Ohm. C = 22 nF .
... Differential **Phase-Filter** for Dual-Notch at +/- 200 Hz of f(0)
... driving Final Hi-Q MFB filters

Note:
Calcs for the All-Pass components at f(0) : f = 1 / (2 pi) (R2) (C3)
Original design was for all f() to be 700 Hz. : R = 10 K Ohm. C = 23 nF .
Circuit #1: Revised f(600) in order to align the ‘zero’ frequencies closer together.
f = 600 Hz R = 12.1 K Ohm. C = 22 nF .
Circuit #2: Revised f(677) in order to align the ‘zero’ frequencies closer together.
f = 677 Hz R = 11.1 K Ohm. C = 22 nF .

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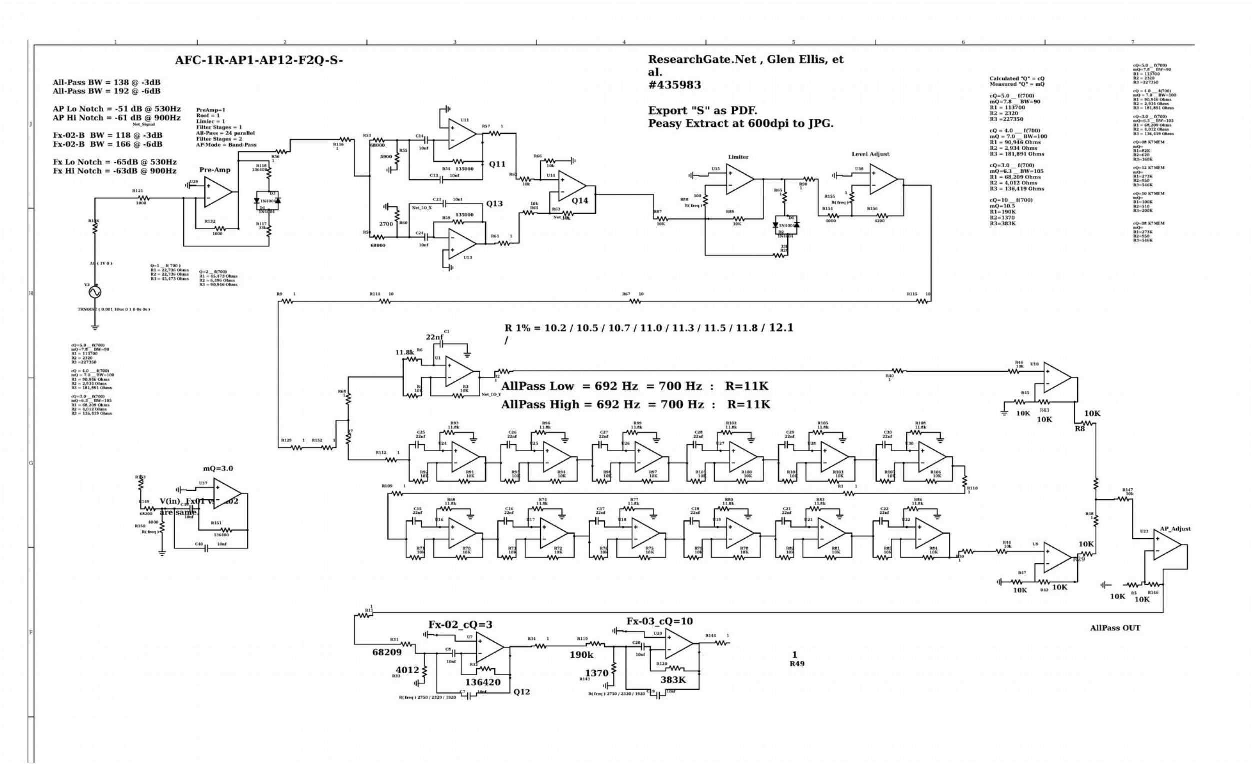
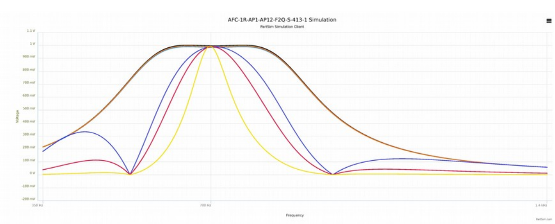
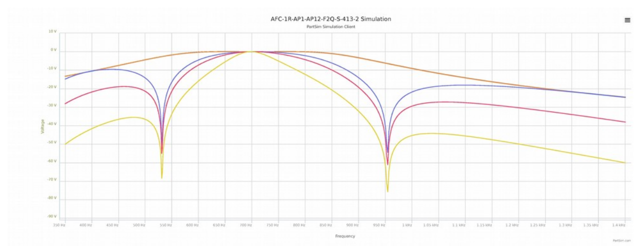
Circuit #1 :   AFC-1RL-AP1-AP12-F2Q-S



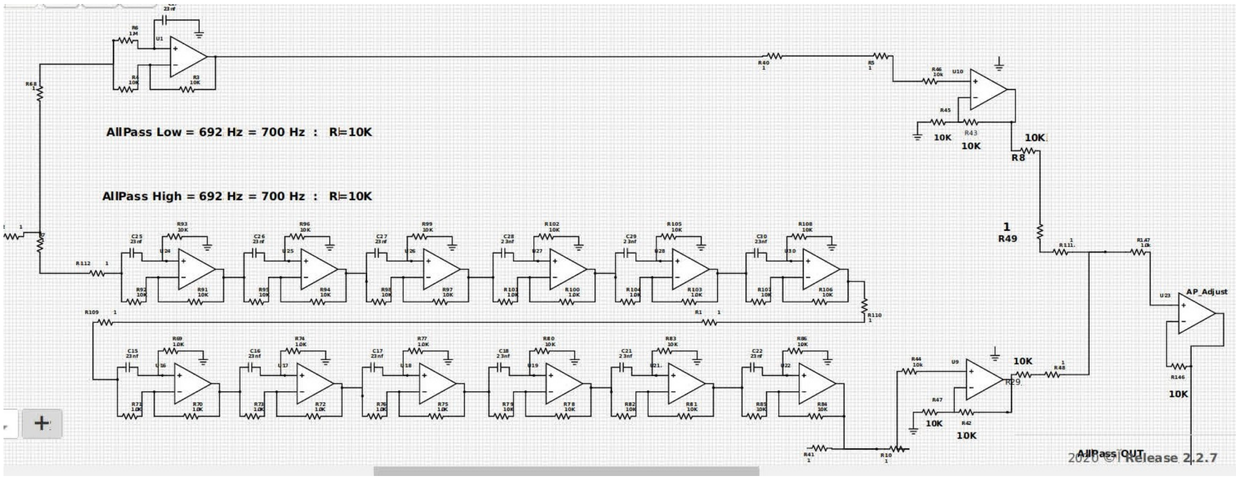




...

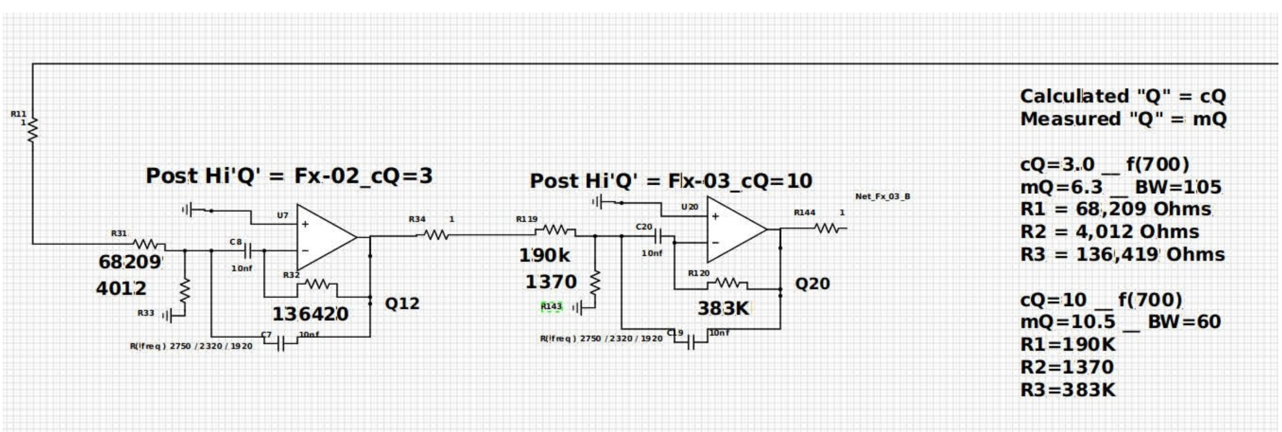
Detail of Circuit #1 : example :   All-Pass general schematic

All-Pass module generates Dual-Notches :
The All-Pass-Lo and the All-Pass-Hi are Buffered and Summed
to generate the Narrow Dual-Notch Pass-Band. .



Dual Notch Bode Plot is generated from this Phase-Filter stage,
via a Voltage-Summing final stage.

Below: example Detail of shaping the All-Pass V(out) :
Hi ‘Q’ Filters sharpen the Band-Pass inside of the Dual-Notches :
via Resonant Hi-Q MFB Post-Filters
which produce a very Narrow Band-Pass.



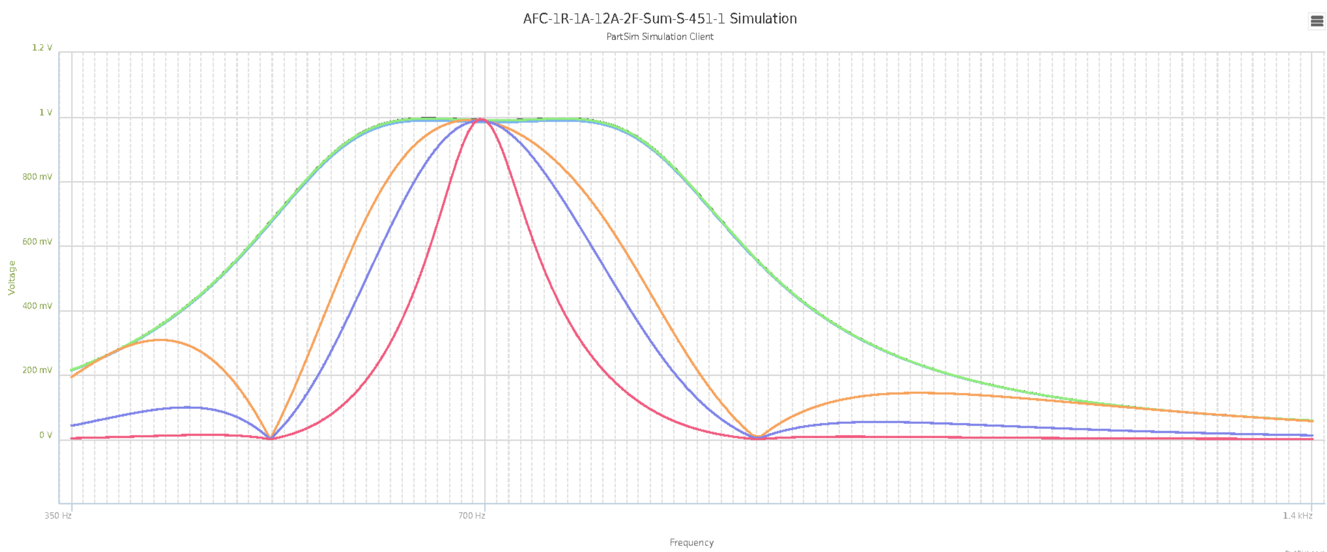
Bode Plot :   Circuit #1

Bode Plot :   f(0) is 700 Hz.

The flat-topped GREEN trace is the "Roofing-Filter"
Notice the -BW=350Hz   and
the 27-dB-per-Octave attenuation of side-band signals.

***** ORANGE trace is the Dual-Notches around f(0)**
***** BLUE trace is the first MFB filter, with 150 Hz Band-Pass.**
***** RED trace is the final MFB filter, with 35 Hz Band-Pass.**

***** Notice: Fx-02 Q=3 and Fx-03 Q=10**
are for narrow Band-Pass, inside ORANGE Dual-Notches.
***** ORANGE trace is AllPass circuit V(output)**

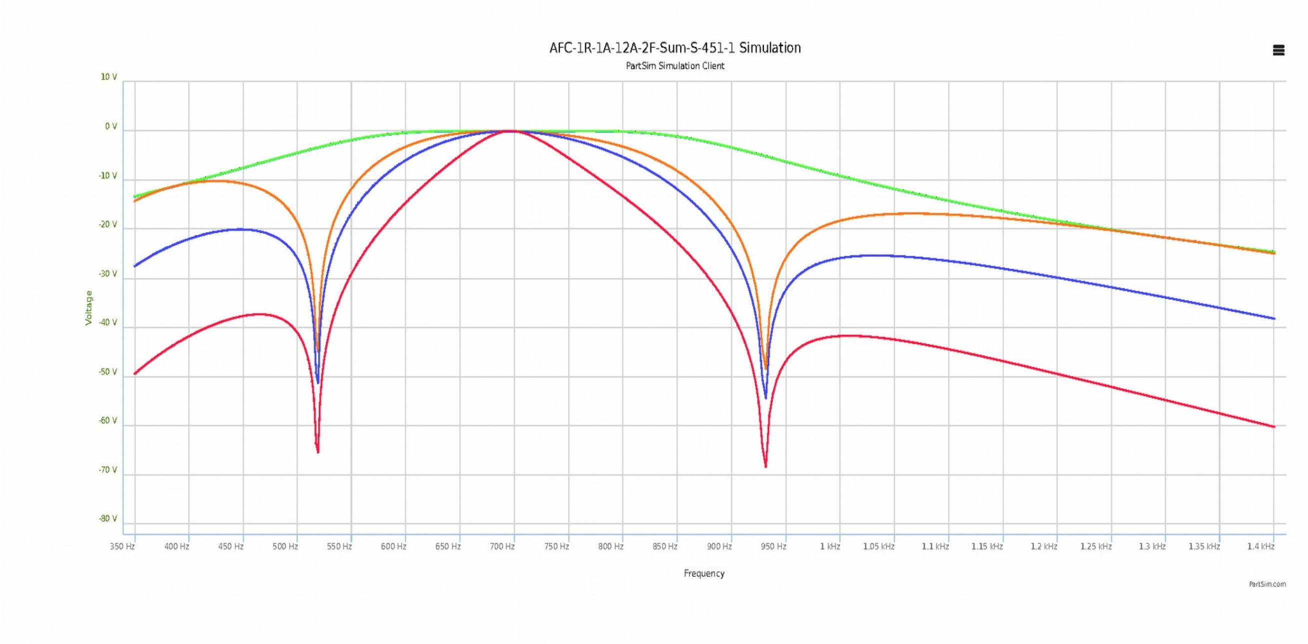


***** Scale : 700mV = -3dB ; 500mV = -6dB**
***** Notch High 900 Hz = -53 dB**

Notch BW = 100 Hz @   -3 dB
Notch BW = 360 Hz @ -53 dB

Variance = 5.2 Hz sideband expansion per -dB attenuation which is analogous to the DSP “brick-wall” effect.

Magnitude Plot :   Circuit #1



GREEN trace is Roof Filter Output plot.
ORANGE trace is basic AllPass Output plot.

***** RED trace is final Q=20 result**
***** f(0) Shifts are confined within the ORANGE AllPass Output plot.**
(aprox DSP “**Brick-Wall**”)

***** Magnitude Scale readings : V = -dB**
*** Scale : 700mV = -3dB ;   500mV = -6dB

Notch BW = 100 Hz @   -3 dB
Notch BW = 360 Hz @ -53 dB
Variance = 5.2 Hz sideband expansion per -dB attenuation which is analogous to the DSP “Brick-Wall” effect.

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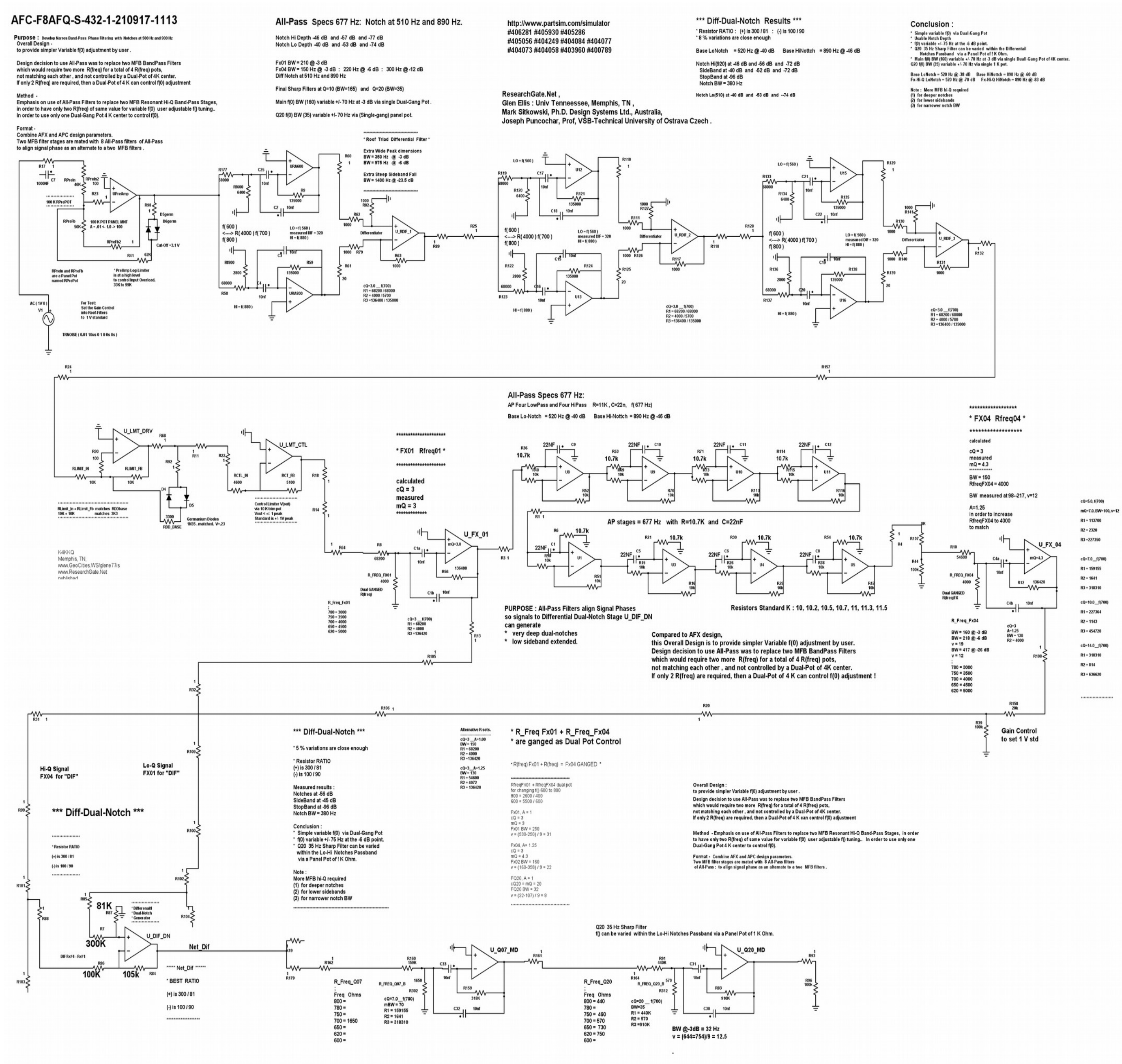
Circuit #2 : AFC_3R-2F-8A-Dif

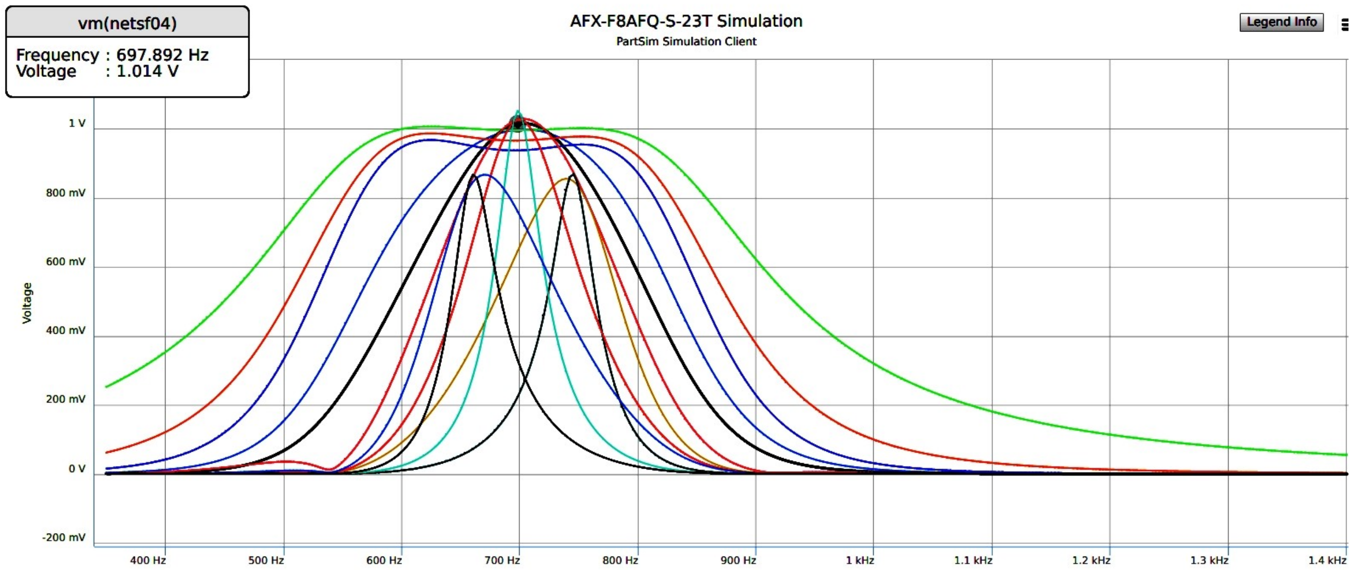
more complex All-Pass Band-Pass circuit combining AFX and AFC concepts.



***** Two Variable MFB Filters with 8 All-Pass stages inserted in the middle**
Result One Dual-Pot can control f(0) +/- 75 Hz, reading at the ~~3dB level.~~
+/ 75 Hz is equivalent to one piano note.

***** ONE f(Lo) ~ 677 Hz and Twelve f(Hi) ~ 677 Hz .**
***** 677 Hz freq is based on R=10.7K , C=22nF.**

Bode Plot : Circuit #2 ... from a test circuit



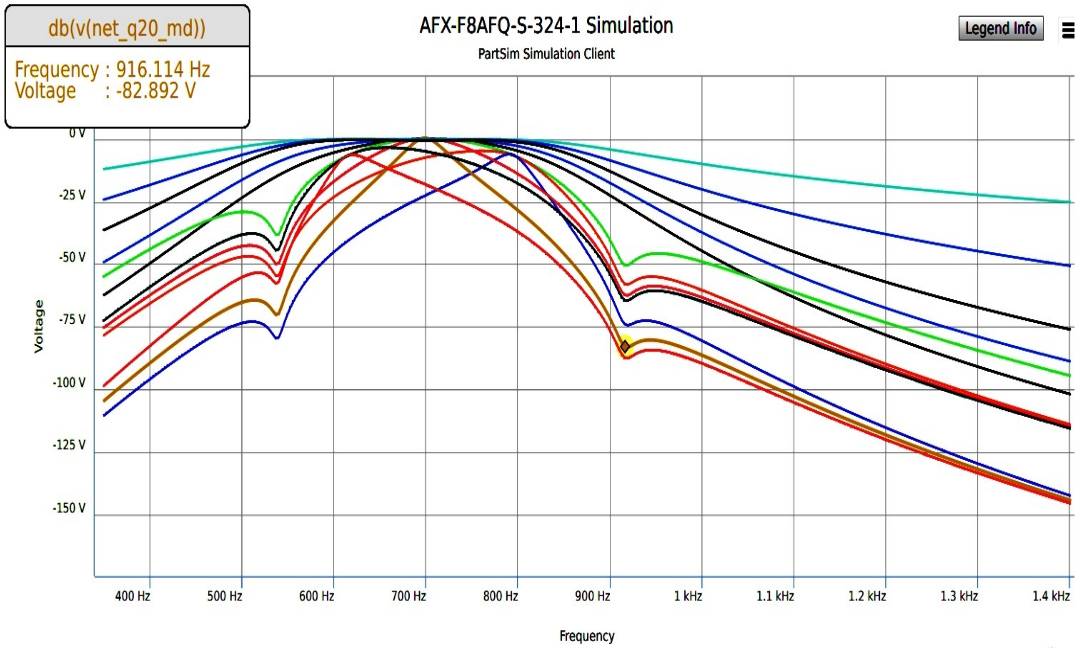
Showing All Bode plots while varying Final Sharp Filters (black),
(Variable Final Sharp Filters stay within All-Pass (red)).
Final variable BandPass drops only to -2dB at +/- 75 Hz..

Notch BW = 100 Hz @ -3 dB
Notch BW = 360 Hz @ -53 dB

Variance = 5.2 Hz sideband expansion per -dB attenuation which is analogous to the DSP "Brick-Wall" effect.

This signal has a slightly expanding side-band skirt.
This signal eventually drives an audio loud-speaker.

Magnitude Plot : Circuit #2 ... from a test circuit



ngSpice Magnitude Scale : V = -dB ; 700mV = -3dB ; 500mV = -6dB

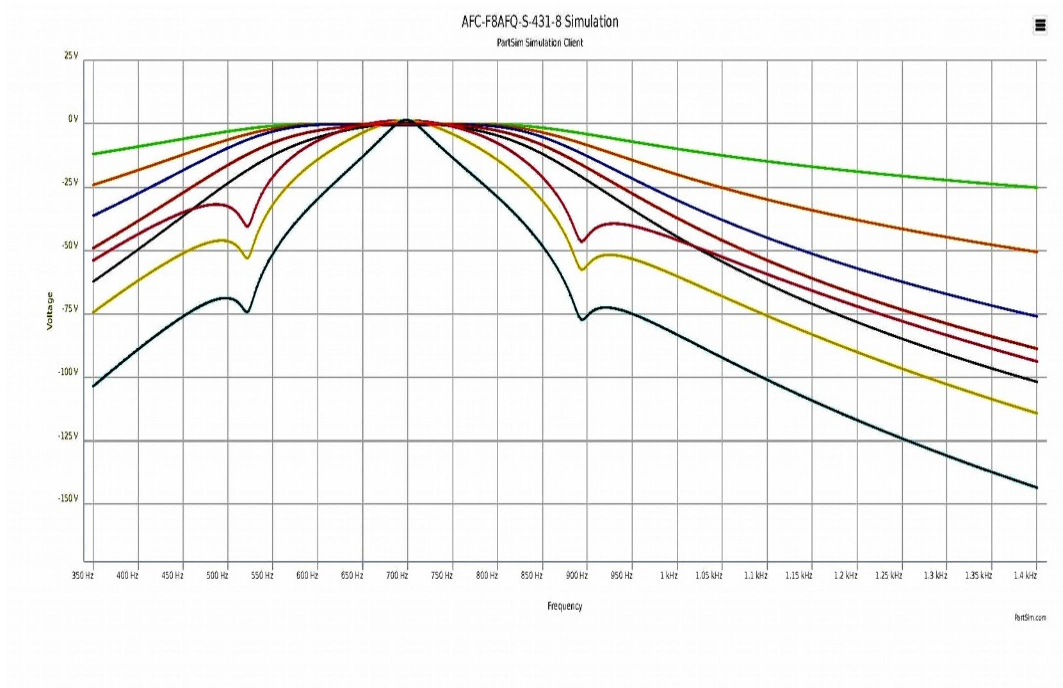
*** Notch Generator freq tracks with f(0) band-pass varying.
*** at +/- 100 Hz f(0) is 0.5dB down

Notch BW = 100 Hz @ -3 dB
Notch BW = 360 Hz @ -53 dB

Variance = 5.2 Hz sideband expansion per -dB attenuation which is analogous to the DSP "Brick-Wall" effect.

This signal has a slightly expanding side-band skirt.
This signal eventually drives an audio loud-speaker.

Magnitude Plot :   Circuit #2 ...   from a test circuit



Magnitude Scale : V = -dB ; 700mV = -3dB ; 500mV = -6dB
***   Magnitude plot: At f(700) normal settings

Red trace is Raw AllPass , Yellow is Fx(01) , Green is Fx(02)
Hi-Notch at 890 Hz, -46 dB and -57 dB and -77 dB .
Lo-Notch at 510 Hz, -40 dB and -53 dB and -74 dB .

*** dark Green trace is Fx(02) (Final Sharp Filter)
Spreads at the rate of 5. Hz per dB attenuation
This final signal is aprox DSP "**Brick-Wall**" .

Variance = 5.2 Hz sideband expansion per -dB attenuation
which is analogous to the DSP "Brick-Wall" effect.

This final V(out) signal has a slightly expanding side-band skirt.
This signal eventually drives an audio loud-speaker.

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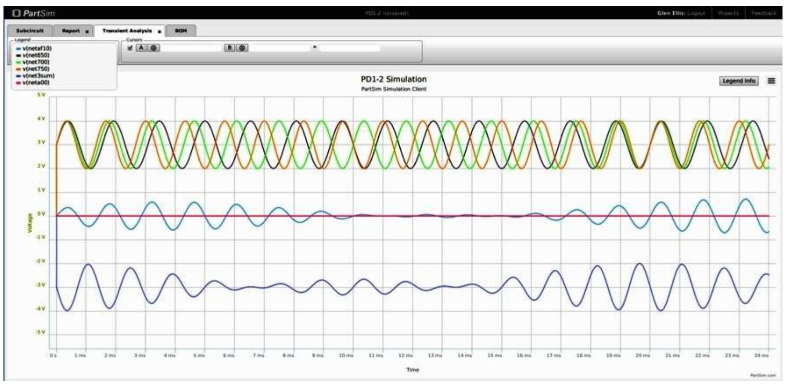
Rationale for this project:

In Amateur Radio CW operations,
we commonly tune for a 700 Hz audio signal,
but other signals may also be present,
interferring with the 700 Hz signal.
These ‘other’ interferring signals are presented here
as 600 Hz and 800 Hz.



Above:
Transient Plot shows Phase Delays and inter-actions based on
“triple-signals “(600Hz , 700Hz , 800Hz) injected simultaneously.

“Triple-Signals” are shown as they phase-shift across time.
“Triple-Signals” are similar to CW radio signals of differing frequencies
as received for processing in the radio receiver and operator ears.

The purpose of the Filter is to select the central 700Hz signal,
while attenuating the other two (600Hz , 800Hz) .

* (red-green-blue trace on top)
Shows “Triple-Signal” phase differences on input signal ,
mixing across time in a Transient Plot.

* (blue trace on bottom)
Filter forces phase shifting on input signal
and produces a symetrical output signal.
Filter responds to the beat frequencies
with peaks and nulls in its output.

* Without any Filtering,
we are left with only the bottom Blue trace
and information is lost in intermodulation mix of three signals.

****

Background Introduction …

The initial Problem for the "AFC" Research Project :

**This project began while we were designing
a very selective Morse-Code Audio Filter.**

**In Morse-Code Amateur-Radio operations
we may receive three signals within a 300Hz audio passband.
Typical signals may be 650Hz, 700Hz , 750Hz.
Commonly we tune for and listen for a 700Hz target signal tone,
but we may also hear signals +/- 50 Hz, +/- 150 Hz, +/- 250 Hz, etc,
which can make accurate copy of the target signal difficult.**

**Therefore we design Narrow Audio Band-Pass Filters,
to pass primarily the 700Hz target signal +/- 50 Hz .**

**Frequently, adjacent signals are 30 dB louder than the target signal
and these strong adjacent signals need to be attenuated
for clear hearing of the morse-code message signal.**

**Having several Morse Code signals at nearly the same audio pitch
is Very confusing to the ear/brain.
Our ear/brain system can only focus on one of them
thus, we need to filter for the 700Hz target signal.**

**Our "AFC" filtering method
may be described as 'phasing-out' the odd signals
and 'phasing-in' the desired signals ;
ie, by controlling / comparing / differentiating the phases
of the confluence of signals passing through the circuit.**

**Our "AFC" analog design is comprised of modular sub-sections
which could be utilized as stand-alone stages in other applications.**

**Our "AFC" analog design functions with-out computer processing;
ie, the "AFC" circuit normally receives the "ear-phone" signal
and then processes the signal and is a "Stand-Alone Circuit".**

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Conclusions:

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Consult :

Robert Arnold Orban   **(Orban Labs Inc.)**
published on ResearchGate.Net 2019.

First, please be aware that the resulting notch filter will be very non-minimum-phase (i.e. many zeros in the Laplace s-plane will have positive real parts), so the filter's step and impulse response are likely to distort waveforms much more than a simpler, minimum-phase realization. So please make sure this is really what you want.

As for analyzing the network, this can be done algebraically in the Laplace Transform domain. If the transmission gain function is

H(s) = [(n1(s)/d1(s)] + [n2(s)/d2(s)]

where n1 and d1 are the numerator and denominator of the upper allpass chain and n2 and d2 are the numerator and denominator of the lower allpass chain,

then the poles of H are all of the poles of the two branches (11 in total, if the top is one stage and the bottom is 10 stages).

The zeros can be found by a numerical root finder, where the polynomial n3 representing the zeros is

n3 = n1 * d2 + n2 * d1

That is, we rearrange H to put it in the form n(3)/d(3), where d3 = d1 * d2 and is the polynomial whose roots are all poles of both branches. The only potential messiness is finding the roots of the high-order polynomial n3.

I would use a computer algebra program like Maple to manipulate the polynomials into the proper form and then to take the roots of n3 numerically.

Once you have the roots of n3 (i.e. the zeros of the notch filter), you should see some complex conjugate roots with a 0 or very small real part (due to numerical accuracy limitations in the rootfinder), putting them on the imaginary axis in the Laplace s-plane. These represent the notch frequencies of the resulting notch filter, and the absolute values of the imaginary parts are the notch frequency in units of radians/second.

As a side note, this differential allpass structure is useful for synthesizing certain classes of lowpass filters, including odd-order Butterworth and odd-order Elliptic Function lowpass filters. However, this technique is mainly popular for making digital filters that have low sensitivity to coefficient quantization.

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Consults on ResearchGate.Net

Barrie Gilbert, Ph.D., VP.Eng, Analog Devices Inc., Beaverton, Or
Mark Sitkowski, Ph.D. , Design Systems Ltd., Australia,
Joseph Puncochar, Ph.D. , Prof, VŠB-Technical Univ of Ostrava Czech .
Robert Arnold Orban, Ph.D., Orban Labs. Inc.
Dobromir Dobrev. Ph.D , profesor, Bulgarian Academy of Sciences

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Bibliography of consulted texts:

``Time Varying Delay Effects'', by Julius O. Smith III and Nelson Lee,
REALSIMPLE Project — work supported by the Wallenberg Global Learning Network .
Released 2008-06-05 under the Creative Commons License
(Attribution 2.5), by Julius O. Smith III and Nelson Lee

Active Filter Design Techniques
Bruce Carter, Ron Mancini, in Op Amps for Everyone (Fifth Edition), 2018

Active Filter Design Techniques
Thomas Kugelstadt, in Op Amps for Everyone (Third Edition), 2009

Analog Low-Pass Filters
Marc T. Thompson Ph.D., in Intuitive Analog Circuit Design (Second Edition), 2014

Frequency-Dependent Schroeder Allpass Filters
by Sebastian J. Schlecht
Department of Signal Processing and Acoustics and Department of Media,
Aalto University, 02150 Espoo, Finland
Appl. Sci. ***2020****, 10(1), 187; https://doi.org/10.3390/app10010187*

Analog Low-Pass Filters, Marc T. Thompson Ph.D.,
in Intuitive Analog Circuit Design (Second Edition), 2014

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