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Bilinear transformation filter matlab

I have a butterworth filter in first place with a cutoff frequency of ω_c. The transfer function is then $$H(s) = \frac{\omega_c}{s+\omega_c}$$ Using the bilinear transformation to find $H(z)$ (what function is called?), I get $$H(z)=\frac{\omega_c}{\frac{2}{T}\frac{z-1}{z+1} + \omega_c} = \frac{\omega_c z + \omega_c}{\left(\frac{2}{T}+\omega_c\right)z + \omega_c-\frac{2}{T}}$$ However, I cannot reconcile this result with what Matlab did. It seems wrong, no matter what the value $T $. I assume that B and A below are the coefficients $H (z)$. >> [B,A] = butter(1,0,5) B = 0.5000 0.5000 A = 1.0000 -0.0000 >> [B,A] = butter(1.0.6) B = 0.5792 0.5792 A = 1.0000 0.1584 >> [B,A] = butter(1.0.7) B = 0.6625 0.6625 A = 1.0000 0.3249 >> [> B,A] = butter(1.0.8) B = 0.7548 0.7548 A = 1.0000 0.5095 What do I misunderstand? Hi, I'm currently designing a digital filter based on analog design. However, when I use the bilinear command I get two different results depending on the computer I have used. The difference between Freq's response is significnat, especially in phase response. I was hoping for -180, which I got on a GOOD computer and on a BAD computer I got -504. I run the same script on both computers and I don't know what the reason for the problem is. When I use impulse invariance I have no problem. Both computers use the same version of Matlab (2015b 8.6.0.267246) and the same Win7 enterprise 64bit and the same CPU. I wrote a script that compares the results of bilinear transformations on a computer to see if it's the same as GOOD or BAD. I did this test on another computer and some of them showed GOOD results and others showed BAD results. I would love to know what is the reason bilinear transformation gives different results depending on the computer it runs. Attached is the code and file mat required to run it. Thank you, Jonathan. Classic IIR filter design techniques include the following steps. Find an analog lowpass filter with cutoff frequency 1 and translate this prototype filter into the specified band configurationTransform the filter to a digital domain. Distinguish filters. The toolbox provides a function for each of these steps. Alternatively, the butter, cheby1, cheb2ord, elliptical, and besself functions perform all filter design steps and the buttord, cheb1ord, cheb2ord, and elliptical functions provide minimum order computing for IIR filters. These functions are sufficient for many design problems, and lower-level functions are generally not required. But if you have an application in which you need to change the edges of the analog filter tape, or pause the rational transfer function, this section describes the tools for doing so. This toolbox provides a number of functions for creating lowpass analog prototype filters with cutoff frequency 1, first in the classic approach to IIR filter design. The table below summarizes analog analog prototypes functions for each supported filter type; plots for each type are displayed in the IIR Filter Design. Filter TypeAnalog Prototype FunctionBessel[z,p,k] = besselap(n)Butterworth[z,p,k] = buttap(n)Chebyshev Type I[z,p,k] = cheb1ap(n,Rp)Chebyshev Type II[z,p,k] = cheb2ap(n,Rs)Elliptic[z,p,k] = ellipap(n,Rp,Rs)The second step in analog prototype prototype design techniques is lowpass prototype frequency transformation. The toolbox provides a set of functions for converting analog lowpass prototypes (with a cutoff frequency of 1 rad/s) into bandpass, highpass, bandstop, and lowpass filters with the specified cutoff frequency. Frequency TransformationTransformence FunctionLowpass to lowpass [numt,dent] = lp2lp (num,den,Wo)[At,Bt,Ct,Dt] = lp2lp (A,B,C,D,Wo)Lowpass to highpass [numt,dent] = lp2hp (num,den,Wo)[At,Bt,Ct,Dt] = lp2hp (A,B,C,D,Wo)Lowpass to bandpass [numt,dent] = lp2bp (num,den,Wo,Bw)[At,Bt,Ct,Dt] = lp2bp (A,B,C,D,Wo,Bw)Lowpass to bandstop [numt,dent] = lp2bs (num,den ,Wo,Bw)[At,Bt,Ct,Dt] = lp2bs(A,B,C,D,Wo,Bw)As shown, all frequency transformation functions can accept two linear system models: the transfer function and the state space form. For bandpass and bandstop casesandwhere ω1 is the edge of the bottom band and ω2 is the edge of the upper band. The frequency transformation function performs a substitution of frequency variables. In the case of LP2BP and LP2BS, this is a second-order substitution, so the output filter is twice the input order. For lp2lp and lp2hp, the output filter is the same order as the input. To start designing chebyshev Type I 10 bandpass order filters with a value of 3 dB for passband ripples, enterOutputs z, p, and k contain zeros, poles, and lowpass analog filter acquisition with a cutoff frequency of Ωc equal to 1 rad/s. Use the function to convert this lowpass prototype into a bandpass analog filter with ribbon edges of Ω1=π/5 and Ω2=π. First, convert the filter to a state-space form so that the lp2bp function can accept it:[A,B,C,D] = zp2ss(z,p,k); % Convert to country-space form. Now, find the bandwidth and frequency center, and contact lp2bp:u1 = 0.1*2*pi; u2 = 0.5 * 2 * pi; % in radians per second Bw = u2-u1; Wo = sqrt(u1*u2); [At,Bt,Ct,Dt] = lp2bp(A,B,C,D,Wo,Bw); Finally, calculate the frequency and plot response:[b,a] = ss2tf(At,Bt,Ct,Dt); % Convert to TF form with = linspace(0,01,1,500)*2*pi; % Generate vector frequency h = freqs(b,a,w); % Semilogy computational frequency response (w/2/pi,abs(h)) % Plot log magnitude vs freq xlabel('Frequency (Hz)') grid The third step in analog prototyping techniques is filter transformation to a discrete-time domain. The toolbox provides two methods for this: impulse and bilinear invariant transformation. The filter design works butter, cheby1, cheby2, and elliptical using bilinear transformation to be discretized in this step. Analog to Digital TransformationTransformation FunctionImpulse invariance[numd,dend] = impinvar (num,den,fs)Bilinear transform[zd,pd,kd] = = bilinear (num,den,fs,Fp)[Ad,Bd,Cd,Dd] = bilinear (At,Bt,Ct,Dt,fs,Fp)The impinvar toolbox function creates a digital filter whose impulse response is a sample of the continuous impulse response of the analog filter. This function only works on filters in the form of transfer functions. For best results, analog filters must have negligible frequency content above half the sampling frequency, as such high frequency content is added to the lower bands during sampling. Impulse invariance works for some lowpass and bandpass filters, but is not suitable for highpass and bandstop filters. Design a Chebyshev Type I filter and plot its frequency and phase response using FVTool:[bz,az] = impinvar(b,a,2); fvtool(bz,az) Click the Magnitude and Response Phase toolbar button. Impulse invariation maintains a cutoff frequency of 0.1 Hz and 0.5 Hz. Bilinear transformation is a continuous nonlinear mapping of the domain to a discrete domain; this maps the s-plane into a z-plane by which the jΩ-axis map transformations the continuous domain to a circle of discrete domain units according to the bilinear toolbox function implementing this operation, where the constant k warping frequency equals twice the sampling frequency (2*fs) by default, and equal to 2πfp/tan(πfp/fs)if you provide a trailing argument bilinear representing the fp match frequency. If the Fp matching frequency (in hertz) exists, the bilinear maps the frequency of Ω = 2πfp (in rad/s) to the same frequency in the discrete domain, normalized to the sampling rate: ω = 2πfp/fs (in rad/sample). Bilinear functions can perform this transformation on three different linear system representations: zero-pole-gain, transfer function, and state-space form. Try calling the bilinear with a state-space matrix that describes the Chebyshev Type I filter from the previous section, using a sampling frequency of 2 Hz, and maintaining the lower band edge of 0.1 Hz:[Ad,Bd,Cd,Dd] = bilinear(At,Bt,Ct,Dt,2,0.1); The resulting digital filter frequency response is[bz,az] = ss2tf(Ad,Bd,Cd,Dd); % Convert to TF fvtool(bz,az) Click the Toolbar Magnitude and Response Phase buttons. The lower band edge is at 0.1 Hz as expected. Note, however, that the edge of the upper band is slightly less than 0.5 Hz, although in the analog domain it is exactly 0.5 Hz. It describes the nonlinear properties of bilinear transformation. To ward off this nonlinearity, it is necessary to create an analog domain filter with the edges of the prewarped tape, which maps to the correct location after the bilinear transformation. Here the planned frequencies u1 and u2 generate Bw and Wo for the lp2bp:fs = 2 function; % Sampling frequency (hertz) u1 = 2*fs*tan(0.1*(2*pi/fs)/2); % lower band edge (rad/s) u2 = 2*fs*tan(0.5*(2*pi/fs)/2); % top tape edge (rad/s) Bw = u2 - u1; % Bandwidth Wo = sqrt(u1*u2); % Middle frequency [At,Bt,Ct,Dt] = lp2bp(A,B,C,D,Wo,Bw); Digital bandpass filter with edges correct 0.1 and 0.5 times nyquist nyquist frequency =bilinear(At,Bt,Ct,Dt,fs); Examples of bandpass filters from the last two sections can also be created in a statement using cheby1's complete IIR design function. For example, the analog version of the Chebyshev filter example is[b,a] = cheby1(5,3,[0,1 0.5]*2*pi,'s'); Note that the edges of the tape are in rad/s for analog filters, while for digital cases, frequencies are normalized:[bz,az] = cheby1(5,3,[0.1 0.5]); All complete design functions call bilinear internally. They prewarp the edge of the band as needed to get the digital filter right. Filter.

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