Report

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Question 1:

(a)

The coefficients are calculated by using the formula for IDFT:

$$x[n] = \frac{1}{2\pi}\int_{-\pi}^{\pi} X(w)e^{jw(n)}dw$$

(b)

No, the phase does not seem to be linear with frequency, in fact we obtain a sort of zig-zagged
phase when plotted against the frequency.

(d)

We obtain a better result with the Blackman window as the filter dips to -100 dB attenuation once it reaches the cutoff frequency whereas we see some ripples appearing after the cutoff frequency in the filter that uses a rectangular window.

(f)

The now-constructed filter is of the form the allows frequencies above a threshold to pass through it. It is a high-pass filter. It provides no attenuation to w's such that $|w| > w_c$

Question 2:

(d)

Changing the value of r_0 basically changes the location of poles of the transfer function of the filter, which affects the output of the filter.

Increasing the value of r_0 sharpens the notches of the filter, restricts the range of frequencies that pass through it even more, making it more frequency-specific, and reducing it on the other hand allows a comparatively wider range of frequencies to pass through.

(e)

The sound's quality seems reduced / the sound starts to become very distorted once the noise is added on top of it.

Question 3:

(d)

On changing the design method to least squares the obtained filter seems pretty similar to the equiripple one. One notable difference is that in the equiripple design, we get a constant ripple amplitude after the stoppage frequency, and in this case the attenuation in the stoppage band seems to be higher than that in the least squares filter, thus in this case making it the better choice.
