Digital signal Processing

1. **Speaker Recognition**: Record vowel 'a' as in bat from one male and one female speaker. Use sampling rate of 8 kHz. Use 30 ms of each utterance and plot the spectrum of both using 512 point DFT. Comment on the two spectra. If a spectrum of vowel is given, will you be able to identify the gender of the speaker with that?

2. **Understanding the Goertzel filter**: Plot the magnitude response of the first order Goertzel filter used to compute the strength of 500 Hz frequency component in a signal sampled at 8000 Hz. Also plot poles and zeros. Assume 8000 frequency points. Comment on the magnitude response.

3. **DTMF Generation and Detection**: Generate two single tones of frequency 697 Hz and 1209 Hz. Use a sampling rate of 8 kHz. Add the two signals. Plot the spectrum of this sum signal. Decode the DTMF signal thus generated. Use Goertzel's algorithm.

4. **Understanding the difference between DFT and DCT in terms of energy compactness**: Read the vowel given 'fa.wav'. Find the sampling rate of the signal. Select 30 ms of the signal from mid of the vowel. How many samples are present in this 30 ms signal? Compute N point DFT of these selected samples where N-is the length of the selected signal. Also, compute N point DCT of these selected samples. Compute N-point IDFT and IDCT of these with first N1 points where (i) N1=10, (ii) N1=50, (iii), N1=100 and N1=200. Plot the original selected 30 ms signal, IDFT and IDCT obtained in each of these cases and compare. What is the inference? Does this illustrate high energy compactness of DCT compared to DFT?

5. **Application related to overlap and save**: Read the given wav file, "machali.wav" using "wavread" function. Find the sampling rate of the signal. Generate a sinusoid of frequency 3000 Hz of length same as that of the given wave file. Add this sinusoid as noise to the signal. Listen to the original and its noisy version. Let the impulse response of filter be h=[1 -2cos(2π3000/fs) 1]. Use overlap and save method with block length 100 to

perform linear filtering and listen to original signal, noisy version and filtered signal. Also, plot these three signals and compare.

6. **Application related to overlap and add**: Read the given wav file, "machali.wav" using "wavread" function. Find the sampling rate of the signal. Generate a sinusoid of frequency 3000 Hz of length same as that of the given wave file. Add this sinusoid as noise to the signal. Listen to the original and its noisy version. Let the impulse response of filter be h=[1 -2cos(2π3000/fs) 1]. Use overlap and add method with block length 100 to perform linear filtering and listen to original signal, noisy version and filtered signal. Also, plot these three signals and compare.

7. **Effect of passing a signal through a moving average filter**: Generate 8 single tones of frequencies varying from 1 to 8 kHz in step of 1 kHz. Use a sampling rate of 8 kHz. Add all 8 signals. Plot the resultant time domain signal and its spectrum. Assume 8000 frequency points. Pass the signal through moving average filters of length-3 and length-5. Plot the output signals and their spectra. Compare the three time domain signals and the three spectra and comment.

8. **Understanding the application of chirp-z transform**: Chirp-Z transform computes z-transform on a spiral contour. One of the applications of chirp-Z transform is spectral zooming. Generate two sinusoids of 2000 Hz and 2060 Hz at a sampling rate of 10 kHz for duration of 0.01 second add them. Let this sequence be x(n). Compute 512 point DFT of the resulting signal. Now compute 512 point DFT of the sequence, $x(n)r^{-n}$ where r=0.96 (circle of radius 0.96). Compare the two plots and comment.

9. **Design of Graphic equalizer**: Design a 3-band graphic equalizer with the following specifications.

   Band1-Low frequency components up to 500 Hz (bass),

   Band2-Band of frequencies from 500 Hz to 1.5 kHz (vocal),

   Band3-High frequency components from 1.5 kHz to 20 kHz (treble).

Assume sampling rate of 44.1 kHz. Verify the design by passing an audio sampled at 44.1 kHz through the equalizer designed.

10. **Compute fundamental frequency of a speaker using autocorrelation**: Record vowel ‘a’ as in bat from one speaker. Use sampling rate of 8 kHz. Extract 30 ms of the signal from mid of the vowel. Compute its autocorrelation. Measure the difference between two peaks of the autocorrelation function. Measure fundamental frequency of the speaker as reciprocal of that difference.

11. **Design a notch filter**: Design a notch filter to stop a disturbance with frequency F_d=1.25 kHz. Assume a sampling frequency of Fs=10 kHz. [Hint: Place a pair of complex conjugate zeros at $z=e^{\pm jwd}$ and a pair of complex conjugate poles at $z=re^{\pm jwd}$ where $w_d=2\pi F_d/F_s$ and r=0.98 or 0.99 (near to unit circle but inside the unit circle)].

12. **Design a resonator**: Design a digital resonator that resonates at 1000 Hz. Assume Fs=8000 kHz. [Hint: Place a pair of complex conjugate poles at $z=re^{\pm jwd}$ where $w_d=2\pi F_d/F_s$ and r=0.98 or 0.99 (near to unit circle but inside the unit circle)].

13. **Design a comb filter**: Design a comb filter that suppresses 50 Hz AC frequency components and its harmonics. Assume Fs=500 Hz. So, frequencies to be suppressed are 50 Hz, 100 Hz, 150 Hz, 200 Hz, 250 Hz, 300 Hz, 350 Hz, 400 Hz, 450 Hz and 500 Hz. [Hint: $H(z)=(1+z^M)/z^M$ where M=number of zeros uniformly spaced on the unit circle which is equal to 10 in this case.]

14. **Sketch the spectrum of an all pass filter**: Plot the magnitude response of an all pass filter and verify $H(z)=(a+z)/(1+az)$ where $0<a<1$.

15. **Echo Generation**: Design an echo generator that generates a single echo.