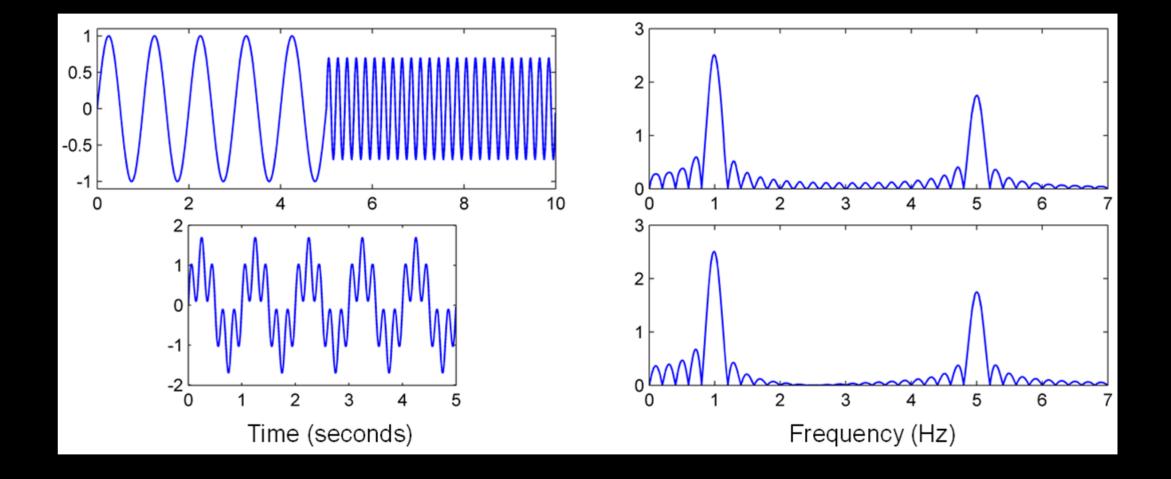
# Supplement

HW 2 2017/3/31

- Fourier analysis assumes that the amplitude/frequency/phase of a signal do not change over time.
- However this is not the case in real world.
- This is why we need STFT to model the temporal dynamics of real world audio signal.



- Continuous-time STFT
  - Simply, in the continuous-time case, the function to be transformed is multiplied by a window function which is nonzero for only a short period of time.
  - The Fourier transform (a one-dimensional function) of the resulting signal is taken as the window is slid along the time axis, resulting in a two-dimensional representation of the signal.

$$\mathbf{STFT}\{x(t)\}( au,\omega)\equiv X( au,\omega)=\int_{-\infty}^{\infty}x(t)w(t- au)e^{-j\omega t}\,dt$$

w(t) is the window function, commonly a Hann window or Gaussian window centered around zero, and x(t) is the signal to be transformed.

#### Discrete STFT

- In the discrete time case, the data to be transformed could be broken up into chunks or frames (which usually overlap each other, to reduce artifacts at the boundary).
- Each chunk is Fourier transformed, and the complex result is added to a matrix, which records magnitude and phase for each point in time and frequency.

$$\mathbf{STFT}\{x[n]\}(m,\omega)\equiv X(m,\omega)=\sum_{n=-\infty}^{\infty}x[n]w[n-m]e^{-j\omega n}$$

likewise, with signal x[n] and window w[n]

- Discrete STFT
  - The magnitude squared of the STFT yields the spectrogram of the function:

$$\operatorname{spectrogram}\{x(t)\}( au,\omega)\equiv \left|X( au,\omega)
ight|^2$$

- Implementation
  - 1. Slice the signal into frames of segments.
    - Each segment may overlap with its previous segment.
  - 2. Multiply the short segments by a window function.
  - 3. Do FFT for each segment.
  - 4. Aggregate the FFT result of each segment into a matrix S.
  - 5. The magnitude squared of the matrix S is the spectrogram.

