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ECE 203 Lab 4: Synthesis of Sinusoidal Signals -- Music Synthesis 2/26/2017

## Section 4.3

```
clear all; close all;

load bach_fugue;
```

### Section 4.3.1

```
% timingInfo.m
fprintf('Timing Information:\n')
bpm = 80
beats_per_second = bpm/60
seconds_per_beat = 1/beats_per_second
seconds_per_pulse = seconds_per_beat/4
[sppn,sppd] = rat(seconds_per_pulse);
fprintf('Seconds per pulse = %g/%g = %g\n',sppn,sppd,
seconds_per_pulse)
PulsesPerMeasure = 16;
fprintf('Pulses per measure = %g\n',PulsesPerMeasure)
fs = 11025; % Sampling rate
fprintf('Sampling rate = %g Hz\n',fs)
```

*Timing Information:*

*bpm =*

*80*

*beats\_per\_second =*

*1.3333*

*seconds\_per\_beat =*

*0.7500*

```
seconds_per_pulse =
```

```
0.1875
```

```
Seconds per pulse = 3/16 = 0.1875
```

```
Pulses per measure = 16
```

```
Sampling rate = 11025 Hz
```

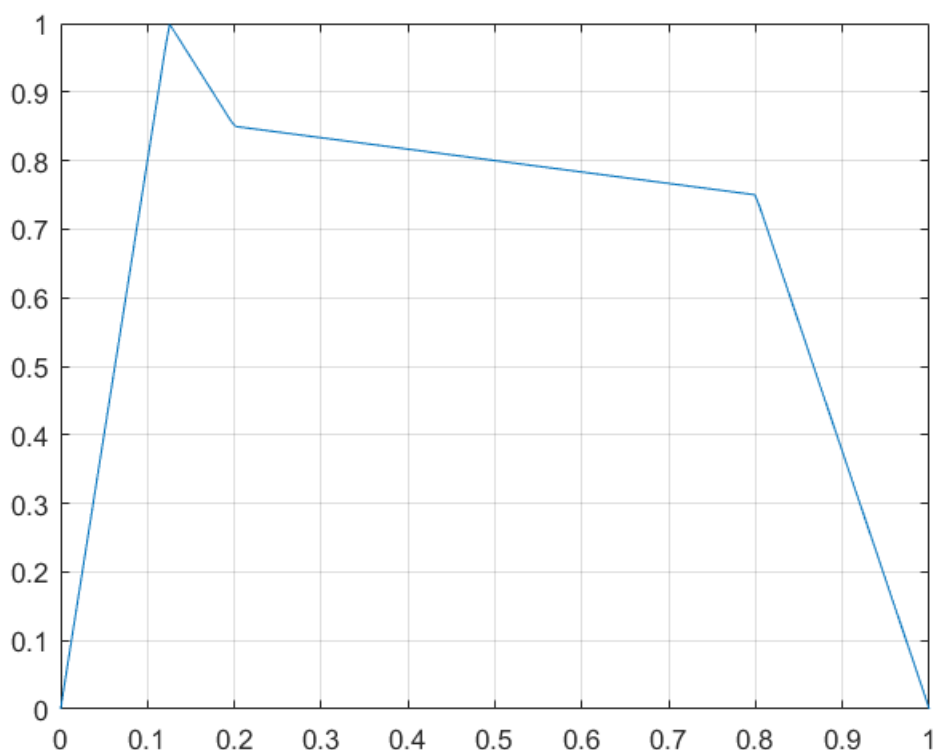
## Section 4.4

```
% myline.m;
%function y = myline(t,t1,y1,t2,y2)
%y = ( (y2-y1)/(t2-t1)*(t-t1) + y1 ) .* (t1<=t & t<t2);

% ADSR.m;
%function y = ADSR(t,t1,t2,t3,t4,A1,A2,A3)
% Sum of 4 straight lines:
% line 1: (0,0) to (t1,A1)
% line 2: (t1,A1) to (t2,A2)
% line 3: (t2,A2) to (t3,A3)
% line 4: (t3,A3) to (t4,0)
%y = myline(t,0,0,t1,A1) + myline(t,t1,A1,t2,A2) + ...
%myline(t,t2,A2,t3,A3) + myline(t,t3,A3,t4,0);

% envelope.m;
%function y = envelope(tin,seconds_per_pulse)
% Shift time from [tin(1),tin(end)] to [0,T],
% where T = tin(end)-tin(1).
%t = tin - tin(1);
%T = t(end); % The attack, release, and delay are always the same, it
% is the slope of the sustain that we are changing.
%t1 = 1/8 * seconds_per_pulse;
%A1 = 1;
%t2 = .2 * seconds_per_pulse;
%A2 = .85;
%t3 = T - .2 * seconds_per_pulse;
%A3 = .75;
%t4 = T;
% When t2 = .2 and t3 = .8, t3-t2 = .6,
% giving us the slope of the sustain.
%A3p = (A3-A2)/.6*(t3-t2) + A2;
%if A3p < 0 % prevent sustain from becoming negative
%    t3 = t2 + .6*A2/(A2-A3);
%    A3p = 0;
%end
%y = ADSR(t,t1,t2,t3,t4,A1,A2,A3p);

tt = linspace(0,1,200);
plot(tt,envelope(tt,1)); grid on
```



## Section 4.5

```
% Analyze MyVoices data file
% showVoice;
fprintf('\n\nShow initial voice data\n\n')
nVoices = numel(theVoices); % Get number of voices
numNotes = zeros(1,nVoices); % Pre-allocate array
lastNotePulseStart = numNotes; % Pre-allocate array
for v=1:nVoices
    numNotes(v) = numel(theVoices(v).noteNumbers);
    lastNotePulseStart(v) = max(theVoices(v).startPulses);
    fprintf('Voice %g has %g notes, last note starts at pulse %g\n', ...
        v, numNotes(v), lastNotePulseStart(v))
    k = min(10, numNotes(v));
    fprintf('startPulses durations noteNumbers\n')
    disp([theVoices(v).startPulses(1:k)'
theVoices(v).durations(1:k)' ...
theVoices(v).noteNumbers(1:k)'])
    fprintf('...\n');
    disp([theVoices(v).startPulses(end)
theVoices(v).durations(end)+1e-12 ...
theVoices(v).noteNumbers(end)']);
    fprintf('\n')
end
```

```

veryLastNotePulseStart = max(lastNotePulseStart);
fprintf('Among all voices, the last note starts at pulse %g\n',...
    veryLastNotePulseStart)
NumMeasures = ceil(veryLastNotePulseStart/PulsesPerMeasure);
fprintf('This piece +-contains %g measures\n',NumMeasures)
PulsesPerSong = NumMeasures*PulsesPerMeasure;
TotalTime = PulsesPerSong*seconds_per_pulse;
Nsamp = 1 + floor(fs*TotalTime);
fprintf('Implementation is %g samples, or about %g seconds\n',...
    Nsamp,TotalTime)

% build voice
%function [noteNums,durs] = buildVoice(voiceIn,PulsesPerSong)
% Expand voice description.
% The input data only tells us the pulse positions where notes
% start and their durations. We use this to build a list of
% pulses for the whole piece. If no note starts at pulse
% k, then we set the note and duration for that pulse to zero.
%durs = zeros(1,PulsesPerSong); % Pre-allocate array
%noteNums = durs; % Pre-allocate array
%for i=1:numel(voiceIn.startPulses)
%    k = voiceIn.startPulses(i);
%    noteNums(k) = voiceIn.noteNumbers(i);
%    durs(k) = voiceIn.durations(i);
%end

% build music
%function waveform =
    buildMusic(Nsamp,noteNums,durs,seconds_per_pulse,fs,env,harm)
%waveform = zeros(1,Nsamp);
%for pulse=1:numel(durs)
%    d = durs(pulse);
%    if d>0
%        k1 = ceil( (pulse -1)*seconds_per_pulse*fs );
%        k2 = floor( ((pulse+d)-1)*seconds_per_pulse*fs );
%        tt = (k1:k2)/fs;
%        xx =
            key2noteModified(2,noteNums(pulse),tt,env,harm,seconds_per_pulse);
%        waveform(k1+1:k2+1) = xx;
%    end
%end

% Then build the waveforms
%buildWaveform;
tf = {'false' 'true' };
env = true;
harm = true;
fprintf('Envelope = %5s, Higher harmonics = %5s\n',tf{env+1},tf{harm
+1})
waveform = zeros(1,Nsamp);
wvf = zeros(3,Nsamp);
for v=1:nVoices
    [noteNums,durs] = buildVoice(theVoices(v),PulsesPerSong);

```

```

        wvf(v,:) =
        buildMusic(Nsamp,noteNums,durs,seconds_per_pulse,fs,env,harm);
        waveform = waveform + wvf(v,:);
    end

    % then plot and waveform and spectrogram
    %plotWaveform;
    p1 = 1;
    p2 = 235; % 235; % max pulse is 235
    k1 = ceil((p1-1)*seconds_per_pulse*fs);
    k2 = floor((p2-1)*seconds_per_pulse*fs);
    tt = (k1:k2)/fs;
    figure(2)
    for v=1:nVoices
        subplot(nVoices,1,v)
        plot(tt,wvf(v,k1+1:k2+1)); grid on
        title([ 'Voice ' num2str(v) ])
        xlabel('Time (s)')
    end
    figure(3)
    plot(tt,waveform(k1+1:k2+1)); grid on

    %plotSpectrogram;

    wl = 512;
    for v=1:nVoices
        figure(3+v)
        specgram(wvf(v,k1+1:k2+1),wl,fs)
        title([ 'Spectrogram of Voice ' num2str(v) ])
        xlabel('Time (sec)')
        ylabel('Frequency (Hz)')
        axis([0 39 0 2500])
    end

    % play the sound
    soundsc(waveform,fs);

    % Scale waveform to have values between -1 and 1
    %ScaleSound.m;
    %function y = ScaleSound(x)
    % Do the scaling operation performed by soundsc and
    % return the result. Let a = min(x(:)), b = max(x(:)), and
    %  $f(x) = 2(x-a)/(b-a) - 1$ .
    % Notice that f is linear and satisfies
    %  $f(a) = -1$  and  $f(b) = +1$ .
    %a = min(x(:));
    %b = max(x(:));
    %if b==a % then x is a constant vector, and we put
    %    y = zeros(size(x));
    %else
    %    y = (x-a)/(b-a)*2-1;
    %end

    waveform=ScaleSound(waveform);

```

```

%plotWaveform;
p1 = 1;
p2 = 235; % 235; % max pulse is 235
k1 = ceil((p1-1)*seconds_per_pulse*fs);
k2 = floor((p2-1)*seconds_per_pulse*fs);
tt = (k1:k2)/fs;
figure(2)
for v=1:nVoices
    subplot(nVoices,1,v)
    plot(tt,wvf(v,k1+1:k2+1)); grid on
    title([ 'Voice ' num2str(v) ])
    xlabel('Time (s)')
end
figure(3)
plot(tt,waveform(k1+1:k2+1)); grid on

% Write .wav file
audiowrite('Lab04solutionOutput.wav',waveform,fs)

```

*Show initial voice data*

*Voice 1 has 123 notes, last note starts at pulse 233*

*startPulses durations noteNumbers*

3.0000	1.0000	52.0000
4.0000	1.0000	51.0000
5.0000	1.2000	52.0000
7.0000	1.2000	47.0000
9.0000	1.2000	48.0000
11.0000	1.0000	52.0000
12.0000	1.0000	51.0000
13.0000	1.2000	52.0000
15.0000	1.2000	54.0000
17.0000	1.2000	47.0000

...

233.0000	2.0000	43.0000
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*Voice 2 has 130 notes, last note starts at pulse 233*

*startPulses durations noteNumbers*

35.0000	1.0000	59.0000
36.0000	1.0000	58.0000
37.0000	1.2000	59.0000
39.0000	1.2000	52.0000
41.0000	1.2000	55.0000
43.0000	1.0000	59.0000
44.0000	1.0000	58.0000
45.0000	1.2000	59.0000
47.0000	1.2000	61.0000
49.0000	1.2000	54.0000

```
...
233.0000    2.0000    52.0000
```

*Voice 3 has 88 notes, last note starts at pulse 225*

*startPulses durations noteNumbers*

```
99.0000    1.0000    40.0000
100.0000    1.0000    39.0000
101.0000    1.2000    40.0000
103.0000    1.2000    35.0000
105.0000    1.2000    36.0000
107.0000    1.0000    40.0000
108.0000    1.0000    39.0000
109.0000    1.2000    40.0000
111.0000    1.2000    42.0000
113.0000    1.2000    35.0000
```

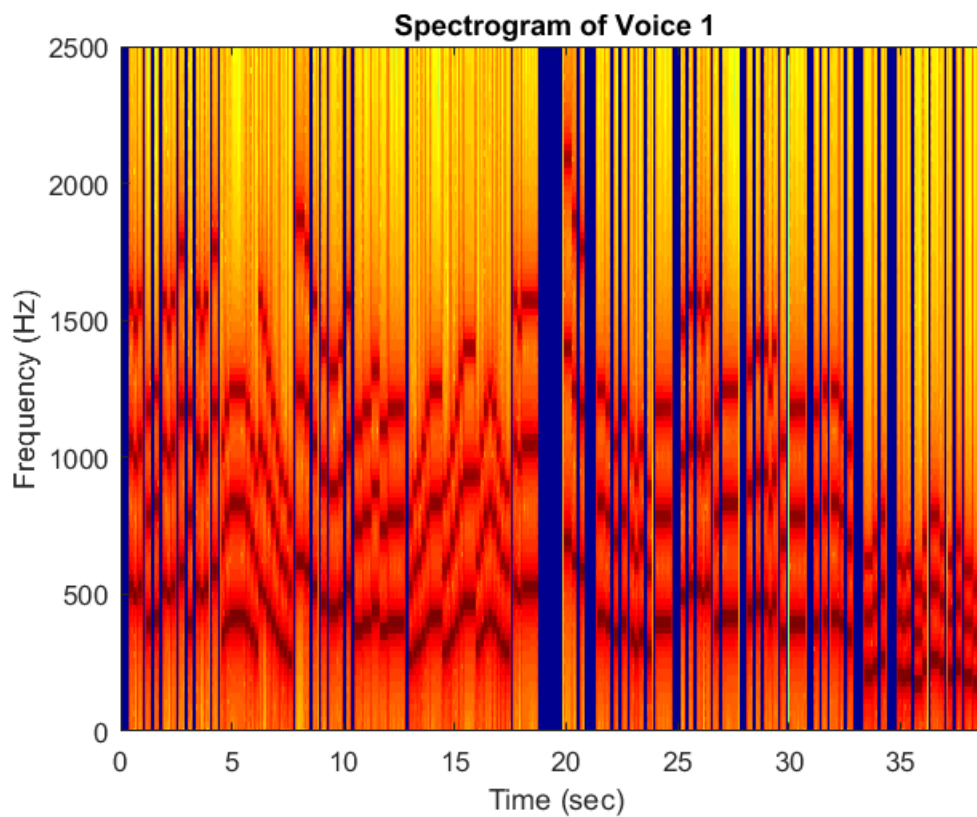
```
...
225.0000    1.2000    31.0000
```

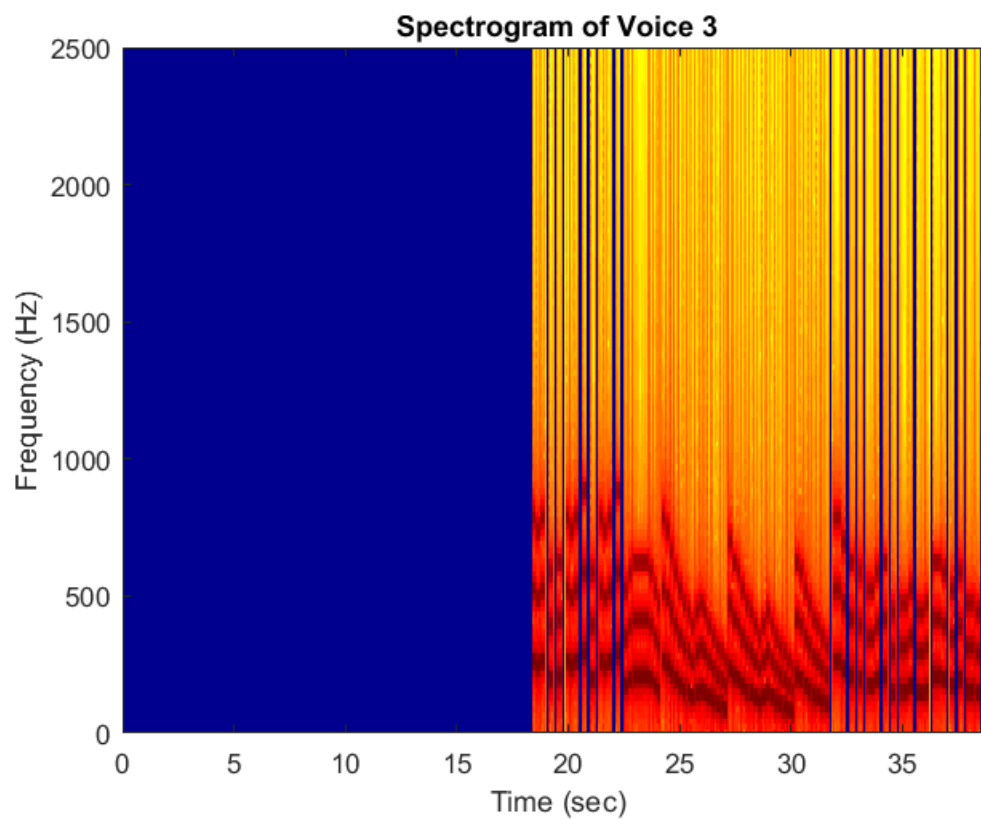
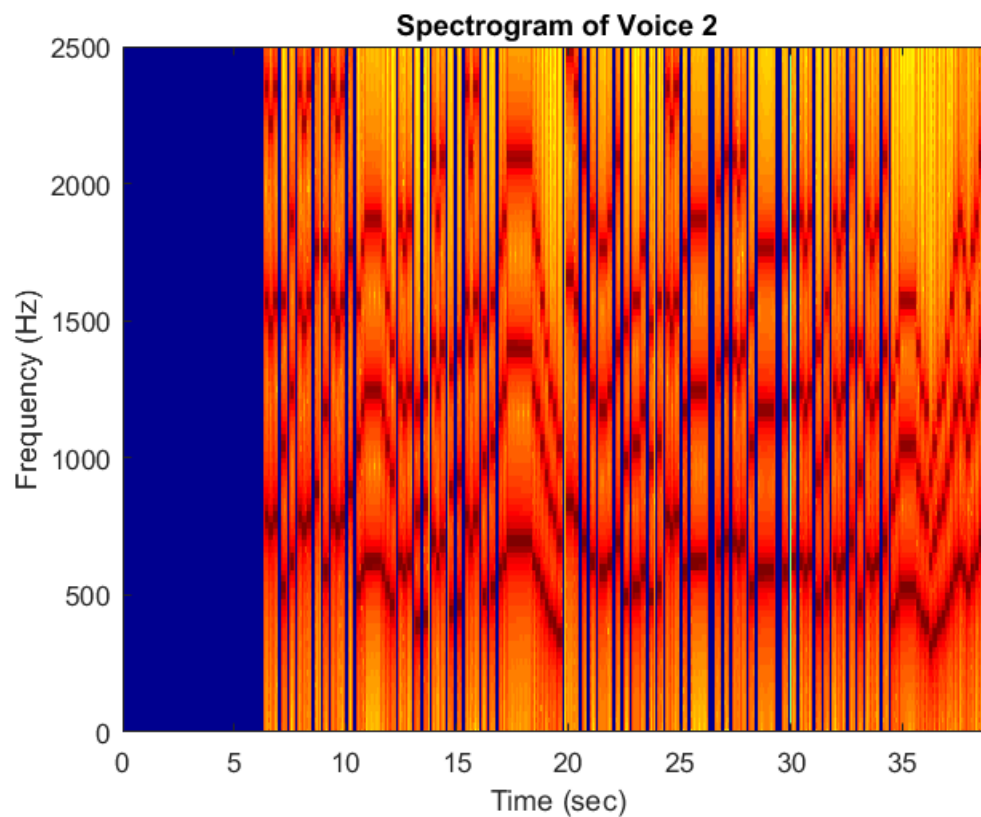
*Among all voices, the last note starts at pulse 233*

*This piece +-contains 15 measures*

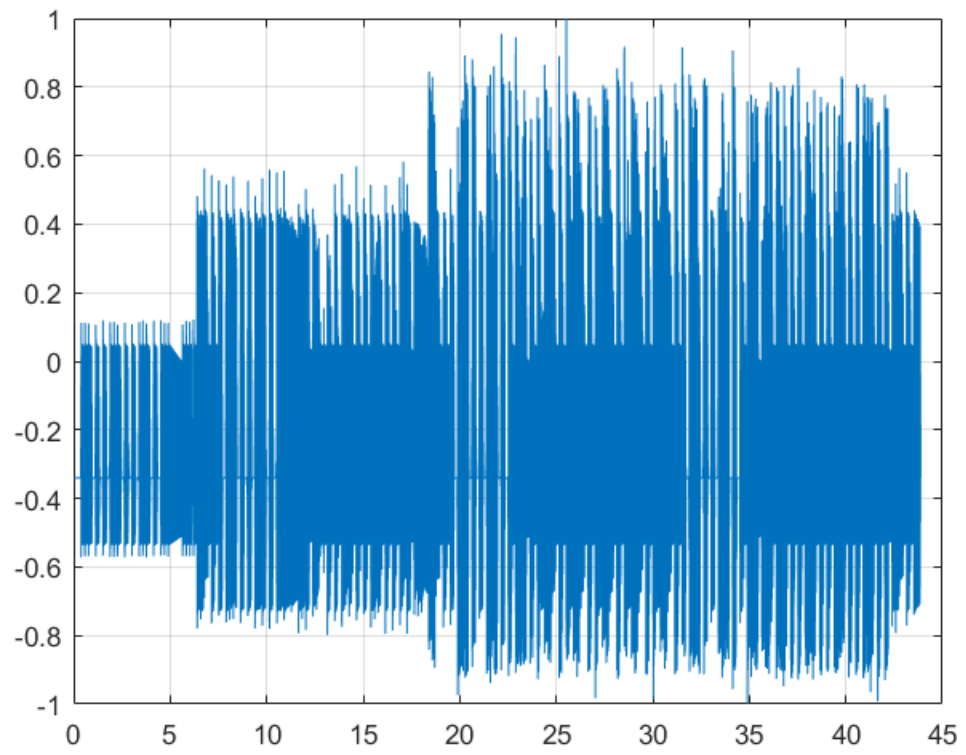
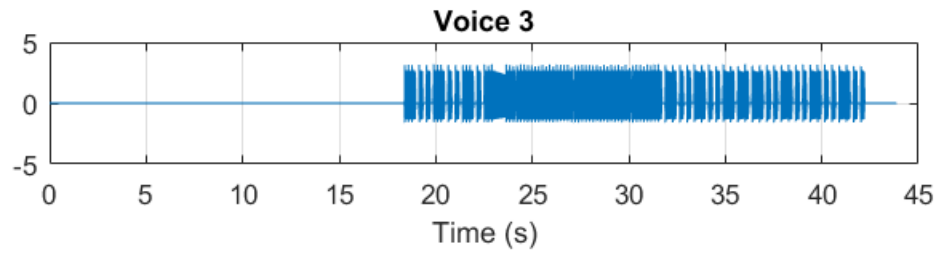
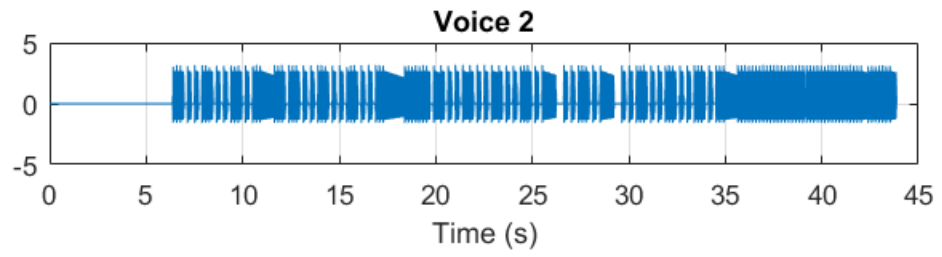
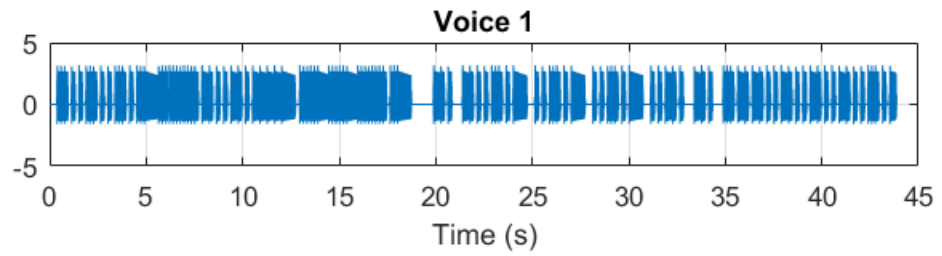
*Implementation is 496126 samples, or about 45 seconds*

*Envelope = true, Higher harmonics = true*









*Published with MATLAB® R2016a*