MusicalSynthesis

```
clear %fresh start
samplingFrequency = 44100; % hardcoded frequency using which all the samples
were generated
%LOADING TRACK AND INSTRUMENT FILES
workDir = pwd; %getting working directory path
track1 = strcat(pwd,'\track1.txt');
track2 = strcat(pwd,'\track2.txt');
track3 = strcat(pwd,'\track3.txt');
xylophone = strcat(pwd,'\Xylo_A4.wav');
distGtr = strcat(pwd,'\GtrDist_A4.wav');
ionosphere = strcat(pwd,'\Ionosphere_A4.wav');
chillPad = strcat(pwd,'\Chill_A4.wav');
bell = strcat(pwd,'\Bell_A4.wav');
%CHOOSE TRACK(S)
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%display dialog box
prompt = 'What track(s) would you like to play?';
list = {'Track 1' 'Track 2' 'Track 3'};
[track,v] = listdlg('PromptString',prompt,...
    'SelectionMode','multiple',...
    'ListString',list, 'ListSize', [300 50], 'Name', 'Musical Synthesis');
if (isempty(track)) %pressing cancel stops the program
     return
end
tracks=0; %initialize an array to store information about which tracks to use
finished = 0; % acts as a boolean
while(finished ==0)
     finished = 1;
     %assign chosen tracks to the tracks array. The index of tracks
     %indicates the number of tracks and values indicate which file to read
     %from i.e. tracks(1) =2, tracks(2) = 3 means Play tunes created from
     %track2.txt and track3.txt
     for i = 1 : length(track)
           tracks(i) = track(i);
     end
end
%CHOOSE VOLUME PER TRACK(S)
volumeArr = 0; %initialise an array of volume levels
if length(tracks) == 1
                                   %when there's only one track there's no point
                                   %mixing volume levels as it's going to be
normalised at the end anyway
     volumeArr(1) = 1:
else %mix volume leves
     for i=1: length(tracks)
           finished = 0; % acts as a boolean
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while (finished==0)
             finished = 1;
             %display dialog box
[volume,v] = listdlg('PromptString',prompt,...
'SelectionMode','single',...
                  'ListString', list, 'ListSize', [300 100], 'Name', 'Musical
Synthesis');
             if (isempty(volume)) %pressing cancel stops the program
                  return
             end
             %assign volume level values to the array. i.e. volumeArr(1) means
             %volume level for the first track
             switch (volume)
                  case {1.2.3}
                      volumeArr(i) = volume;
             end
         end
    end
end
%CHOOSE INSTRUMENT(S)
instruments = 0; %initialise array of instruments
for i=1: length(tracks)
    finished = 0; % acts as a boolean controller for the loop
    while (finished==0)
        finished = 1;
%display dialog box
prompt = horzcat('Choose an instrument for Track
',num2str(tracks(i)),':');
list = {'Xylophone' 'Distorted Electric Guitar' 'Ambient Ionosphere'
'Mellow Pad' 'Bell'};
         [instrument,v] = listdlg('PromptString',prompt,...
              'SelectionMode','single',
             'ListString', list, 'ListSize', [300 100], 'Name', 'Musical
Synthesis'):
         if (isempty(instrument)) %pressing cancel stops the program
             return
         end
        %assign the chosen instrument to the array. i.e. instruments(1) means
        %instrument for the first track
         switch (instrument)
             case {1,2,3,4,5}
                  instruments(i) = instrument;
         end
    end
end
%CHOOSE BPM
%dialog box
prompt = {'Choose the speed of the song in BPM(40-400):'}; %this range is dictated by the duration of original samples dlg_title = 'Musical Synthesis';
num_lines = 1;
```

```
def = {'60',};
BPM = 0; %initialise
finished = 0; %acts as a boolean control for the loop
while(finished==0)
    speed = (inputdlg(prompt,dlg_title,num_lines,def));
    if isempty(speed)
         return %if user clicks on X the program closes
    speed = str2num(speed{:});
    if (speed>=40 && speed<=400) %error checking
         BPM = speed;
         finished = 1; %exit the loop
    else
         prompt = {'Sorry, invalid input. Enter number: 40-400'};
finished = 0; %keep looping until the correct value is input
    end
end
%SYNTHESISE
song=0;
for i=1 : length(tracks)
    %initialise variables track/instrument/volume
    tr=''
    ins='':
    vol = 0;
    %GET THE TRACKS USED IN THE SONG
    switch(tracks(i))
        case 1
             tr = track1;
         case 2
             tr = track2;
        case 3
             tr = track3;
    end
    %GET THE INSTRUMENTS USED IN THE SONG
    switch(instruments(i))
        case 1
             ins = xylophone;
         case 2
             ins = distGtr;
         case 3
             ins = ionosphere;
         case 4
             ins = chillPad;
         case 5
             ins = bell;
    end
    %GET THE VOLUME PER TRACK
    switch(volumeArr(i)) %arbitrary numbers
        case 1
             vol = 1;
         case 2
             vol = 5;
         case 3
             vol = 15;
    end
```

```
%synthesise the track with the chosen instrument at chosen BPM
    tune = synthesise(tr,BPM, ins);
    tune = tune*vol;
    if(i \ge 2)
         if(length(tune)>length(song))
              tune = tune(1:length(song));
         if (length(tune)<length(song))</pre>
              tune = [tune;0];
         end
    end
    %add to the whole song making it polyphonic(when there's more than one
    %track)
    song = song + tune;
end
result = PlaySave(); %1 - play and plot, 2 - save, 3- exit
finished =0;
while (finished ==0)
    switch(result)
         case 1 %play and plot
              close all %making sure we don't get multiple figure windows
              soundsc(song, samplingFrequency); %play the song
             %plot waveform and spectrogram
figure('Name','Waveform','NumberTitle','off')
              plot(song);
              figure('Name','Spectrogram','NumberTitle','off')
spectrogram(song, 1024, 800, 1024, samplingFrequency,'yaxis');
             %wait until the song is finished and display the dialog box
             %again
             pause on;
             pause(length(song)/samplingFrequency);
         case 2 %save
             %normalising the waveform so that the peaks lie between -1 and \%1 and there is no clipping in the exported wav file
             delta = 0.1; %variable that ensures that we never go beyond 1 and
-1
              song = song/(max(abs(song))+delta);
             %save file to the working directory
             savePath = strcat(pwd,'\Export.wav');
wavwrite(song,samplingFrequency,savePath);
             %wait for user to click ok
             waitfor( msgbox('File saved as Export.wav in the working
directory','Musical Synthesis'));
         case 3 %exit
              close all %close all figure windows
              return %exit the program
    end
    %display the box again
    result = PlaySave();
```

synthesise

```
function [ notes ] = synthesise( track, BPM, instrument)
%Synthesises a tune using a resample mechanism
%Input:
%track -
                  String to txt file containing frequencies and durations of
the notes in a
                  "f:d" format
%BPM -
                  integer denoting the number of Beats Per Minute to be used
%instrument -
                  String to wav file containing an instrument sample
%Output:
%notes -
                  array containing a synthesised tune
crotchetDurationSec = 60/BPM;
txt = fopen(track);
values = textscan(txt, '%d:%f');
fclose(txt);
% put the frequency and duration values into two vectors
m = cell2mat(values(1));
for i =1 : length(m)
end
d = cell2mat(values(2));
%load the instrument file
[x,fs] = audioread(instrument);
% create an array to store the resampled notes
notes = zeros;
%resample
for i=1:length(m)
    % O is a rest. If it's a note do the resampling.
    if m(i) \sim = 0
        %NOTE
             sample sounds are an A4 note (440Hz)
        originalFreq = 440;
        %transpose midi number to frequency
        f(i) = nearest(midiToFreq(m(i)));
% doubling both the original freq and intended freq to enable the
use of
             notes whose frequencies are not integers (p/q ratio is preserved)
        note = resample(x, originalFreq*2, double(f(i)*2));
             alter the duration of the note to match the notation
        duration = crotchetDurationSec * fs * d(i);
        note = note(1:duration);
             create ASDR envelope (modified version of http://194.81.104.27/~brian/DSP/ReadMusic.pdf)
        target = [0.99999;0.25;0.05];
gain = [0.005;0.0004;0.00075];
        duration = [125;800;75];
        a = adsr_gen(target,gain,duration,length(note));
           modulate the note to implement ASDR envelope
```

adsr_gen

```
function a = adsr_gen(target,gain,duration,noteDuration)
%Creates an Attack Decay Sustain Release envelope
% target - vector of attack, sustain, release target values
% gain - vector of attack, sustain, release gain values
% duration - vector of attack, sustain, release durations in ms
% Output
% a - vector of adsr envelope values
a = zeros(noteDuration,1); % ASDR envelope has the same duration as the note
duration = round(duration./1000.*noteDuration); % envelope duration in samp
% Attack phase
start = 2;
stop = duration(1);
for n = [start:stop]
a(n) = target(1)*gain(1) + (1.0 - gain(1))*a(n-1);
end
% Sustain phase
start = stop + 1;
stop = start + duration(2);
for n = [start:stop]
a(n) = target(2)*gain(2) + (1.0 - gain(2))*a(n-1);
end
% Release phase
start = stop + 1;
stop = noteDuration;
for n = [start:stop]
a(n) = target(3)*gain(3) + (1.0 - gain(3))*a(n-1);
end;
```

midiToFreq

```
function [ freq ] = midiToFreq( midi )
%converts midi notation to frequency in Hz
% Using a4 as base frequency: midi notation - 69, frequency - 440
a4freq = 440;
a = (double(midi-69)/12);
freq = (2^a) * a4freq;
end
```

PlaySave()

Track1

77:0.5 81:0.5 72:1 74:0.5 77:0.5 70:1 69:0.5 72:0.5 65:1 64:1 64:2 65:1

Track2

65:1.5 64:0.5 65:1 62:1 58:2 60:1.5 0:0.5 58:0.5 57:0.5 55:1.5 58:0.5 60:1

Track3

60:1.5 58:0.5 60:1 0:1 58:2 60:2 58:1 60:2 65:1