

Engineering 100
Music Signal Processing
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- Computer synthesis of musical signals using:
- Digital turntable synthesis of instruments by upsampling/downsampling of one snippet;
- Additive synthesis of artificial instruments: Make your own kind of music.

Computer Synthesis of Music

- Compose your own music using existing instruments (need a snippet [short sample]):
Digital turntable: Up/down-sample snippet.
- Make your own kind of music using artificial computer-generated instruments!
Additive synthesis: Sum Fourier series.
- Subtractive and FM synthesis not covered.

Analog Turntable Synthesis

- Take a recording of (say) single clarinet note.
- Speed up/slow down tape to change its pitch.
- Only need variable-speed motor or turntable.
- Alvin & the Chipmunks: Double tape speed.
- Cut and paste (literally!) or dub tape→music.

Digital Turntable Synthesis

- Take a recording of (say) single guitar note.
- Sample (digitize)→string of numbers→note.
- Easy to cut to desired duration & concatenate different notes & add instruments in Matlab.
- But how do we alter pitch if have one note?
- Upsample & downsample & Circle of Fifths.

Doubling/Tripling Pitch

- Double pitch of the musical note stored in X:
`>>Z=[X X];Y=Z(1:2:length(Z));`
- Take every other sample→halve sampling rate. Take X twice so duration of Z=duration of X. All sinusoids are now oscillating twice as fast. Make sure X bandlimited to $\frac{1}{4}$ (sampling rate).
- Triple pitch of the musical note stored in X:
`>>Z=[X X X];Y=Z(1:3:length(Z));`

Halving Pitch

- Insert a zero between each pair of samples $x[n]$:
`{...0,x[-2],0,x[-1],0,x[0],0,x[1],0,x[2],0...}`
- Lowpass filter (half-band) this upsampled $x[n]$.
- Result has pitch halved, but duration doubled.
- Why? See [synth.pdf](#) on the CTools website.
- `>>L=length(X);Z=[X;zeros(1,L)];F=fft(Z(:));`
`>>F(L/2+1:3*L/2+1)=zeros(1:L+1);Y=ifft(F);`

Altering Pitch of a Snippet

- Since all semitones are related by ratios of small integers, can obtain all semitones from any single one by up/down-sampling the one.
- “Upsampling and downsampling” is also known as “interpolation and decimation.”
- Another way: Use the musical Circle of Fifths

Musical Circle of Fifths

$\log_{1.5}[f/440 \text{ Hz}]$	0	1	2	3	4	5	6	7	8	9	10	11
octave	0	0	1	1	2	2	3	4	4	5	5	6
Hertz	440	660	495	742	557	835	626	470	705	529	793	595
	440	659	494	740	554	830	622	466	698	523	784	587
Note	A	E	B	F#	C#	G#	D#	A#	F	C	G	D

Repeatedly downsampling by 2 and upsampling by 3 (so no aliasing)→ can obtain all 12 semitones from any one. Why? $1.5 \approx 2^{7/12} \rightarrow (1.5)^{12} \approx 2^7$.

Musical Circle of Fifths

- Place 12 semitones in order on a clock face.
- Take 12 jumps of 7 hours each around face.
- Land on all 12 semitones (relatively prime).
- Jump of 7 semitones=5 whole tones=“fifth” =frequency ratio of 1.5 (A→E: 440→659).
- Worked perfectly until “Well-Tempered Clavichord” (Bach); still works pretty well.

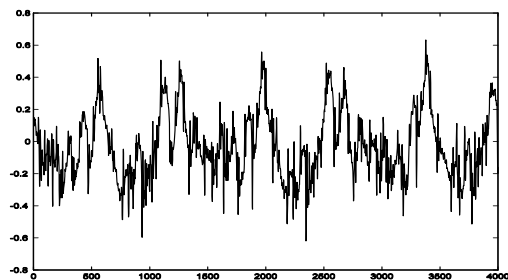
Using Circle of Fifths for Synthesis

- Start with snippet of instrument at G (392 Hz)
- Downsample $Y=X(1:2:\text{length}(X))$ for 784 Hz
- 12 times: Downsample by 2 & upsample by 3
- 12 times: $F=\text{fft}(X); Y=(3/2)*\text{real}(\text{ifft}([\text{F}(1:L/2) \text{ zeros}(1,L/2+1) \text{ F}(L/2+2:L)])); Y=Y(1:L);$
- No aliasing: frequencies down to 2/3 each time
- Have all 12 semitones! Get correct durations.

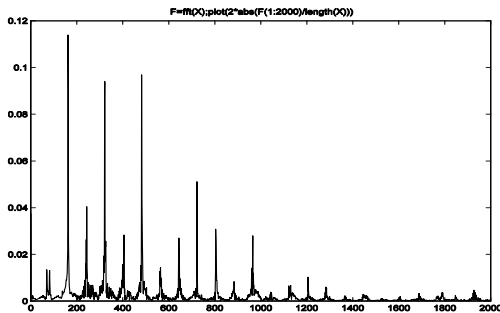
Example: Guitar Note Snippet

- `>>[X,FS]=wavread('guitar.wav');`
- `>>length(X)→114000 and >>FS→44100.`
- Duration of $X=114000/44100=2.585$ sec.
- Period of note $X=1425/44100=.0323$ sec.
- “Pitch” of note $X=44100/1425=30.9$ Hertz.
- $\text{Fund}=(81-1)(44100)/(114000)=30.9$ Hertz.
- 3 instruments: each length $32768=2^{15}$ (why?)

Guitar Note: Time Waveform



Guitar Note: Line Spectrum



Additive Synthesis of Music

- Musical note “A” signal is periodic. Expand:
- $x(t)=c_1\cos(2\pi 440t)+c_2\cos(2\pi 880t)+\dots$
- Amplitudes c_k determine timbre of sound.
- Omit phases, since we can't perceive them.
- Finite #terms, since harmonics can't be heard if > 10 kHz. Also: $< \frac{1}{2}$ (sample frequency).

Additive Synthesis of Music

- c_k are specified by the instrument. BUT:
- Design your own instrument: Choose c_k !
- Hammond and pipe organs: do physically: Opening organ pipe stops varies the c_k .
- See synth.pdf for two examples of c_k sets.
- Only problem: Generating and summing the Fourier series: Lot of real-time computation.

Fast Generation of Sinusoids

- No need to use Matlab to generate sinusoids.
- Recall: $x[n]=A\cos(2\pi Fn+\theta)$ satisfies formula
- $x[n+1]+x[n-1]=2\cos(2\pi F)x[n]$. Rearrange to:
- $x[n+1]=2\cos(2\pi F)x[n]-x[n-1]$. Initial conditions:
- $x[0]=A\cos(\theta)$. $x[1]=A\cos(2\pi F+\theta)$. Recursively.
- Each recursion: One addition and multiplication.
- Do in parallel in Matlab-Will save computation.

Subtractive Synthesis of Music

- Instead of generating harmonics and summing.
- Start with harmonics from a simple waveform (square wave, triangle wave, rectified sine) that is easy to generate with analog circuits.
- Filter waveform with analog lowpass filter.
- Result has different harmonics and timbre.
- Used in 1970s (Moog synthesizer).

FM Synthesis of Music

- FM=Frequency Modulation (vary f with t).
- $x(t)=\cos(2\pi[f_0+I\cos(2\pi f_1 t)]t)$. Can add θ 's.
- Frequency varies sinusoidally around f_0 Hertz. This frequency varies with frequency f_1 Hertz. Amplitude of variation= I =modulation index.
- Computationally easy way of generating a “rich” periodic signal with many harmonics.
- Chowning 1973 paper posted on Ctools site.