

Analog style synthesis with the web audio API

Chris Eberly - Gobbler engineering -
chris.eberly@gobbler.com - @chriseberly

seqwww.io

The screenshot displays the seqwww.io web application interface, which is a digital music workstation. The interface is divided into several sections:

- Piano Roll:** A large grid on the left side of the interface. The vertical axis (y-axis) represents pitch, with labels for notes: OH, CH, SD, BD, E, D#, D, C#, C, B, A#, A, G#, G, F#, F, E. The horizontal axis (x-axis) represents time. Red squares indicate notes or events placed on the grid.
- Drum-78:** A section on the right side of the piano roll, containing four circular meters for drum parameters: bass (100), snare (43.99), closed hh (37.99), and open hh (83.99).
- blue planet:** A section on the right side of the drum-78, containing four circular meters for synth parameters: osc mix (40), attack (60), sustain (28.00), and cutoff (2260).
- SEQ-WWW:** A section on the far right, containing two buttons: "PLAY" and "STOP". Below these are three sections of parameters: "Mix" (drums: 87, poly: 17.00, bass: 14.00), "EQ" (highpass: 80, midrange: 2, lowpass: 20000), and "Compressor" (ratio: 8, attack: 90, release: 400.00, threshold: -24).
- Keyboard:** A row of 12 circular buttons at the bottom of the interface, each representing a note: D#, C, G#, D#, C, G#, C, C, G#, D#, C, G#, G#.

At the bottom right of the interface, there is a small text credit: "SEQ-WWW was made by @chriseberly" and a link: "https://github.com/ceberly/seq-www/".

<https://github.com/ceberly/seq-www>

Subtractive synthesis in 20 or so lines

```
window.AudioContext = window.AudioContext || window.webkitAudioContext;
var context = new AudioContext();

function run(context, now, attack, release, cutoff, resonance) {
  var lp = context.createBiquadFilter();
  var osc = context.createOscillator();
  var env = context.createGain();

  var freq = 120.0;

  osc.connect(env);
  env.connect(lp);
  lp.connect(context.destination);

  lp.frequency.value = cutoff;
  lp.Q.value = resonance;

  osc.start(now);
  osc.type = "sawtooth";
  env.gain.value = 0;

  env.gain.setTargetAtTime(1, now, attack / 1000.0);
  env.gain.setTargetAtTime(0, now + attack/1000.0, release/1000.0);
}
```

Synthesizing TR-808 style cymbals - design

CY

The combined square wave outputs of six Schmitt triggers including two for CB generator is separated into high and low range components by two filters composed of IC3. The high range component from pin 7 of IC3 is further separated into two frequency ranges. The output of the gate Q16 has the highest frequency component of this sound generator. Its decay time is short. The output of Q17 is in a frequency range slightly lower than the above output, and its decay time is controllable.

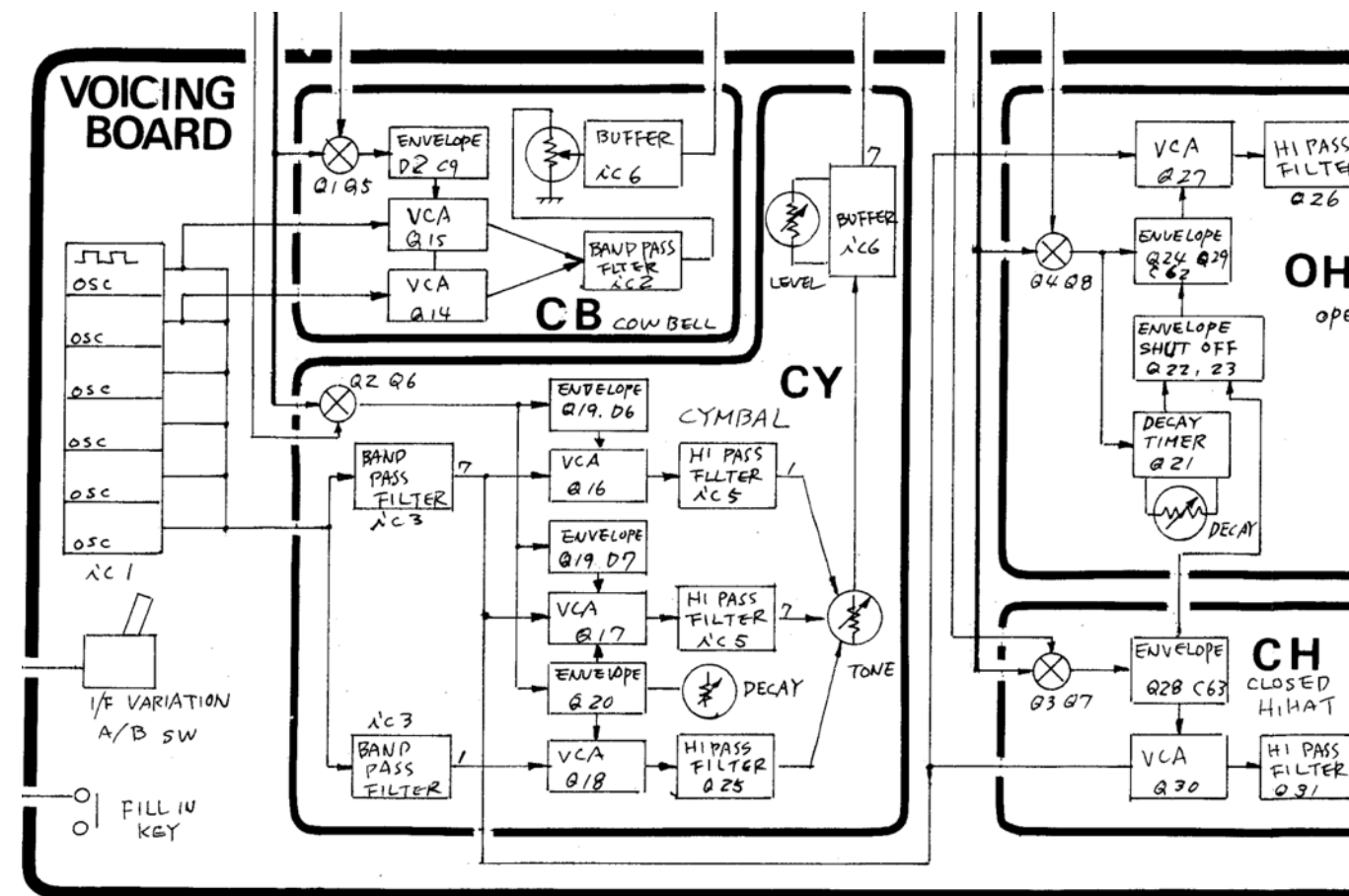
These three signals with different frequency ranges are outputted with their level ratio controlled by VR4.

OH

The high frequency range component signal obtained by the above 1/2 IC3 is gated by Q27 and supplied to the buffer IC7 through the filter Q26. When the CLOSED HI-HAT (CH) is triggered while the OH circuit is activated, Q23 turns on by the voltage applied through R173. At this moment, the decay time of the OH circuit terminates.

CH

This shares the same sound source with the OH. The signal is gated by Q30 and supplied to the filter Q31 and the buffer IC7 (1/2).



Synthesizing TR-808 style cymbals - design

that the sound of a real cymbal is not. So Roland made the decay rate of the

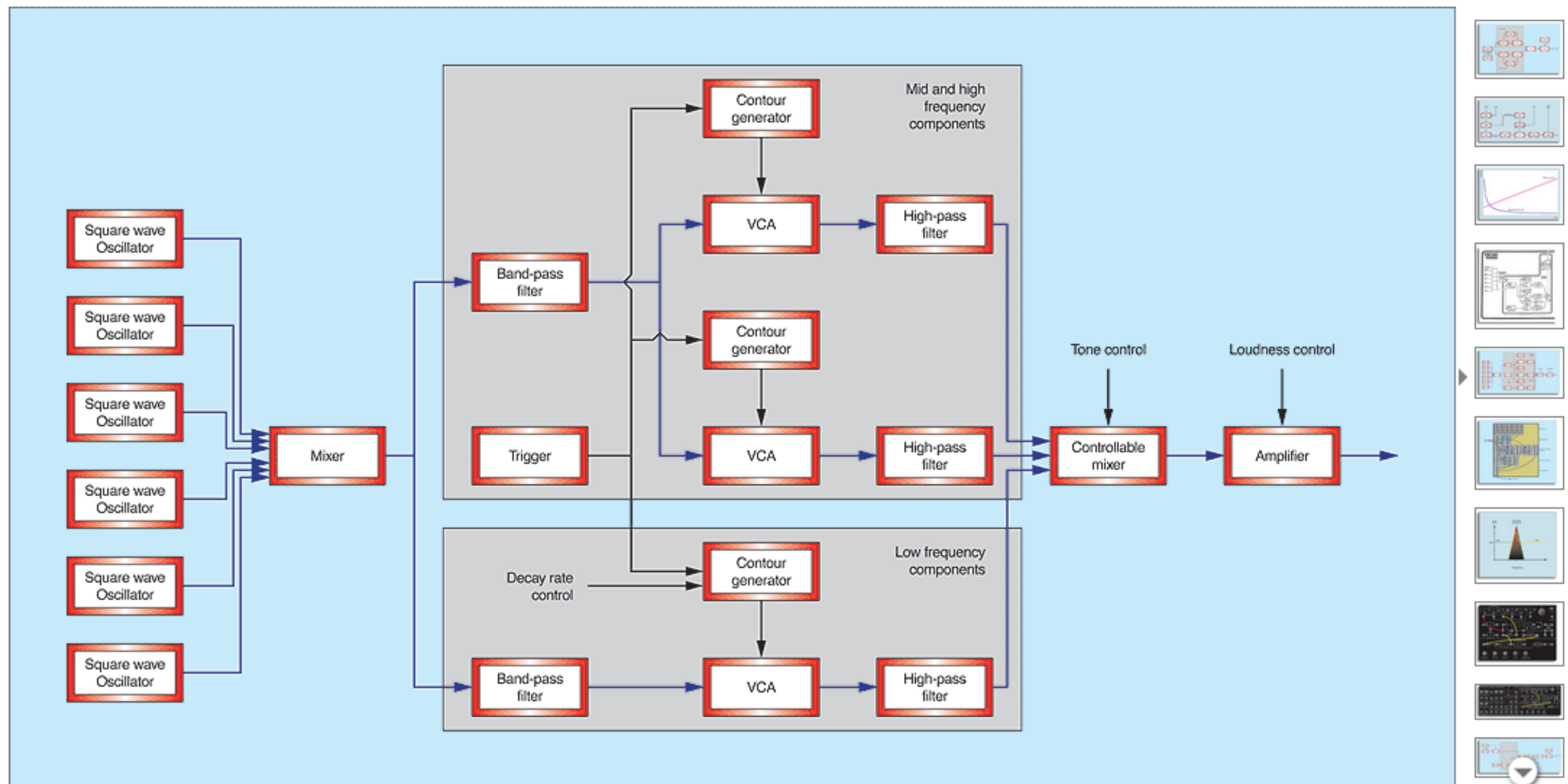


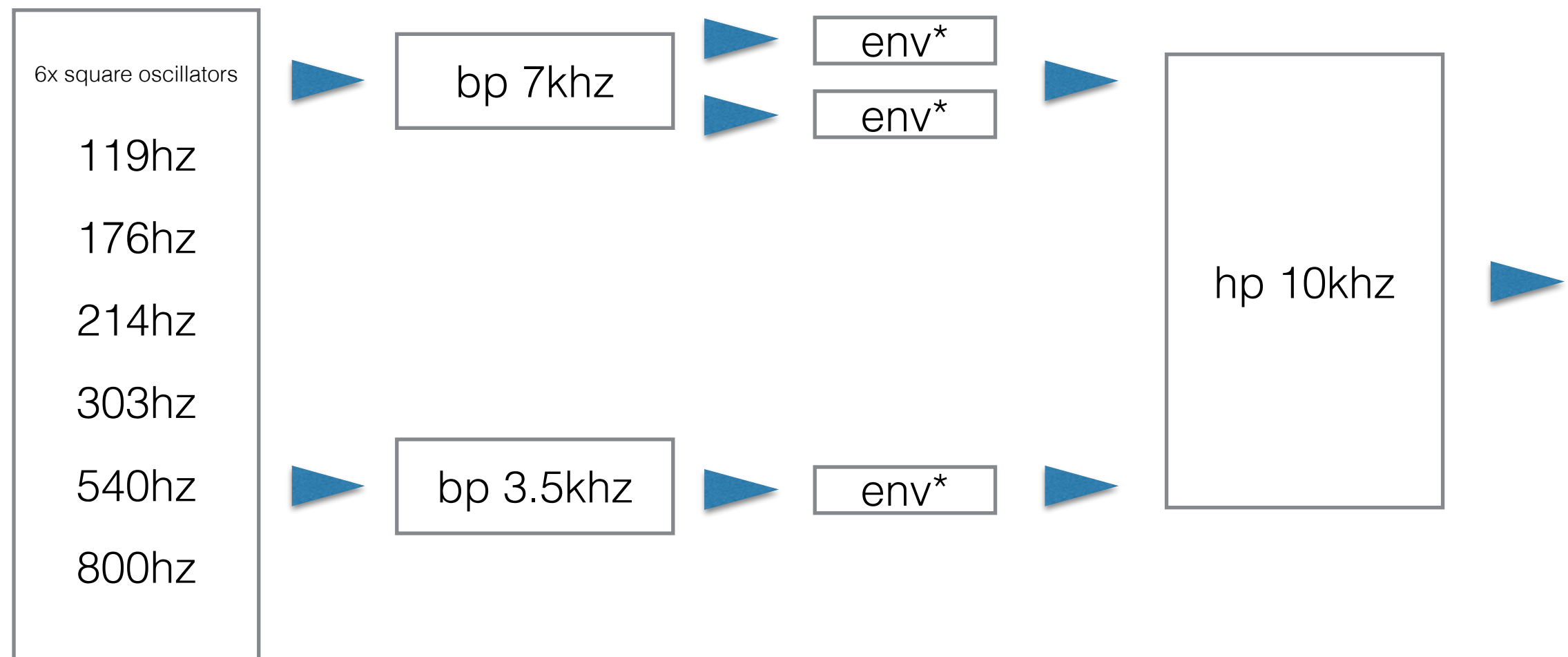
Image 5 of 13

bands has a Decay that lies somewhere near the centre of the range of the

Figure 6. Recreating the TR-808 cymbal

Synthesizing TR-808 style cymbals - design

here's how seqwww does it:



* The previous slide refers to this as a "Contour generator". Our version simply controls attack and decay times.

Synthesizing TR-808 style cymbals - filter section

```
// high pass filter connected to the two cymbal gates
var chHP = context.createBiquadFilter();
chHP.connect(chGate);
chHP.connect(ohGate);
chHP.type = "highpass";
chHP.frequency.value = 10000;

// high pass filter is fed by two gain elements that act as envelope generators.
var chHGain1 = context.createGain();
chHGain1.connect(chHP);
chHGain1.gain.value = 0;

var chHGain2 = context.createGain();
chHGain2.connect(chHP);
chHGain2.gain.value = 0;

var chHBP = context.createBiquadFilter();
chHBP.type = "bandpass";
chHBP.frequency.value = 7000;
chHBP.Q.value = 12;
chHBP.connect(chHGain1);
chHBP.connect(chHGain2);

// low bandpass filter
var chLBGain = context.createGain();
chLBGain.connect(chHP);
chLBGain.gain.value = 0;

var chLBP = context.createBiquadFilter();
chLBP.type = "bandpass";
chLBP.frequency.value = 3500;
chLBP.Q.value = 12;
chLBP.connect(chLBGain);
```

Synthesizing TR-808 style cymbals - oscillator section

```
var ch0sc1 = context.createOscillator();
ch0sc1.type = "square";
ch0sc1.frequency.value = 303;
ch0sc1.connect(chLBP);
ch0sc1.connect(chHBP);
ch0sc1.start(0);
```

```
var ch0sc2 = context.createOscillator();
ch0sc2.type = "square";
ch0sc2.frequency.value = 176;
ch0sc2.connect(chLBP);
ch0sc2.connect(chHBP);
ch0sc2.start(0);
```

```
var ch0sc3 = context.createOscillator();
ch0sc3.type = "square";
ch0sc3.frequency.value = 214;
ch0sc3.connect(chLBP);
ch0sc3.connect(chHBP);
ch0sc3.start(0);
```

```
var ch0sc4 = context.createOscillator();
ch0sc4.type = "square";
ch0sc4.frequency.value = 119;
ch0sc4.connect(chLBP);
ch0sc4.connect(chHBP);
ch0sc4.start(0);
```

```
var ch0sc5 = context.createOscillator();
ch0sc5.type = "square";
ch0sc5.frequency.value = 540;
ch0sc5.connect(chLBP);
ch0sc5.connect(chHBP);
ch0sc5.start(0);
```

```
var ch0sc6 = context.createOscillator();
ch0sc6.type = "square";
ch0sc6.frequency.value = 800;
ch0sc6.connect(chLBP);
ch0sc6.connect(chHBP);
ch0sc6.start(0);
```


Synthesizing TR-808 style cymbals - envelope section

```

this.DrumMachine.CH.Trigger = function(at) {
  ohGate.gain.value = 0; // silence open hat
  chGate.gain.value = 1; // open closed hat gate

  chHGain1.gain.setTargetAtTime(1 / 3, at, 0);
  chHGain1.gain.setTargetAtTime(0, at + .01, .01);

  chHGain2.gain.setTargetAtTime(1 / 3, at, 0);
  chHGain2.gain.setTargetAtTime(0, at + .02, .01);

  chLBGain.gain.setTargetAtTime(1 / 3, at, 0);
  chLBGain.gain.setTargetAtTime(0, at + .02, .01);
};

this.DrumMachine.OH.Trigger = function(at) {
  chGate.gain.value = 0; // silence closed hat
  ohGate.gain.value = 1; // open open hat gate

  chHGain1.gain.setTargetAtTime(1 / 3, at, 0);
  chHGain1.gain.setTargetAtTime(0, at + .5, 1);

  chHGain2.gain.setTargetAtTime(1 / 3, at, 0);
  chHGain2.gain.setTargetAtTime(0, at + .5, 1);

  chLBGain.gain.setTargetAtTime(1 / 3, at, 0);
  chLBGain.gain.setTargetAtTime(0, at + .5, 1);
};

```

		AMPLITUDE		FREQUENCY			DECAY TIME		
		NORMAL	ACCENT	LOW	MID	HIGH	SHORT	MID	LONG
		Vpp	Vpp	ms (Hz)	ms (Hz)	ms (Hz)	ms	ms	ms
BD		3.5	10	—	18 (56)	—	50	300	800
SD	H	3	10	—	2.1 (476)	—	—	60	—
	L				4.2 (238)				
LC		3.5	12	6.1 (165)	5.4 (185)	4.5 (220)	—	180	—
LT		3.5	12	12.5 (80)	11.1 (90)	10 (100)	—	200	—
MC		3	10	4 (250)	3.6 (280)	3.2 (310)	—	100	—
MT		3	11	8.3 (120)	7.4 (135)	6.3 (160)	—	130	—
HC		3.5	12	2.7 (370)	2.5 (400)	2.2 (455)	—	80	—
HT		3.5	12	6.1 (165)	5.4 (185)	4.5 (220)	—	100	—
C		2.5	8	—	0.4 (2500)	—	—	25	—
RS	H	3	10	—	0.6 (1667)	—	—	10	—
	L				2.2 (455)				
M		3	5	—	—	—	25	—	35
CP		6	2	—	—	—	—	100	—
CB	H	3.5	12	—	1.25 (800)	—	—	50	—
	L				1.85 (540)				
CY		3.5	7	—	—	—	350	800	1200
OH		3.5	7	—	—	—	90	450	600
CH		3	6	—	—	—	—	50	—

values are typical and variable

Web audio oscillator implementation in WebKit

OWNERS

- OfflineAudioCompletionEvent.cpp
- OfflineAudioCompletionEvent.h
- OfflineAudioCompletionEvent.idl
- OfflineAudioContext.cpp
- OfflineAudioContext.h
- OfflineAudioContext.idl
- OfflineAudioDestinationNode.cpp
- OfflineAudioDestinationNode.h
- OscillatorNode.cpp**
- OscillatorNode.h
- OscillatorNode.idl
- PannerNode.cpp
- PannerNode.h
- PannerNode.idl
- PeriodicWave.cpp
- PeriodicWave.h

```
291 while (n--) {
292     unsigned readIndex = static_cast<unsigned>(virtualReadIndex);
293     unsigned readIndex2 = readIndex + 1;
294
295     // Contain within valid range.
296     readIndex = readIndex & readIndexMask;
297     readIndex2 = readIndex2 & readIndexMask;
298
299     if (hasSampleAccurateValues) {
300         incr = *phaseIncrements++;
301
302         frequency = invRateScale * incr;
303         m_periodicWave->waveDataForFundamentalFrequency(frequency, lowerWaveData, higherWaveData, tableInterpolationFactor);
304     }
305
306     float sample1Lower = lowerWaveData[readIndex];
307     float sample2Lower = lowerWaveData[readIndex2];
308     float sample1Higher = higherWaveData[readIndex];
309     float sample2Higher = higherWaveData[readIndex2];
310
311     // Linearly interpolate within each table (lower and higher).
312     float interpolationFactor = static_cast<float>(virtualReadIndex) - readIndex;
```

Web audio oscillator implementation in WebKit

OfflineAudioContext.cpp
OfflineAudioContext.h
OfflineAudioContext.idl
OfflineAudioDestinationNode.cpp
OfflineAudioDestinationNode.h
OscillatorNode.cpp
OscillatorNode.h
OscillatorNode.idl
PannerNode.cpp
PannerNode.h
PannerNode.idl
PeriodicWave.cpp
PeriodicWave.h
PeriodicWave.idl
RealtimeAnalyser.cpp
RealtimeAnalyser.h
ScriptProcessorNode.cpp
ScriptProcessorNode.h
ScriptProcessorNode.idl
WaveShaperDSPKernel.cpp
WaveShaperDSPKernel.h
WaveShaperNode.cpp
WaveShaperNode.h
WaveShaperNode.idl
WaveShaperProcessor.cpp

```
148 // Convert into time-domain wave buffers.
149 // One table is created for each range for non-aliasing playback at different playback rates.
150 // Thus, higher ranges have more high-frequency partials culled out.
151 void PeriodicWave::createBandLimitedTables(const float* realData, const float* imagData, unsigned numberOfComponents)
152 {
153     float normalizationScale = 1;
154
155     unsigned fftSize = m_periodicWaveSize;
156     unsigned halfSize = fftSize / 2;
157     unsigned i;
158
159     numberOfComponents = std::min(numberOfComponents, halfSize);
160
161     m_bandLimitedTables.reserveCapacity(m_numberOfRanges);
162
163     for (unsigned rangeIndex = 0; rangeIndex < m_numberOfRanges; ++rangeIndex) {
164         // This FFTFrame is used to cull partials (represented by frequency bins).
165         FFTFrame frame(fftSize);
166         float* realP = frame.realData();
167         float* imagP = frame.imagData();
168
169         // Copy from loaded frequency data and scale.
170         float scale = fftSize;
171         vsmul(realData, 1, &scale, realP, 1, numberOfComponents);
172         vsmul(imagData, 1, &scale, imagP, 1, numberOfComponents);
173
174         // If fewer components were provided than 1/2 FFT size, then clear the remaining bins.
175         for (i = numberOfComponents; i < halfSize; ++i) {
176             realP[i] = 0;
177             imagP[i] = 0;
178         }
179
180         // Generate complex conjugate because of the way the inverse FFT is defined.
```

Web audio oscillator implementation in WebKit

OfflineAudioContext.cpp
OfflineAudioContext.h
OfflineAudioContext.idl
OfflineAudioDestinationNode.cpp
OfflineAudioDestinationNode.h
OscillatorNode.cpp
OscillatorNode.h
OscillatorNode.idl
PannerNode.cpp
PannerNode.h
PannerNode.idl
PeriodicWave.cpp
PeriodicWave.h
PeriodicWave.idl
RealtimeAnalyser.cpp
RealtimeAnalyser.h
ScriptProcessorNode.cpp
ScriptProcessorNode.h
ScriptProcessorNode.idl
WaveShaperDSPKernel.cpp
WaveShaperDSPKernel.h
WaveShaperNode.cpp
WaveShaperNode.h
WaveShaperNode.idl
WaveShaperProcessor.cpp

```
180 // Generate complex conjugate because of the way the inverse FFT is defined.
181 float minusOne = -1;
182 vsmul(imagP, 1, &minusOne, imagP, 1, halfSize);
183
184 // Find the starting bin where we should start culling.
185 // We need to clear out the highest frequencies to band-limit the waveform.
186 unsigned numberOfPartials = numberOfPartialsForRange(rangeIndex);
187
188 // Cull the aliasing partials for this pitch range.
189 for (i = numberOfPartials + 1; i < halfSize; ++i) {
190     realP[i] = 0;
191     imagP[i] = 0;
192 }
193 // Clear packed-nyquist if necessary.
194 if (numberOfPartials < halfSize)
195     imagP[0] = 0;
196
197 // Clear any DC-offset.
198 realP[0] = 0;
199
200 // Create the band-limited table.
201 OwnPtr<AudioFloatArray> table = adoptPtr(new AudioFloatArray(m_periodicWaveSize));
202 m_bandLimitedTables.append(table.release());
203
204 // Apply an inverse FFT to generate the time-domain table data.
205 float* data = m_bandLimitedTables[rangeIndex]->data();
206 frame.doInverseFFT(data);
207
208 // For the first range (which has the highest power), calculate its peak value then compute normalization scale.
209 if (!rangeIndex) {
210     float maxValue;
211     vmaxmgv(data, 1, &maxValue, m_periodicWaveSize);
212 }
```

Thanks!

<https://github.com/ceberly/seq-www>

Chris Eberly - Gobbler engineering -
chris.eberly@gobbler.com - @chriseberly