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Performing STFT and ISTFT in the Microsound Synthesis Framework of the LC Computer Music Programming Language

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Abstract: This paper describes how the short-term Fourier transform (STFT) and inverse short-term Fourier transform (ISTFT) are integrated within the sound synthesis framework of LC, a new computer music programming language, which the authors prototyped, and discusses its benefits for computer music programming. In addition to the traditional unit-generator-based sound synthesis framework, LC provides a framework for microsound synthesis, which is highly independent from the unit-generator concept, and STFT and ISTFT can be also performed within the same framework. While the unit-analyzer concept in the ChucK audio programming language shows a certain degree of similarity to LC's programming model for STFT and ISTFT, in that both languages allow direct access to low-level spectral data from user programs, due to the dependence on the unit-generator-based sound synthesis framework, a ChucK program that utilizes unit analyzers can exhibit unnecessary complexity in its implementation, when the hop sizes differ among the STFT frames in the program. On the other hand, thanks to the high independence from the unit-generator concept, LC's microsound synthesis framework can provide a simpler and terser programming model and avoid such unnecessary complications. As other unit-generator languages can also exhibit similar problems as seen in ChucK's unit analyzers, depending on its sound synthesis framework design, such a language design of LC would be beneficial, not just as a design exemplar for next generation computer music languages, but also to reconsider the design of existing unit-generator languages on such issues regarding how STFT should be integrated in a unit-generator language and whether unit-generators should fully synchronize the audio computation with the advance of global system time.

Keywords: computer music, unit generator, unit analyzer, spectral sound processing, programming language

1. Introduction

The short-term Fourier transform (STFT) is already an essential feature that a computer music language is expected to support. Various STFT-based sound synthesis and processing techniques are involved in today's creative practices in computer music [3], [18]. Yet, in unit-generator languages, extra care is often required in programming when performing STFT. For instance, in a programming environment that does not provide any useful features to facilitate the implementation of the overlap-add method for FFT frames, users have to make extra effort to write their own code to realize overlapping FFT frames. Furthermore, in many computer music programing languages and environments, spectral processing and feature extraction techniques that utilize STFT are normally encapsulated within low-level built-in modules written in other languages (e.g., C or C++), since the algorithms are often hard to describe just by combining unit-generators.

Some computer music languages try to overcome such deficits by allowing direct access to low-level data within FFT frames. Nyquist [6] is one of the earliest examples of a computer music language of this kind. Such an approach is beneficial in that users can describe desired spectral processing and analysis algorithms within the language itself, without writing built-in modules in another language. While Nyquist is basically designed for non-real-time sound synthesis and analysis, ChucK [18] is an example of a real-time, interactive computer music language that allows direct access to low-level frame data through its *unit analyzers* [19].

A new interactive real-time computer music language we prototyped, LC [9], [10], is another example designed with a similar idea. LC's microsound synthesis framework is highly independent of the traditional unit-generator concept and significantly focuses on microsound synthesis (as its name indicates). Unlike most existing computer music languages, LC can perform STFT and ISTFT within this microsound synthesis framework, without involving unit generators and even without the advance of logical time. In addition, LC provides the unit-generator-based sound synthesis framework and the seamless collaboration between these two frameworks with significant different abstractions. Such language design of LC provides simple and terse programming models for spectral processing as well, whereas ChucK and other unit-generator languages may involve a considerable degree of complexity in the resulting code in certain situations, as we describe in later sections.

Generally speaking, as ChucK's unit analyzers still highly de-

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pend on the unit-generator-based sound synthesis framework, the implementation can exhibit a considerable degree of complication, even for simple STFT related techniques when the hop sizes differ among STFT and ISTFT frames. This problem is largely due to the design of ChucK's sound synthesis framework, in which all unit generators (and unit analyzers) synchronize the audio computation with the advance of the system's global logical time. Hence, the same kind of problems may easily occur in other unit-generator languages with similar software framework design; thus, the language design of LC can benefit not just by offering a design exemplar for next generation computer music languages but also by reconsidering the design of unit-generator-based sound synthesis frameworks.

2. Related Work

2.1 STFT in Unit-generator Languages

As previously mentioned, to perform STFT-related techniques in a unit-generator language, extra effort is often required to realize the overlap-add method. **Figure 1** describes a typical example in Max to perform STFT [4]. The example involves two *fft~* objects to overlap-add two FFT frames [*1] and the windowing is realized by multiplying the incoming audio input by the output signals of one *cycle~* object [*2] for each FFT frame, respectively. When more FFT frames must be overlapped, more objects must be utlized, and the arguments must be also set up according to the number of overlapping frames. Thus, the implementation can be more complicated when increasing the number of overlapping FFT frames.

Since such tasks (i.e., windowing of incoming samples, buffering of the samples for FFT, and overlap-adding of FFT frames) are normally involved when STFT is performed, many computer

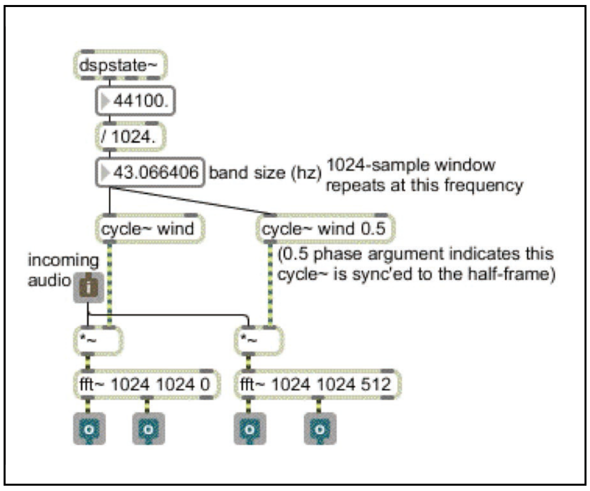

Fig. 1 STFT example in Max [4].

*1 As for Max's *fft~* object, "the first argument specifies the number of points (samples) in the FFT" and "the second argument specifies the number of samples between successive FFTs". "The third argument specifies the offset into the interval where the FFT will start" [5]. Thus, in Fig. 1, the FFT frame sizes for both *fft~* objects are 1024 and the FFT are performed successively with the interval of 1024 samples, where the *fft~* object on the right is given the offset of 512 to overlap-add two FFT frames.

*2 There are two objects and one is given the argument of 0.5 ('*cycle wind 0.5*') so that the *cycle~* object can have a phase difference of half a period from the other object.

music languages provide special objects that encapsulate these tasks to allow users to focus on the implementation of desired algorithms. For instance, Max provides the *pfft~* object and SuperCollider [20] offers the FFT/IFFT unit generators, as described in Ref. [19].

2.2 Accessibility of Low-level Audio Data

While such a strategy to provide certain objects to encapsulate common tasks would be quite useful to facilitate the implementation of STFT-related techniques, the actual spectral processing and analysis tasks must be performed after obtaining FFT frame data. Generally, however, "the analysis functionality of these systems relies on pre-made, black-box objects (e.g., coded and imported from C/C++) to meet the needs for common specific analysis tasks" in many unit-generator languages, as Wang et al. discussed in Ref. [19].

Brandt indicated that the unit-generator concept significantly depends on black box abstraction, and considered it a significant problem in computer music language design, since "if a desired operation is not present, and cannot be represented as a composition of primitives, it cannot be realized within the language" [1, p.4]. As many algorithms for spectral processing and analysis can be complicated and often hardly realizable through just the combination of unit generators, the black-box abstraction nature of the traditional unit-generator concept can be problematic, when a desired operation is not provided as a unit generator, as Brandt discussed.

Thus, to make a computer music language more expressive in spectral processing and analysis, it is desirable to allow direct access to low-level data in each frequency bin and to provide a means to describe a desired algorithm within the language itself. Nyquist [6] is a precursor example (and possibly the first language) that is designed with such a concept. Brandt's Chronic computer music language [1] is another good example, which provides users with accessibility to low-level audio data and the expressiveness to describe desired algorithms. While these two works are of significant interest as design exemplars, these two languages are, however, non-real-time sound synthesis languages and are not designed for real-time interactive computer music.

2.3 ChucK Audio Programming Language and Its Unit-analyzer Concept

The unit-analyzer concept in the ChucK audio programming language [19] provides a good design exemplar for a real-time interactive computer music language with accessibility to low-level spectral data and expressiveness of desired spectral processing algorithms. Yet, before getting into the details of the unit-analyzer concept, we briefly describe ChucK's strongly-timed programming concept, which realizes the sample-rate accuracy in timing behavior. This programming concept is important in order to understand the code examples of the unit analyzers given in the following sections.

2.3.1 Strongly-timed Programming

Broadly speaking, the strongly-timed programming concept is a variation of *synchronous programming*, which is based on the *ideal synchronous hypothesis*. In the ideal synchronous hy-

```
// synthesis patch
SinOsc foo => dac;

// infinite time loop
while(true)
{
  // randomly choose a frequency
  Std.rand2f(30, 1000) => foo.freq;
  // advance time
  100::ms => now;
}
```

Fig. 2 Strongly-timed programming example in ChucK [18].

pothesis, “all the computation and communication is assumed to take zero time (that is, all temporal scopes are executed instantaneously)” and “during implementation, the ideal synchronous hypothesis is interpreted to imply [that] the system must execute fast enough for the effect of the synchronous hypothesis to hold” [2, p.360].

While many other synchronous programming languages are designed for reactive-systems [*3], ChucK adopts the concept of synchronous programming into an imperative programming language for interactive systems, with a significant focus on precise timing behavior. In strongly-timed programing, a user program can explicitly control the advance of its logical synchronous time, otherwise the logical time will not be advanced at all. In ChucK, the audio computation is performed only when the user program explicitly advances the logical time. **Figure 2** describes a simple strongly-timed program example, as Wang described in Ref. [18, p.43] [*4].

Such an issue of timing precision can also be important in spectral processing and analysis algorithms. If the audio computation is performed ‘out-of-sync’ with a spectral processing/analysis algorithm, the result of the computation will be different from what users expect.

2.3.2 Unit-analyzer Concept

To support more expressiveness in spectral processing and analysis, ChucK introduces the concept of the unit analyzer, “which carries with it a set of operations and a connection model that resemble but are distinct from those of a unit-generator” [19]. The unit-analyzer concept resembles the unit-generator concept in that it follows the similar connection model, and a unit analyzer (*UAna, plural UAnae*) “defines a set of control parameters and can be dynamically patched with other UAnae and UGens”. The difference is that unit analyzers “pass generic data that may be in the form of spectral frames, feature vectors, metadata, or any other (intermediate) products of analysis” [19] whereas unit

```
01: //load the sound files to the buffer objects.
02: “/sound/violin.wav”    => string filename1;
03: “/sound/cherokee.aif”  => string filename2;
04: SndBuf src1;
05: SndBuf src2;
06: filename1 => src1.read;
07: filename2 => src2.read;
08:
09: //build a synthesis graph.
10: src1 => FFT fft1 => blackhole;
11: src2 => FFT fft2 => blackhole;
12: IFFT ifft => dac;
13: 800 => ifft.gain; //gain for ifft.
14:
15: //set up FFT parameters.
16: 1024 => fft1.size => fft2.size => ifft.size
17: => int FFT_SIZE;
18: FFT_SIZE / 2 => int HOP_SIZE;
19:
20: Windowing.hann(FFT_SIZE) => fft1.window;
21: Windowing.hann(FFT_SIZE) => fft2.window;
22: Windowing.hann(FFT_SIZE) => ifft.window;
23:
24: //to store the cross synthesis result.
25: complex Z[FFT_SIZE / 2];
26:
27: //main loop.
28: while(true){
29:   //perform FFT for two inputs.
30:   fft1.upchuck();
31:   fft2.upchuck();
32:
33:   //cross synthesis
34:   for (0 => int i; i < fft1.size() / 2; i++){
35:     fft1.cval(i) $ polar => polar a;
36:     fft2.cval(i) $ polar => polar b;
37:     %(a.mag * b.mag, a.phase) => polar c;
38:     c $ complex => Z[i];
39:   }
40:
41:   //perform IFFT.
42:   ifft.transform(Z);
43:
44:   //sleep until the next frame.
45:   HOP_SIZE::samp => now;
46: }
```

Fig. 3 Cross synthesis example in ChucK.

generators stream the audio samples.

ChucK’s unit analyzers are designed to work within the unit-generator-based sound synthesis framework. One of the benefits of the unit-analyzer concept is that the various types of data can be obtained from the unit-generator outputs and even low-level data are made directly accessible from a user program. For instance, in the **Fig. 3** example of cross synthesis [16] in ChucK, the FFT unit analyzers provide direct access to each frequency bin (lines 35–36) and the algorithm of cross synthesis can be performed only by the user code. Unlike many other unit-generator languages, there is no necessity to involve a unit-generator object designed for cross synthesis.

While ChucK is not the first language that allows the direct access to low-level data as already described earlier in this article, such a design of the unit analyzers contributes to avoiding the problem, as Brandt discussed. Even if a desired operation is not present, users can realize it by writing their own code in ChucK.

Yet, since ChucK’s unit-analyzer concept significantly depends on its unit-generator-based sound synthesis framework, such dependency can lead to usability difficulties when the hop sizes differ among the FFT and IFFT frames, as described in detail in the later section.

[*3] Reactive systems are “computer systems that continuously react to their environment at a speed determined by this environment”, while interactive systems “continuously interact with their environment, but at their own rate” [8].

[*4] An approach similar to synchronous programming can be seen in other computer music programming languages. For example, LuaAV [17] utilizes coroutines in its sound synthesis framework to realize synchronous behavior. For another example, PureData (Pd) [14] also adopts a synchronous approach. In PureData, “audio and message processing are interleaved in Pd” [13]. In other words, the audio computation is blocked until the system finishes processing all the events at the timing. It should be noted, however, that strongly-timed programming is a programming concept based on the ideal synchronous hypothesis, while these languages realize similar behavior by implementation.

3. STFT in the Microsound Synthesis Framework of LC

3.1 Microsound Synthesis Framework of LC

Before we describe how STFT can be performed in LC, we provide a brief overview of its microsound synthesis framework, in which STFT modules are integrated. While LC is a computer music programming language, first developed as a host language for the LCSynth sound synthesis language [11], [12], significant extension has been made to its language specification until the current version [*5].

While LC is still equipped with the unit-generator-based sound synthesis framework, it also offers the microsound synthesis framework, which is highly independent of the unit-generator concept, together with the seamless collaboration mechanism between these two different sound synthesis frameworks.

In LC, there are two objects utilized to represent microsounds: *Samples* and *SampleBuffer* objects. Both of these objects can contain an arbitrary number of sample values (as long as enough memory can be allocated in runtime). The difference between *Samples* and *SampleBuffer* is that the former is an immutable object and the latter is mutable [*6]. **Figure 4** briefly describes how *Samples* and *SampleBuffer* can be created. Both objects have various methods for manipulations. As *Samples* is immutable, calling these methods return new *Sample* objects, and the original object will remain unchanged. *Samples* and *SampleBuffer* are mutually convertible by the *toSampleBuffer* and *toSamples* methods, respectively.

Figure 5 describes an example of indexed access. Each sample within these objects is directly accessible by the '[]' operator. While *Samples* is immutable, the samples within *SampleBuffer* can be changed by assigning a new value by indexed access.

Figures 6 and **7** describe examples of synchronous granular synthesis [*7] and pitch shifting by granulation [15, p.197], respectively. As shown in these, LC can perform microsound synthesis without involving unit generators at all (as on lines 01–28 in Fig. 7), while it is also possible to generate a *Samples* object from a unit generator or a patch (lines 15–16 and lines 22–27 in Fig. 4, respectively) and to give a *Samples* object to a unit generator or a patch as the input signal to its inlet (lines 30–51 in Fig. 7). It should be noted that LC is a strongly-timed language with sample-rate accurate timing behavior (line 13 in Fig. 6 and line 49 in Fig. 7).

```
01: //create a new Samples obj from the buf no 0.
02: LoadSndFile(0, "/sound/sample1.aif");
03: var snd = ReadBuf(0, 256::samp);
04:
05: //create another by generating a window.
06: var win = GenWindow(512::samp, \hanning);
07:
08: //create Samples objects by the method calls.
09: var grain    = snd->applyEnv(win);
10: var halfAmp  = snd->amplify (0.5);
11: var octup    = snd->resample(snd.size / 2);
12: var reversed = snd->reverse();
13:
14: //use a unit-generator to create a Samples obj.
15: var sin = new Sin~(freq:440);
16: var out = sin->pread(0.5::second);
17: PanOut(out, 0); //output it directly to the DAC.
18: now += 1::second;
19:
20: //a patch can also generate a Samples obj.
21: var dur = 1::second;
22: var pat = patch {
23:   Sin~(freq:440) => env:TriEnv~(dur)
24:   => defout:Outlet~();
25: };
26: pat.env->trigger();
27: out = pat->pread(dur);
28: PanOut(out, 0);
29:
30: //convert it to a SampleBuffer obj.
31: var sbuf = snd->toSampleBuffer();
32: //convert the SampleBuffer back to a Samples obj.
33: var snd2 = sbuf->toSamples();
34:
35: //create a new SampleBuffer by 'new' operator.
36: var sbuf2 = new SampleBuffer(128);
```

Fig. 4 Samples and SampleBuffer objects in LC.

3.2 Short-term Fourier Transform in LC

As described in Fig. 3, *Samples* and *Samples* objects in LC allow direct access to low-level sample data. Such a feature of LC's microsound synthesis framework can also be beneficial to perform STFT. While the current version of LC offers only basic STFT-related library functions as shown in **Table 1**, this accessibility to low-level data allows users to describe their desired operations within just LC, similarly to ChucK's unit analyzers.

Figures 8 and **10** describe simple examples of cross synthesis and time-stretching by phase vocoding, respectively. While the code uses the *mul* method of *Samples* to multiply the magnitudes

```
01: //create a new SampleBuffer object (size = 128)
02: var sb = new SampleBuffer(128);
03:
04: //assigning sample values by indexed-access.
05: for (var i = 0; i < sb.size; i += 1){
06:   sb[i] = i * 2;
07: }
08:
09: //convert it to a Samples object.
10: var snd = sb->toSamples();
11: //Samples is immutable. Read-only.
12: for (var i = 0; i < snd.size; i += 1){
13:   println("snd[" .. i .. "] = ".. snd[i]);
14: }
```

Fig. 5 Indexed-access example in LC.

```
01: //generate  Samples obj from the ugen output.
02: var sin = new Sin~(freq:SampleRate / 256.0);
03: var tmp = sin->pread(1024::samp);
04:
05: //apply a hanning window to make a grain.
06: var win = GenWindow(tmp.dur, \hanning);
07: var grn = tmp->applyEnv(win)->resample(440);
08:
09: //perform sync gran synth for 5 second.
10: within(5::second) while(true){
11:   PanOut(grn, 0.0); //0.0 = center
12:   //sleep until the next grain sched timing.
13:   now += grn.dur / 4;
14: }
```

Fig. 6 Synchronous granular synthesis example in LC.

[*5] Our paper [10] gives a brief overview of the language features of LC.
[*6] This difference between *Samples* and *SampleBuffer* in LC is similar to the difference between *String* and *StringBuf* in Java [7].
[*7] In synchronous granular synthesis, the sound "results from one or more stream of grain" and "the grains follow each other at regular intervals" [15, p.93].

```
//load a sound file and set up the parameters.
LoadSndFile(0,  "/sound/cherokee.aif ");
var pitch   = 2;          //shift octave upper
var rpos    = 0::second; //the buf read pos
var grnsize = 512;
var grndur  = grnsize::samp;
var win = GenWindow(grndur, \hanning);
var rdur = grndur * pitch;

var entireDur = 2::second;

//perform granular pitch-shifting.
within(entireDur) while(true){
  //get a snd fragment from the buffer.
  var snd = ReadBuf(0, rdur, offset:rpos);

  //resample to pitch-shift. apply an envelope.
  var tmp = snd->resample(grnsize);
  var grn = tmp->applyEnv(win);

  //send the grain to the DAC.
  WriteDAC(grn);

  //advance the time & the read pos.
  //change the inc value for time-stretching.
  now  += grn.dur / 2;
  rpos += grn.dur / 2;
}

//now, we apply the triangle envelope and
//the reverberation, using a patch.
//first, let' s create a patch.
var p = patch {
  defin:Inlet~() => env:TriEnv~(entireDur)
  => Freeverb~()  => DAC~();
};
p.env->trigger(); //trigger the envelope.
p->start();         //play the patch right now.

//perform the same algorithm, but with a patch.
rpos = 0::second; //reset rpos.
within(entireDur) while(true){
  var snd = ReadBuf(0, rdur, offset:rpos);
  var tmp = snd->resample(grnsize);
  var grn = tmp->applyEnv(win);

  //this time, write to a patch instead.
  p->write(grn);
  now  += grn.dur / 2;
  rpos += grn.dur / 2;
}
```

Fig. 7 Granular pitch-shifting example in LC.

```
01: //load two sound files to buf 0 and buf1.
02: LoadSndFile(0,  "/sound/violin.wav" );
03: LoadSndFile(1,  "/sound/cherokee.aif" );
04:
05: //the widnow size and the num of overlap-add.
06: var dur   = 1024::samp;
07: var ovlp1 = 2, ovlp2 = 2;
08:
09: //process 800 frames.
10: for (var i = 0; i < 800; i += 1){
11:   //read snd fragments from the buffers.
12:   var src1 = ReadBuf(0, dur, offset:i * dur / ovlp1);
13:   var src2 = ReadBuf(1, dur, offset:i * dur / ovlp2);
14:
15:   //PFFT returns an array of Samples objects,
16:   //[magnitude, phase].
17:   var pfft1 = PFFT(src1, \hanning);
18:   var pfft2 = PFFT(src2, \hanning);
19:
20:   //perform cross synthesis.
21:   var ppved = pfft1[0]->mul(pfft2[0]);
22:
23:   var pifft = PIFFT(ppved, pfft1[1], \hanning);
24:
25:   PanOut(pifft, 0.0); //center = 0.0
26:
27:   now += src1.dur / ovlp1;
28: }
```

Fig. 8 Cross synthesis example in LC.

```
var tmp = new SampleBuffer(dur / 1::samp);
for (var i = 0; i < pfft1[0].size; i +=1){
  tmp[i] = pfft1[0][i] * pfft2[0][i];
}
var ppved = tmp->toSamples();
```

Fig. 9 Performing cross-synthesis using a for loop.

```
//load two sound files to buf 0.
LoadSndFile(0,  "src0.wav" );

//set up the parameters.
var dur  = 1024::samp;     //the FFT frame size
var ovlp = 4;              //overlap x 4
var strc = 4;    //stretch x 4

//set up the initial phase.
var firstFrame = ReadBuf(0, dur);
var ffted = PFFT(firstFrame);

var prev = ffted[1];
var accum= prev;

//perform time-stretching for 24 second.
within(24::second) while(true){
  //read one more frame and perform FFT.
  var snd  = ReadBuf(0, dur, offset:pos);
  var ffted= PFFT(snd);

  //calc the phase diff and accumulate it.
  var dif = ffted[1]->sub(prev)->amplify(strc);
  accum = accum->accumulatePhase(dif);

  //perform IFFT with the accumulated phase.
  var ifft = PIFFT(ffted[0], accum);
  PanOut(iffted, 0.0); //send to the DAC output

  //update prev phase
  prev = ffted[1];

  //update pos and sleep until the next frame.
  pos += dur / (ovlp * strc);
  now += dur / ovlp;
}
```

Fig. 10 Phase vocoder example (time-stretching) in LC.

Table 1 Library functions for STFT in LC.

FFT(var samples)
IFFT(var real, imag)
FFT (The current version just perform real FFT only) and Inverse Fast Fourier Transform. FFT returns an array of [real, imag].

CarToPol(var real, imag)
PolToCar(var mag, phase)
conversion between Cartesian coordinate and polar coordinate

PFFT(var samples, window) **PIFFT(var mag, phase, window)** FFT/IFFT with windowing. PIFFT returns the frame data in polar coordinate. PFFT returns an array of [mag, phase].

of two FFT frames in Fig. 8 (on line 21), the same operation can be performed entirely in the user code as in **Fig. 9**, which can replace line 21 in Fig. 8. Likewise, the other methods (*sub, amplify, accumulatePhase)* in Fig. 9 can also be expressed entirely within the user code.

Thus, LC also provides direct access to low-level data together with good expressiveness of the algorithms to perform spectral synthesis and analysis. Moreover, in LC, STFT can be performed within just its microsound synthesis framework without involving any unit generator, whereas ChucK's unit analyzers still significantly depend on its unit-generator-based sound synthesis framework. Such independence from the unit-generator concept is one of the key factors in the language design of LC that makes it possible to avoid unnecessary complexity in the implementation (in other words, usability difficulties in programming), which ChucK and other unit-generator languages with similar designs exhibit in certain situations. We discuss this issue in more detail in the next section.

4. Discussion

One of the typical problems in performing STFT within the unit-generator-based sound synthesis framework is the difficulty in implementing windowing and overlap-adding. The Fig. 1 example in Max exhibits this typical problem; windowing must be implemented and multiple FFT objects must be utilized to implement overlapping FFT frames. Furthermore, the time must be advanced to feed the input samples. This can introduce latency, even when all the sample data is already loaded in the buffer.

While the problems as above (except the latency) can be solved by providing a utility object that automatically performs buffering, windowing and overlap-adding all at once, (e.g., *pfft~* in Max and the *FFT* unit generator in Supercollider), there remains a serious problem of the expressiveness of the related algorithms and accessibility to lower-level data, as Brandt and Wang discussed, which is described in Section 2. Based on the idea similar to Nyquist and Chronic, both of which are non-real-time computer music languages, ChucK's unit-analyzer concept seems to succeed in providing one solution for an interactive real-time computer music language by allowing direct access to low-level spectral data. Strongly-timed programing in ChucK also provides sample-rate accurate synchronization between the algorithms written within the language and the unit-generator-based sound synthesis. Thus, Chuck achieves a considerable degree of expressiveness in spectral processing and analysis; however, as it is highly dependent on the unit-generator-based sound synthesis framework, the latency issue remains. As it is still required to advance the logical time to feed the input samples to the *FFT* unit analyzer, the latency of the FFT frame size must be involved.

To make matters worse, the dependency on the unit-generator-based framework can cause another problem in ChucK when the hop sizes can differ among the FFT frames. Think of the situation in which two FFT objects are utilized in cross synthesis and the hops sizes differ between them. For instance, one may want to use a smaller hop size for one of the sound sources (from which we extract its formant for cross synthesis), as the duration of the sound source is shorter than the other. **Figure 11** describes such a ChucK example in which the hop sizes for two input sources differ. As presented, the code is much more complicated than the Fig. 3 example.

To comprehend this example better, consider the situations as follows. First, we discuss what is performed in the previous example in Fig. 3. In Fig. 3, if the time is advanced for 512 samples (the hop size for *IFFT*), both the *FFT* unit analyzers (*fft1* and *fft2*) discard the oldest 512 samples in the internal buffers and receive the input of 512 new samples from their sound sources (*src1* and *src2*). Thus, the hop sizes for these *FFT* unit analyzers become the same as *IFFT*.

Now, suppose that we want to read the second sound source at half speed. To implement such a program, one has to change the hop size for *fft2* to 256 samples but still keep the hop sizes for *fft1* and *ifft* at 512 samples. In this case, *fft2* must discard only the oldest 256 samples and accept 256 new input samples instead, but *fft1* and *ifft* still require the advance of 512 samples. For this reason, one must utilize two pairs of *SndBuf* and *FFT* for the second sound source (*source2a* and *fft2a, source2b* and *fft2b*) as in the Fig. 11 example, and then update the reading positions of each *SndBuf* so that the samples can be fed from the correct positions of the buffers, according to the given hop sizes. This problem is indeed rooted in the design of ChucK's unit-generator-based sound synthesis framework, in which all the unit generators and unit analyzers synchronize their audio computations with one global system time. Hence, a similar problem can also be exhibited in other unit-generator languages, if they are similarly designed.

It may be possible to describe the same algorithm just by the combination of unit generators. For instance, the algorithm to update the read position may be implemented by a phasor unit generator. The reading position from the sound buffer can be given as the sum of the output from a phasor unit-generator that ranges from zero to 1,023 and the offset value, which represents the starting position to read from the buffer for the current frame. This offset value must be updated at the beginning of each frame. The resulting code for the synthesis graph, however, would be more complicated and less comprehensive for users.

Another possible solution for this problem, which is a little simpler, is to control the number of the samples to be fed into the *FFT* unit analyzer for the second source by temporarily disconnecting a connection between the *FFT* unit analyzer and a *blackhole* unit generator. In ChucK, a *blackhole* unit generator, which simply discards any given input, is a unit-generator that can be the starting node of a depth-first-search for the audio computation, as well as the *dac* unit generator. A unit generator (or unit analyzer) does not produce its output, if a unit generator (or unit analyzer) is not traceable from any *dac* or *blackhole* unit generator in ChucK's sound synthesis framework.

Thus, one can control the amount of the samples to be fed into an FFT unit analyzer by disconnecting it from a *blackhole* unit generator and can temporarily suspend its audio computation. **Figure 12** describes such an example. The example first advances the time only for the hop size for *fft2* and then immediately disconnects the connection between *fft2* and *blackhole* to deactivate *fft2*. After this disconnection, no more samples are fed from *soruce2* to *fft2*. Then, the code advances the time until the next timing when cross synthesis must be performed. Right after waking up from sleep, the code connects *fft2* again to *blackhole* so that it can be fed the input samples. In this way, modifying the synthesis graph in this example controls the hop size for *fft2*. As clearly seen, the example does not utilize an additional pair of *SndBuf* and *FFT* and is much simpler than the Fig. 11 example.

```
//load two sound files
 “/sound/violin.wav”   => string filename1;
 “/sound/cherokee.aif” => string filename2;

SndBuf source1;
SndBuf source2a;
SndBuf source2b;
filename1 => source1.read;
filename2 => source2a.read;
filename2 => source2b.read;

//build a synthesis graph for cross synthesis.
source1  => FFT fft1  => blackhole;
source2a => FFT fft2a => blackhole;
soruce2b => FFT fft2b => blackhole;

IFFT ifft => dac;
800 => ifft.gain; //adjust the output volume.

1024 => fft1.size => fft2a.size => fft2b.size
=> ifft.size => int FFT_SIZE;

FFT_SIZE / 2 => int HOP_SIZE_OUT;
FFT_SIZE / 4 => int HOP_SIZE_SRC2;

Windowing.hann(FFT_SIZE) => fft1.window;
Windowing.hann(FFT_SIZE) => fft2a.window;
Windowing.hann(FFT_SIZE) => fft2b.window;
Windowing.hann(FFT_SIZE) => ifft.window;

complex Z[FFT_SIZE / 2];

0 => int posA;
0 => int posB;

//after 512 samples, we need to the read posi-
//tion for source2b so that it can be fed the
//samples from the correct position (the start
// the pos of the source2a + hop size).
HOP_SIZE_OUT::samp / 2 +=> now;
HOP_SIZE_SRC2 => posB => source2b.pos;
HOP_SIZE_OUT::samp / 2 +=> now

//now, source1 and source2a have been fed
//1024 samples from the heads of their buffers.
//the source2b has been fed 512 samples from
//the head of its buffer + the hop size.

//we are ready to start cross synthesis.
while(true){

  //first, process source1 & source2a.
  fft1.upchuck();  //perform FFT for source1.
  fft2a.upchuck(); //perform FFT for source2a.

  for (0 => int i; i < fft1.size() / 2; i++){
    fft1 .cval(i) $ polar => polar a;
    fft2a.cval(i) $ polar => polar b;
    %(a.mag * b.mag, a.phase) => polar c;
    c $ complex => Z[i];
  }
  ifft.transform(Z);

  //update the next read pos for the source2a
  //so that the next frame can be read from
  //(the head of the source2b + the hop size).
  HOP_SIZE_SRC2 * 2 +=> posA;
  posA => source2a.pos;

  //advance the time to process the next frame.
  HOP_SIZE_OUT::samp +=> now;

  //now, process source1 & source2b.
  fft1.upchuck();  //perform FFT for source1.
  fft2b.upchuck(); //perform FFT for source2b.
  for(0 => int i; i < fft1.size() / 2; i++){
    fft1 .cval(i) $ polar => polar a;
    fft2b.cval(i) $ polar => polar b;
    %(a.mag * b.mag, a.phase) => polar c;
    c $ complex => Z[i];
  }
```

Fig. 11 Another example of cross synthesis in ChucK.

```
  ifft.transform(Z);
  //update the next read pos for the source2b
  HOP_SIZE_SRC2 * 2 +=> posB;
  posB => source2b.pos;

  //advance the time to process the next frame.
  HOP_SIZE_OUT::samp => now;
}
```

Fig. 11 Another example of cross synthesis in ChucK (continued).

```
//load the sound files to the buffer objects.
 “/sound/violin.wav”   => string filename1;
 “/sound/cherokee.aif” => string filename2;
SndBuf src1;
SndBuf src2;
filename1 => src1.read;
filename2 => src2.read;

//build a synthesis graph.
src1 => FFT fft1 => blackhole;
src2 => FFT fft2 => blackhole;
IFFT ifft => dac;
800 => ifft.gain; //gain for ifft.

//set up FFT parameters.
1024 => fft1.size => fft2.size => ifft.size
=> int FFT_SIZE;
FFT_SIZE / 2 => int HOP_SIZE;
FFT_SIZE / 4 => int HOP_SIZE_SRC2;

Windowing.hann(FFT_SIZE) => fft1.window;
Windowing.hann(FFT_SIZE) => fft2.window;
Windowing.hann(FFT_SIZE) => ifft.window;

//to store the cross synthesis result.
complex Z[FFT_SIZE / 2];

//main loop.
while(true){
  //perform FFT for two inputs.
  fft1.upchuck();
  fft2.upchuck();

  //cross synthesis
  for (0 => int i; i < fft1.size() / 2; i++){
    fft1.cval(i) $ polar => polar a;
    fft2.cval(i) $ polar => polar b;
    %(a.mag * b.mag, a.phase) => polar c;
    c $ complex => Z[i];
  }

  //perform IFFT.
  ifft.transform(Z);

  //advance the time for the source2 hop size.
  HOP_SIZE_SRC2::samp +=> now;
  //disconnect fft2 from the blackhole ugen.
  //so that no more samples are fed from src2.
  fft2 =< blackhole;

  //sleep until the timing for the next frame.
  (HOP_SIZE::samp  - HOP_SIZE_SRC2::samp) +=> now;
  //connect fft2 again to the blackhole ugen.
  fft2 => blackhole;
}
```

Fig. 12 Yet another example of cross synthesis in ChucK.

However, aside from whether or not other unit-generator languages allow direct access to low-level data from algorithms that users describe themselves, as in Chuck’s unit analyzers, the solution in the Fig. 12 example may be more specific to ChucK than the Fig. 11 example since some unit-generator languages may not support the dynamic modification of a sound synthesis graph as required in Fig. 12. Moreover, even in the language that allows the dynamic modification of a unit-generator graph, the disconnection and reconnection may not be performed with sample-rate accuracy, failing to modify the synthesis graph at the expected timing. Furthermore, both examples can be even more complicated when the hop sizes for all the FFT/IFFT frames differ. In the Fig. 11 example, one would need to add more pairs of *SndBuf* unit generators and *FFT* unit analyzers. In Fig. 12, more dynamic modification of the synthesis graphs must be involved.

In contrast, the Fig. 8 example in LC can simply use the different hop sizes by changing the parameters *ovlp1* and *ovlp2* on line 07. Moreover, no latency is involved, as it is not necessary to advance the time to feed the samples to an FFT object, as in ChucK or other unit-generator languages. Thus, the independence of LC's microsound synthesis framework from the unit-generator concept plays a significant role in avoiding unnecessary complexity in these situations. Furthermore, since LC's unit generators and patches can compute the output samples even without the advance of time (lines 14–15 and lines19–25 in Fig. 4), even when one of the sound sources is generated by a unit generator or a patch, there would not be much difference in complexity. A *Samples* object obtained from the output of a unit generator or patch can be directly used for spectral processing.

In addition, LC's microsound synthesis framework may also be suggestive to the sound synthesis framework design for unit-generator languages. The complexity in using different hop sizes among FFT frames is largely due to the software design in which unit generators fully synchronize their behaviors with one global system time. Yet, this issue has not been discussed in the design of existing computer music languages and has been neglected, while the use of different hop sizes among FFT frames can be important when computer musicians creatively explore spectral synthesis and analysis techniques. As described in this article, even to utilize a simple time-stretching technique for the formant source in cross synthesis, hop sizes can differ among FFT frames and then lead to complexity in the resulting code. The code can be more complicated when applying various spectral synthesis and analysis techniques at once. As this issue can damage the expressiveness of a computer music programming language and hinder creative exploration by computer musicians, it should be reviewed as a usability difficulty in computer music programming.

5. Conclusion and Future Work

We described how STFT can be performed in LC within its microsound synthesis framework and compared it to ChucK's unit-analyzer concept, which shows a certain degree of similarity. Both languages provide accessibility to low-level data and the expressiveness to describe desired operations on the low-level data within just the language, without using dedicated built-in objects. Yet, due to the dependence on the unit-generator-based sound synthesis framework of ChucK's unit-analyzer concept, these two languages can show significant differences in the complexity of the resulting implementation, even for a simple spectral processing technique in certain situations. It was observed that ChucK can exhibit more complexity in the resulting code when the hop sizes differ among FFT frames.

As this complexity is not specific to ChucK but is caused by the design of the sound synthesis framework that fully synchronizes the audio computation of unit generators with only one global system time, the same problem can also be exhibited in other unit-generator languages. Thus, LC's sound synthesis framework design not only provides a design exemplar for further research in computer music language design but also is beneficial for reconsidering the design of existing unit-generator languages.

For the support of better usability and expressiveness in computer music languages, it would be desirable to consider the following issues for further discussion, which we described by comparing LC's microsound synthesis framework and ChucK's unit-analyzer concept: how STFT should be integrated in computer music programming language and whether unit generators should synchronize the audio computation completely with the advance of global system time.

For future work, since the implementation of LC is still just a proof-of-concept prototype and the number of library functions is limited at this point, we are planning to provide more STFT-related library functions and to investigate how other STFT-related techniques can be implemented in LC and other languages for further inquiry in computer music language design.

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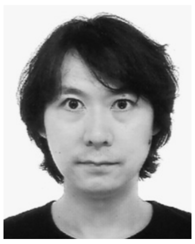

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