AUDIOVISUAL INTEGRATION WITH CALVR AND LIBCOLLIDER

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ABSTRACT

We present libCollider [3], a client library for SuperCollider's *scsynth* sound synthesis engine that provides C++ application developers with direct access to scsynth's sophisticated capabilities for real-time audio synthesis and rendering through an API with abstracted bi-directional network support. Its development is driven by the demand for an audio component to the CalVR visualization framework under development at the Immersive Visualization Laboratory at the Qualcomm Institute, San Diego. We describe common problems with audio integration into C++ realtime graphics applications, and the specific demands we are trying to meet with libCollider's multi-level API.

1. WHAT'S IN THE WAY OF AUDIOVISUAL INTEGRATION IN REAL-TIME RENDERING?

Sound is an important sensory modality of human experience, but even in 2013, it is often still addressed very insufficiently by applications for data and information display.

A coherent integration of audio components into visualization environments has much promise—some of which can be discovered in Michel Chion's book *Audio-Vision* [8]. However, attempts to tap into these potentials of intermodal perception have to overcome a variety of obstacles.

First of all, in the interdisciplinary discourse that surrounds the development of visualization systems, audio components tend to occupy the role of a peripheral add-on, as something that comes into perspective after the work on the visualization has already been successfully completed.

On the technical level, the resources left to meet the demands of qualitative sound projection are often scarce—the need for good loudspeakers and their placement, of low ambient noise around the display, considerations of room acoustics and screen reflections, et cetera.

Additionally, the expressive potentials of multi-channel sound systems as they have been explored in the area of computer music and more recently multichannel movie soundtracks often remain unconsidered in the planning and conceptualization of the display as do the manifold potentials to display data through non-speech sound [10].

But next to the lack of opportunity to experience high quality sound projections first person and of awareness for the potential roles of audio in audio-visual display, there are also important structural differences between image and sound that tend to be overlooked.

Even though auditory and visual media components have the potential to generate unified audio-visual impressions, the respective senses are characterized by a different responsiveness to temporal development, which results in divergent requirements for form and structure of their display and their digital rendering. Music and sound diverge in their temporal morphology from the dynamic behaviors of visual images and animation.

This results in divergent demands for temporal behavior and accuracy of perception-oriented process scheduling, and the temporal morphology of respective real-time processing in general. A separation of the real-time computation for both modalities into independent parallel processes is unavoidable, and as a result, audio-visual software projects tend to be assembled on separate development platforms, which in turn is reflected in a segregation of the respective development teams, severely complicating the creation of audio-visual potentials.

We can also see this problem reflected in the emergence of applicable standards.

1.1. Standards and their adoption

In the visual domain, open programming interfaces such as OpenGL and geometric scene graphs [4] have been adopted across a wide variety of platforms. Similar attempts in the audio domain, such as MPEG-4 and OpenAL for example, have not developed a similar universal appeal and have not transcended specific application niches. Even less is heard about unified audio-visual platform standards.

In the late 1990s it seemed as if sound and image were coming together in programming environments such as Max/MSP/Jitter and PD/Gem, but these environments have mostly been used by musicians to do visuals, while visualization researchers and visual artists continued to pursue sound-independent infrastructures like Processing, openFrameworks et cetera, that rely on the definition of a *custom network protocol* to connect to an independent real-time audio programming environment.

While the potential to share data between independent programming environments through a network protocol such as *OSC* [16] opened up many possibilities, the resulting structural complexity of indepement components is also a source of much friction and frustration.

Next to establishing the network protocol as such, the two communicating environments need to be kept manually in sync, and often specialist programmers—one for the visual and one for the audio domain—are required to operate in close communication.

This lack of a unified coding platform for visual and auditory rendering stands in the way of the successful design of detailed inter-modal experiences.

2. SOME EXISTING STRATEGIES

In **Table 1**, we list a small array of real-time audio rendering infrastructures that are used in application development today.

2.1. Integrated vs. standalone API

Auditory and visual components of interactive applications need to operate in temporal coherence, but in symmetry to their distinct physiological and cognitive processing, rendering and interactive adaptation are best addressed by distinct processing architectures. For example, vision-oriented processing usually proceeds in frames that are updated every 40 or so milliseconds, while audio processing attempts to produce a continuous stream of much finer temporal resolution, without the inherent need of frame-oriented temporal subdivisions. While most graphics processing is spatial—transforming 2D images or rendering perspectives on 3D geometry—audio processing usually operates on a set array of correlated time-series and their continuous temporal evolution.

Adding real-time audio to a visual application therefore implies the addition of a distinct temporal processing architecture, and the solutions can be roughly divided into two categories:

1. running a connected stand-alone programming environment for real-time audio in parallel to the core application,
2. including a suitable real-time audio library into the application's native code.

2.2. Standalone-API solutions

In a frequently encountered scenario in category 1), a library of audio functionalities is implemented in the audio-programming language as a *Sound-Server* that provides services such as audio playback and synthesis.
Often we find sound servers implemented in standalone domain-specific programming environments that provide libraries of primitives and abstractions for audio synthesis and data processing that can be combined into audio processing graphs. Examples are Pure Data, Max/MSP, Audio Mulch, CSound, and SuperCollider.

Ideally a server implemented in one of these languages runs in a parallel processing thread on the host machine or on an independent remote machine controlled by a bi-directional protocol of network packets, either direct UDP or Open Sound Control [16]. This separation of audio client and server also permits the use of specialized machine hardware and software that helps to manage the overall CPU load.

In order to be used with a separate application framework like our target application framework CalVR [13], communication protocols are defined that control the audio processing graph in the remote sound process. Previous efforts in this direction include Gerhard Eckel's Sound-Server [9], and we have undertaken various efforts at our own institute to provide visualization and VR applications with Max/MSP and PD-based sound-server components that communicate to the visual application using specifically created OSC-based network protocols [14, 15].

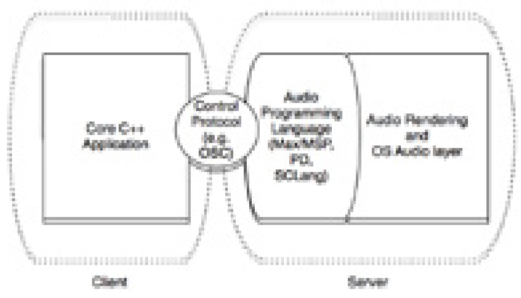

Core C++ Application
Control Protocol (e.g. OSC)
Audio Programming Language (Max/MSP, PD, SCLang)
Audio Rendering and OS Audio layer
Client
Server

Figure 1. Connecting the application core to a standalone audio-API with a custom communication protocol.

The use of standalone audio synthesis solutions requires the maintenance of the coherence between the client and the server component, forcing the developer of the client application to deal with an environment they may not be specialized in (or prefer to stay away from). In certain situations the management of domain-specific code such as SCLang or abstractions in PD and Max/MSP might be unacceptable. A suitable library and API in the native language of the client code may be preferred, which leads us to category 2) - integrated audio solutions.

2.3. Integrated solution

As an alternative to the creation of a detached *Sound-Server* process, it is also possible to include audio libraries into the core application directly, consolidating the sound design with the application development while dispensing with the requirement for system-level coordination of independent client and server components.

Most of the integrated audio libraries available today—OpenAL, FMOD Ex, irrKlang to name a few—provide functionality for a specific application domain, for example video game applications. Shortcomings are often a lack of flexibility in i/o channel configuration, routing,

Strategy	Language	Application Target
OpenAL	C++, C	gaming
FMOD	C++	gaming
irrKlang	C++, C#	gaming
Max/MSP, PD, SuperCollider	custom languages	custom applications
libpd	C, C++, Obj-C, Java	custom applications

Table 1. Overview over existing real-time audio software module libraries.

multichannel rendering, but most of all strong limitations on dynamic real-time audio synthesis.

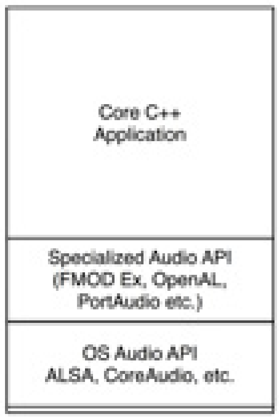

Core C++ Application
Specialized Audio API (FMOD Ex, OpenAL, PortAudio etc.)
OS Audio API ALSA, CoreAudio, etc.

Figure 2. Integrating system audio into the core application directly through high-level audio API.

Integrated API solutions are of great advantage when it comes to the deployment of applications as they provide a seamless interface between the core application and the audio functionality. On the other hand, the currently available libraries provide only restricted and pre-formalized capabilities for real-time rendering that are not sufficient for the level of interactivity our prototype applications demand and tend to rely on proprietary components which limits their development potential.

3. AUDIO API: DESIRABLE FEATURES

In the following paragraphs we summarize our requirements for an optimized integrated solution to flexible audio rendering in C++ applications. These emerge both from our own experience as well as interviews and meetings with our collaborators in the visualization group.

3.1. Simple access to real-time audio playback and synthesis

The API should provide the ability to control real-time audio, playback and synthesis for a variety of C++ applications in the domain of visualization, music and interdisciplinary media. Ideally, the API comes in the form of a client library (written in C++) with the necessary classes to instantiate, control, create, and stop audio processes as needed in the application areas listed above.

3.2. Portable, light, efficient and easy to install

The demands for audio support arise on different platforms, computing infrastructures and application models. This places specific demands on cross-platform portability, computational efficiency and independence from platform specific libraries: Server and client should be able to run on at least Mac OS, Linux, Windows, potentially Android and iOS, and should provide support for virtual reality displays, immersive environments, desktop apps, mobile apps, large scale display walls, unexpected art installations as well as computer-based audiovisual instruments, performances, et cetera.

3.3. Intuitive, multi-level API

We want our programming library to accommodate a range of different approaches to audio programming. On the one hand, standardized functionality such as the controls for localized sound playback needs to be made accessible in a convenient way that does not require a full knowledge of the API details. On the other hand, we want to equip our API with the potential to control and modify the audio processing and synthesis in way comparable to the standalone APIs we have previously used. In order to enable powerful interactive sound design, the server should support responsive and adaptive real-time synthesis with controllable filters, modulators and other real-time processing units such as those found in SuperCollider, Max/MSP, Pure Data, etc.

Naturally this leads us to the requirement of a multi-level API that provides full low-level access as well as encapsulated functionality that can be used without specialized knowledge. The API should be completely sufficient to control audio rendering and make additional specialized programming environments unnecessary.

3.4. Flexible and interchangeable support for different multichannel techniques

Industry standard C++ libraries for sound design rarely offer flexible and extensible multichannel support used in experimental speaker layouts. Since we are exploring different spatial audio rendering techniques such as Wavefield Synthesis, VBAP, Ambisonics, Beamforming, et cetera, we need our Audio API to integrate the respective rendering easily. We need support for multichannel spatial audio rendering algorithms, flexible multichannel routing, and easily managed multichannel objects.

3.5. Open source audio backend

The server should be, or be built on, a well maintained open source software for accomplishing the tasks listed above to ensure that projects created now will see continued improvement over time as the server improves from community development, since our current personnel will not be able to maintain a proprietary custom solution on its own.

4. WHY SUPERCOLLIDER?

SuperCollider is a computer music programming language developed by James McCartney. In its third implementation, also known as *SC3*, McCartney split the system architecture into two independent components—a language process that permits a temporal structuring and organization of the processing, and an independently running synthesis engine. The two components are coupled via an internal network message protocol implemented in Open Sound Control [16]. This makes *scsynth*, the component

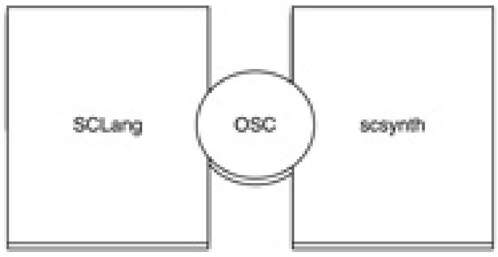


Figure 3. SuperCollider's client+server architecture.

performing the sound rendering, interesting for our purposes in two aspects. On the one hand, it represents a highly flexible sound-rendering process that can be entirely controlled through network messages allowing it to become deployed on any system accessible via network. On the other hand, the network messages that control the sound synthesis do not have to be generated by SCLang, but could also come from another language, as McCartney himself suggests in the conclusion of his paper presentation from 2002 [11].

Our strategy is therefore a supposition of SCLang with a C++ client library that can be directly included into the development of a C++ application—in our case, the CalVR visualization and virtual reality framework [13]. The resulting API structure can be seen in overview in figure (4) and will be described in the following paragraphs. A similar use of *scsynth* has been proposed by Hans Holger Rutz's jCollider [12] albeit for the Java language, and our approach also shares some similarity to the strategy used by *libpd* to employ Pure Data as a sound engine [1]. A comparison between libCollider and libpd will allow us to shed additional light on our approach.

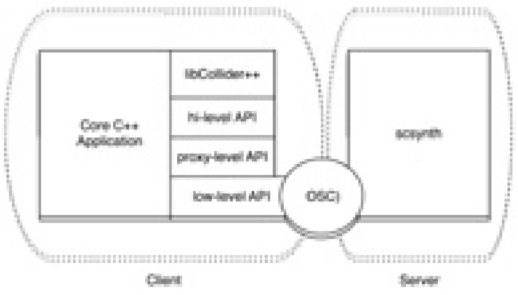


Figure 4. libCollider permits direct coupling between C++ application and scsynth, without the need for an additional language process like SCLang.

5. HOW IS LIBCOLLIDER DIFFERENT FROM LIBPD?

libpd is a software library that allows one to run an instance of Pure Data [5] as an audio processing callback and to allow MIDI and control message communication between application code and Pure Data code [7, 1]. Programmers can create a *PD patch*, load it into an instance of Pure Data running in a parallel thread to the application proper, and address it as a custom audio rendering engine by communicating to it through a set of language bindings provided by libpd. While this approach has some aspects in common with the way libCollider uses *scsynth*, there are important differences. On the one hand libCollider is currently only accessible from C++, while libpd implements bindings for an array of additional languages such as Java, Objective-C, Python, et cetera. On the other hand, libCollider attempts to make the functionality provided by SuperCollider's scsynth completely accessible from an external API, with no need for an additional graphical or text-based patching language such as the graphical patching of Pure Data or SCLang. Instead of delegating the low-level audio programming to an external language, libCollider organizes its functionality into different layers, from low-level functionality that wraps the commands available to control *scsynth*, to mid-level classes that allow a use of C++ in a similar way as SCLang, to a high-level API layer that provides the programmer with a simplified interface to often-used and basic audio functionality comparable to what a library like OpenAL provides. The multi-level layout of the API is further explained in section .

The architecture libCollider proposes has the advantage that the audio rendering process can reside on a different machine connected by network messages, thereby providing the advantages of a standalone *Sound-Server*

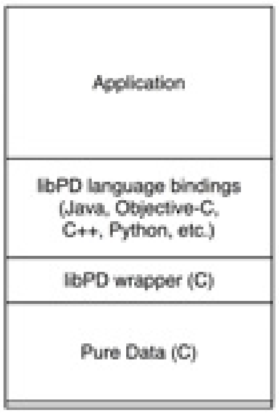


Figure 5. libpd's integrated solution for app audio: A C-based wrapper around Pure Data with multi-language bindings.

solution, while at the same time providing a close coupling between the sound rendering process and the application programming interface.

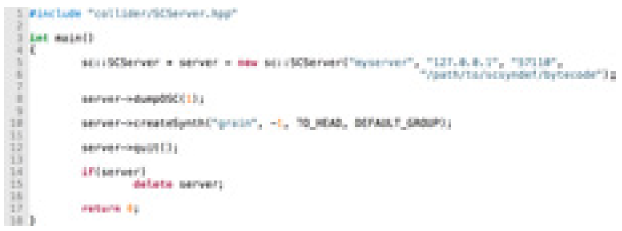
libpd achieves a separation of the application thread from the sound rendering thread, but does in itself neither permit an on-the-fly reorganization of audio rendering, nor—without the addition of a custom network layer—to displace the sound-rendering process onto another machine.

In the future, libCollider might be able to provide variable language bindings comparable to those of libpd, however, similar client libraries in other languages already exist, for example jCollider [12].

6. LIBCOLLIDER MULTI-LEVEL API

6.1. Low-level API - The Server Proxy

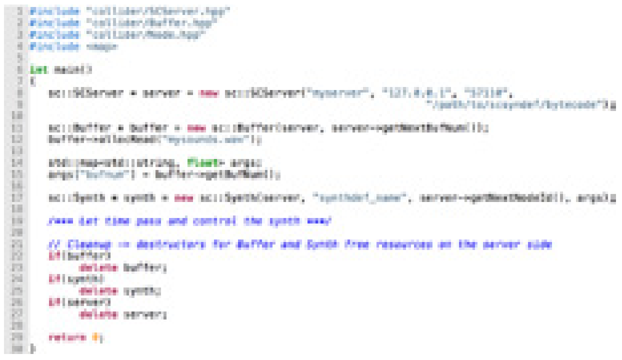
Since scsynth has a clearly documented set of the commands it understands and replies it returns [6], the initial motivation behind the API was to create one or more classes that handle sending those commands and properly dispatching any replies from the server. The initial result is the low level API contained in the class SCServer. SCServer has a full featured set of public members that currently include almost all commands found in the scsynth command reference [6]. SCServer also serves as a client-side proxy of a remote server instance. Thus, in a project you would typically have as many SCServer instances as you have server instances. SCServer conveniently abstracts all of the network and OSC code that is used to send and dispatch messages to and from the server via UDP or TCP. A code example of the low-level API looks like so:



```
#include "collider/SCServer.hpp"

int main()
{
        sc::SCServer * server = new sc::SCServer("myserver", "127.0.0.1", "57110",
                                                 "/path/to/scsyndef/bytecode");

        server->dumpOSC(1);

        server->createSynth("grain", -1, TO_HEAD, DEFAULT_GROUP);

        server->quit();

        if(server)
                delete server;

        return 0;
}
```

6.2. Mid-level API - Quasi-SCLang

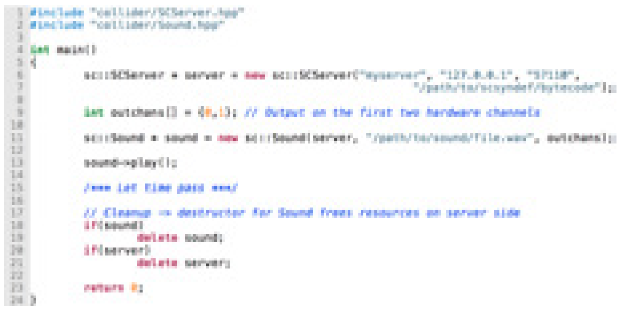
The motivation behind the middle portion of the API is to provide a set of proxy classes that mirror the structures involved in audio synthesis one works with when using scsynth. These include the familiar objects Buffer, Bus, Node and its subclasses, Synth and Group. These proxy classes provide a means to instantiate and retain a handle on server-side instances of these objects. Most of these classes are passed a pointer to a valid SCServer instance. The SCServer instance conveniently provides the necessary commands to instantiate and control the objects that wrap a pointer to it. An obvious benefit of this model is that each proxy class instance can leverage the same single SCServer instance for their instantiation and control, and perhaps most importantly, it can enable synchronization between the client application and server. For example, by instantiating a Buffer and calling one of its methods to load an arbitrary soundfile, the Buffer can use the SCServer for the low-level work of sending commands to the server, and in turn, populate the Buffer's members such as channel count and sampling rate based off the reply from the server. This synchronization is important to the client in cases when channel count or sampling rate of the file determine what module is loaded or what event happens next in the client application, or perhaps when another class uses that information during initialization as we will see in the high-level portion of the API. An example of the mid-level API looks like so:



```
#include "collider/SCServer.hpp"
#include "collider/Buffer.hpp"
#include "collider/Node.hpp"
#include <map>

int main()
{
    sc::SCServer * server = new sc::SCServer("myserver", "127.0.0.1", "57110",
                                             "/path/to/scsyndef/bytecode");

    sc::Buffer * buffer = new sc::Buffer(server, server->getNextBufNum());
    buffer->allocRead("mysounds.wav");

    std::map<std::string, float> args;
    args["bufnum"] = buffer->getBufNum();

    sc::Synth * synth = new sc::Synth(server, "synthdef_name", server->getNextNodeId(), args);

    /*** let time pass and control the synth away/

    // Cleanup -- destructors for Buffer and Synth free resources on the server side
    if(buffer)
        delete buffer;
    if(synth)
        delete synth;
    if(server)
        delete server;

    return 0;
}
```

6.3. High-level API - Sound made easy

The most recent addition to the library, the Sound class, represents the start of the high-level portion of the API. This class is intended to be as intuitive and easy to use as possible, targeted towards developers with casual exposure to audio practices. The Sound class and future high-level classes are intended to make it easy to build and control one or many instances of simple to complex audio generators. These high-level classes can build directly on the classes from the low and mid-level portions of the API. For example, the Sound class represents a standard soundfile player, with looping, tape-head transposition, n-channel playback, seeking, gain control, fade in/out envelopes, and of course typical stop/start playback. In the simplest case, it uses internal Synth and Buffer members to load the necessary soundfile and control it via a synth definition on the server. The Sound class can be used for simple soundfile playback like so:



```
#include "collider/SCServer.hpp"
#include "collider/Sound.hpp"

int main()
{
        sc::SCServer * server = new sc::SCServer("myserver", "127.0.0.1", "57110",
                                                  "/path/to/scsyndef/bytecode");

        int outchans[] = {0,1}; // Output on the first two hardware channels

        sc::Sound * sound = new sc::Sound(server, "/path/to/sound/file.wav", outchans);

        sound->play();

        /*** let time pass ***/

        // Cleanup -> destructor for Sound frees resources on server side
        if(sound)
                delete sound;
        if(server)
                delete server;

        return 0;
}
```

7. CALVR PROJECTS USING LIBCOLLIDER

We are working on a series of example applications, some of which demonstrate a simple generalized use of spatialized sound projection. Our specific interests in the use of libCollider can best be understood from the perspective of two projects.

7.1. Seismic Viewer: Increasing spatio-temporal resolution through audio-visual syncretism

While the visual sense is arguably superior when it comes to a precise spatial analysis of our surroundings, the auditory sense appears to be of a significantly higher temporal resolution. An integration of both is especially valuable as each sense can contribute its own potentials and strengths in the fused emergence of *audio-vision* [8]. While CalVR proceeds in its rendering frame-by-frame, producing about 20-40 fps—or one frame every 25-50 ms, the temporal resolution of the auditory sense is evidently much higher.

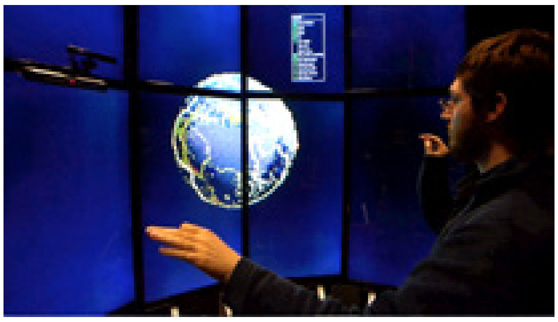


Figure 6. Interactive audio-visual display of seismological event catalogs [2]

Our VR application *Seismic Viewer* employs this phenomenon by a temporal synchronisation of localized flashes of brightness with a stream of localized percussive sounds that represent a realization of global earthquake events presented in temporal scaling. Auditory events are scheduled at every CalVR frame, but a much higher resolution is achieved by using scsynth's potential for a micro-delayed execution of the transmitted commands.

7.2. Virtual acoustic simulator

Another application of libCollider in the context of CalVR is the creation of a real-time architecture design tool that includes the rendering of an auditory impression of the resulting spatial structure. While the computational efficiency of scsynth is already beneficial, the inherently dynamic and scalable nature of scsynth's rendering engine allows us to produce increased navigability and the support for more and more complex simulated geometries. A more detailed look at the virtual acoustic simulator will be the topic of a future publication.

8. CONCLUSION

While our own applications for libCollider are dominantly in the field of immersive visualization and audio-visual display, we hope that the application architecture of libCollider, both the separation of auditory and visual rendering into different threads controlled from a single C++ application as well as the use of SuperCollider's powerful *scsynth* audio rendering system will prove to be a successful model to integrate audio into a wide variety of cross-platform applications—and that the multi-level API will be attractive to programmers of different interests and specialization levels. We are looking forward for further development and lively contact with other potential users and co-developers around the globe [3].

9. ACKNOWLEDGEMENTS

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