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Oasis Rose the Composition – Real-time DSP with AudioMulch

Ross Bencina
Email: rossb@audiomulch.com
Web: http://www.audiomulch.com.au/

Abstract.

Oasis Rose is a composition incorporating live instrumentalists and real-time signal processing. The work makes use of algorithmically controlled non-linear signal processing techniques while attempting to maintain the "signal processing as extension of instrument" aesthetic. Real-time signal processing is implemented using the authors AudioMulch interactive music environment for Windows 95/NT™. This paper discusses the algorithms utilised in *Oasis Rose***,** and documents the authors experiences in bringing traditionally "studio" or "non real-time" signal processing techniques to the stage.

1.0 Introduction

Since my earliest experiences with audio effects and digital signal processing, I have been fascinated by their ability to transform and augment the sound of acoustic and electronic instruments. For electric guitarists and classic knob-box synthesists, the application of effects has been an integral part of musical performance for over three decades. While the use of "live electronics" in performance has an established history, the use of computer based dsp systems is a relatively new area of performance practice, especially outside the confines of academic institutions.

This paper documents my own experience in making real-time dsp assisted music with affordable personal computers, AudioMulch – the software system I have developed for this task, and some of the algorithms I have applied in *Oasis Rose* a recent work for Cor Anglais, Marimba, Viola and AudioMulch.

2.0 [Concrete, Meta, Hyper] – [Instrument, Performance, Improvisation, Composition]

The twentieth century has heralded an awakening to the plasticity of musical sound, and a rediscovery of both western and non-western improvisatory traditions. The application of computers to the exploration of an infinite void of musical structuring processes has introduced new possibilities for "dynamic", "interactive" and "reactive" performance. Further more the relationship between performers, instruments, meta- and hyper-instruments, compositions, and meta compositions becomes difficult to ascertain using extablished criteria.

For the most part, I'll leave the above areas to the theorists – but in creating music I have approached this context from a specific angle. As I have explored the realm more I have chosen to focus of specific interrelationships between the performance, the participants, and the musical/sonic design. In *Oasis Rose* I have sought to refine and strengthen the relationships between the "composition" or musical design and the dsp algorithms – by designing and tuning algorithms specifically for the work. While at the same time seeking to achieve a grounded performance where the performers could dwell within a sonic space, and where that sonic space was conducive to their musical performance. I was particularly interested in extending the improvisational possibilities of traditional instruments passing into and through processed sound rather than set "against" or "in dialogue with" it, as is often the case in interactive computer music performance.

2.1. Instrument Extension

I believe that applying real-time audio signal processing to an improvisers sonic expression grounds the performance in that expression to a greater degree than when processing is applied only to event or control signal data, as in many MIDI based interactive music systems. The presentation of acousmatic sound often evokes a sense of the other, in my experience this can easily unsettle performers - expected to improvise within an alien sonic context, and listeners – required to listen to highly skilled human performers conversing with arbitrary musical structures. Cues such as timbral and gestural coherence and re-iteration go a long way to fusing the worlds of performer and acousmate, as the large repertoire of tape/digital delay based compositions and performances evidences.

3.0 Previous Works

In some recent live performances I have utilised the following grouping of musical elements: instrumental or vocal performer(s), real-time signal processing, and sometimes additional pre-prepared material mixed live from multiple soundfiles and/or CD players. In all cases, there has been a large improvisational component, where the final musical outcome is unknown. The signal processing techniques employed have always involved non-linear sampling or granulation effects that introduce both pitch and time distortion to the input signal.

3.1 overSYTE

In all works prior to *Oasis Rose*, signal processing was performed using overSYTE [1] – a granulator running on Power Macintosh computers. OverSYTE provided a number of sliders and range sliders[1] for varying parameters, presets for storing entire parameter sets, and a preset interpolator for varying multiple parameters simultaneously in performance. All parameters could be modified in real-time.

One interesting feature of overSYTE was the preset interpolator, which allowed the user to place four previously stored presets at the corners of a square, and to interpolate between all of them at once. When a number of disparate presets were used, it was possible to perform wild time varying modulations within the parameter space. This lead to some great sonic effects, but quite often overwhelmed and disoriented the performers, as control of the sonic outcome had shifted entirely to the computer operator. In response to this experience *Oasis Rose* uses static parameter settings or very slow moving parameter modulations to allow the performers to remain grounded within the signal processing "world." It is my present opinion that fast or extreme parametric modulations are best applied under the performers control – either directly using physical controllers such as foot pedals, or tacitly using control data derived from the performers audio stream (such as envelope following and pitch tracking.)

3.2

Following the completion of my studies I no longer had access to Macintosh hardware, and development of overSYTE was suspended. At this time I already had plans for a system allowing the composition of multiple complex signal processing effects such as the overSYTE granulator and those of the kind found in GRM-Tools[2] with an interactive audio patching scheme along the lines of the Max Patcher[3].

In mid-1997 Borland released C++Builder for Windows 95/NT, this rapid application development environment seemed to offer all of the user interface development tools that I would need to develop the ideas initially encountered with overSYTE...

4.0 AudioMulch

AudioMulch is a real-time sound synthesis and signal-processing environment for Pentium and Pentium II computers running Windows 95/98/NT. It allows the user to interactively route audio signals through a variety of processing modules using a patcher interface [3]. The results of changes to the signal flow of the patch or to module parameters are heard in real-time, no compilation or "patch locking" is required.

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[1] A range slider operates in a similar fashion to a traditional slider except that it has two thumbs, allowing the specification of a range rather than a single value. The mouse is used in combination with various keyboard modifiers to move the maxima or minima, or the maxima and minima in parallel or contrary directions. In overSYTE, ranges are used for stochastic generation of a number of parameters including grain interonset time, grain transposition factor and grain duration.

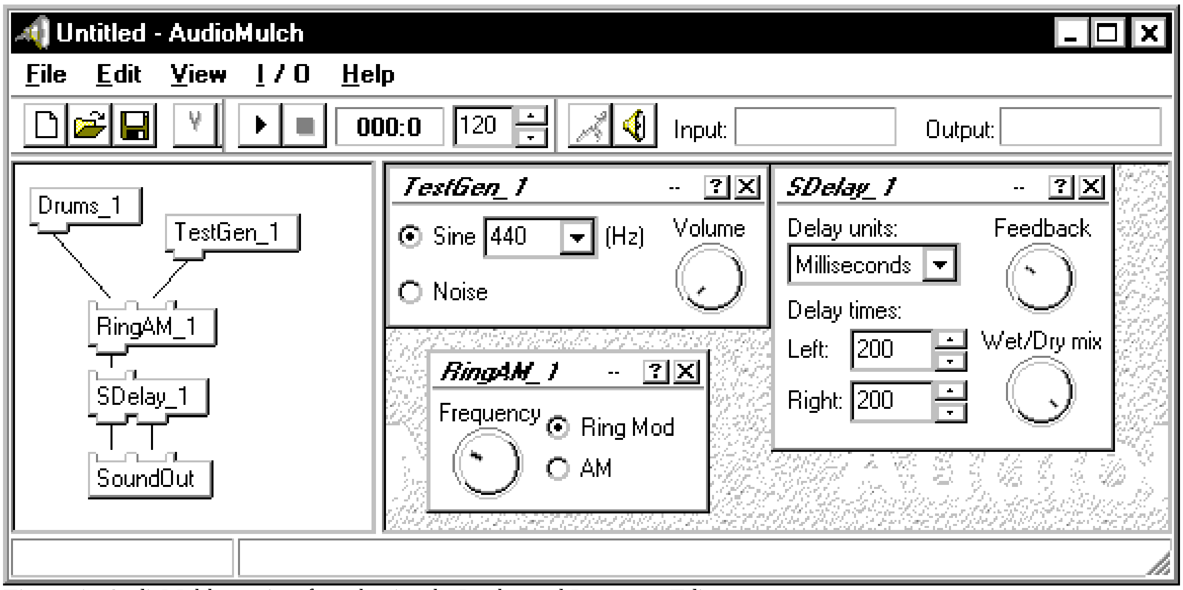

Figure 1. *AudioMulch user interface, showing the Patcher and Parameter Editors.*

Unlike many other computer music environments, AudioMulch provides a visual interface for every module. These "parameter editors" present knobs, sliders, range sliders and custom graphical interfaces for each module and may be hidden or tiled in the "properties pane" area of the user interface.

AudioMulch contains a number of modules for generating and processing sound, included are "popular" modules such as a techno bassline synthesiser and a drum machine, effects such as ring modulation, flanging, reverb and delays, and a number of more morphologically sophisticated units such as a delay-line granulator and stereo spatialiser. Input sound can be taken from sound files or real-time audio input, the output is heard in real-time and can be simultaneously recorded to a sound file.

The version of AudioMulch used for *Oasis Rose* was modified to provide four audio inputs and four audio outputs to allow three instrumental inputs, and a quadraphonic output. Custom modules were added for quadraphonic spatialisation, and automatic loop sampling. The custom processing units are described later in this paper. A future release of AudioMulch will include the multi-channel facility, and may include some of the other custom units.

5.0 Oasis Rose

Oasis Rose was devised as a live improvisation to be performed as part of the contemporary sound and music series at the 1998 Melbourne Next Wave Festival - held in conjunction with the National Academy of Music contemporary music workshop. The performers were attendees of the academy at that time. The work involved three live instrumentalists: Oboe / Cor Anglais, Viola, and Marimba / Percussion. Signal processing was performed using AudioMulch, which delivered a quadraphonic field to the sound diffusion desk. The live mix included unprocessed instrumental sound.

5.1 Performance Instructions

The following description is taken from the instructions to the performers:

> *Oasis Rose* is intended as a slowly evolving kaleidoscope or mandala of sound. The performers are asked to interpret / improvise phrases of their own creation based on four prescribed sets of four notes. The sound is spatialised using a quadraphonic diffusion field, surrounding both the players and the audience. Each performer is processed using a different set of signal processing techniques, which goes some way to establishing the role of each instrument within the musical texture. The computer and loudspeakers are used to musically extend the contribution of each player to the whole. The computer is intended to work cooperatively with the players in establishing the final outcome, which could also be viewed as a "communion" between the audience, players and signal processing algorithms.

Beyond this description, the pitch material, and some ideas for starting points, the performers were given free reign to explore and develop their own ideas within the signal-processing patch provided. The processing was largely predetermined, with some minor adjustments to spatial and temporal parameters during performance.

Three rehearsals were held to acquaint the performers with the processing being applied to their instruments, and to develop the shape of the work.

5.2 Technical Setup

All signal processing was performed using AudioMulch running on a Pentium II 266 machine with 64mb of RAM running Windows 95. A Frontier Design WaveCenter card provided ADAT light pipes to and from a ZULU converter box which provided the three instrumental inputs and four outputs. Although it was not used in the performance, a stereo delay effect was configured from the main sound rig as backup incase AudioMulch or the entire computer needed to be reset during the performance.

Next Wave had installed a thirty eight speaker diffusion system in the concert hall, a combination of these speakers were used to diffuse the quadraphonic sound field around the audience. The instrumentalists were lightly reinforced by front speakers to balance the mix between live and processed sound. Minimal adjustments were made to this mix during the performance.

6.0 Signal Processing Techniques In *Oasis Rose*

In *Oasis Rose* an independent signal processing network processes each instrument. The marimba is processed by a delay-line granulator ("DLGranulator") sequentially fed into a stereo delay line. The Viola is processed by a ring modulator tuned to 434Hz, and then into a delay-line granulator. The Cor Anglais is processed through a real-time loop sampling unit called "QuadraDrone" which generates four independent output streams which are then symmetrically spatialised to a quadraphonic field using a unit called "MandalaSpat". The individual algorithms and settings are described in more detail below.

6.1 DLGranulator – A Delay-line Granulator

DLGranulator is a delay line granulator using trapaziodal grain envelopes, implemented along similar lines to the SAM variant of Barry Truax's DMX-1000 based system[4]. DLGranulator differs from the SAM model in the following key areas:

- A longer delay line (10 seconds)
- Envelope skew for creating grains with hard attacks and slow decays or slow attacks and hard decays
- A feedback parameter to allow the granulated output to be re-injected into the input
- Quantize rate and amount parameters allowing grains to be synchronised to the AudioMulch tempo clock
- The use of a single onset scheduler sequencing all grains as a single polyphonic stream

All of the parameters of DLGranulators are listed below. Value ranges are shown in brackets, parameters marked * make use of range sliders to specify a range of values; in such cases each grain is assigned a random value from within the specified range.

Input Gain (0.0 – 1.0)
Grain Amplitudes (0.0 – 1.0) *
Grain Pans (left – right)*
Grain Sampling Delay (0ms – 10sec)*
Delay line freeze (bool)
Feedback (0.0 – 1.0)
Wet / Dry Mix (0.0 – 1.0)
Grain Transposition Factor (+/-2 octaves)*
Grain Interonset Time (5ms – 2000ms)*
Maimumx Grains (0 – 20)
Grain Quantization Grid (32nd triplet - quarter note)
Grain Quantization Amount (0 – 1)
Grain Durations (10ms – 500ms)*
Envelope Shape (0.0 [triangle] – 1.0 [rectangle])*
Envelope Skew (-0.95 - +0.95)

6.2 Application of DLGranulator to the Marimba

The Marimba was granulated into a particled texture of quite long grains (190ms – 500ms), the grain envelopes were negatively skewed to provide fast attacks giving the grains a percussive quality. Grain interonset times were selected from the range 26ms 333ms, creating a sparse texture with few overlaps. The delay range was initially set to select grains from the delay line between 0 and 3 seconds. The grain transposition factor was set to raise by one octave, and a moderate amount of feedback was applied creating a shimmering octave cycling effect above the live sound.

The overall effect of the above settings was of a cloud of high-pitched marimba particles inhabiting the same harmonic field as the live marimba. As the marimba part ebbed and flowed during the course of the piece subtle adjustments were made to the settings. Adjustments were made to: The delay time range – shorter ranges meant less time smearing, which generally led to a sparser texture resulting from the smaller window of potential sound events. The grain interonset time range – both smaller and larger values were used to increase and decrease the grain density per unit time at various points during the piece.

6.3 Application of DLGranulator to the Viola

Parameter settings contrasting those of the Marimba granulator were employed to process the Viola: A short delay range, moderate feedback, a small transposition range with the bottom bound at unity, moderate grain durations (28ms - 38ms), symmetrical grain envelopes, and an interonset time creating single grain overlaps (19ms – 24ms).

The result of these settings is an upwards moving reverberant glissandi, and is a classic pitch-feedback granulation effect. With percussive attacks it creates extremely dynamic gestures, which successfully cut through the dense texture of the other sonic elements in *Oasis Rose*, at other times the viola played harmonic glissandi which were further enhanced by this granulation effect.

6.4 QuadraDrone – Automatically Sampled Drones

QuadraDrone was designed specifically for *Oasis Rose*, and was conceived to generate slowly changing standing chords based on the pitch material of the input signal. It samples stable pitched segments of the input signal and loops them into continuous notes. Up to four notes are generated simultaneously.

The QuadraDrone algorithm consists of an event extractor loading eight wavetables used by four crossfade looping oscillators (fig. 1). The event extractor examines the incoming audio stream and extracts one second segments of stable fundamental frequency and amplitude into one of eight wavetables. The event extractor is then constrained to wait for a “minimum intersample time” before it resumes looking for suitable segments. Segments are stored into the wavetables in an interleaved manner such that each oscillator in turn has its first table updated, then each oscillator in turn has its second table updated, then back to the first tables and so on.

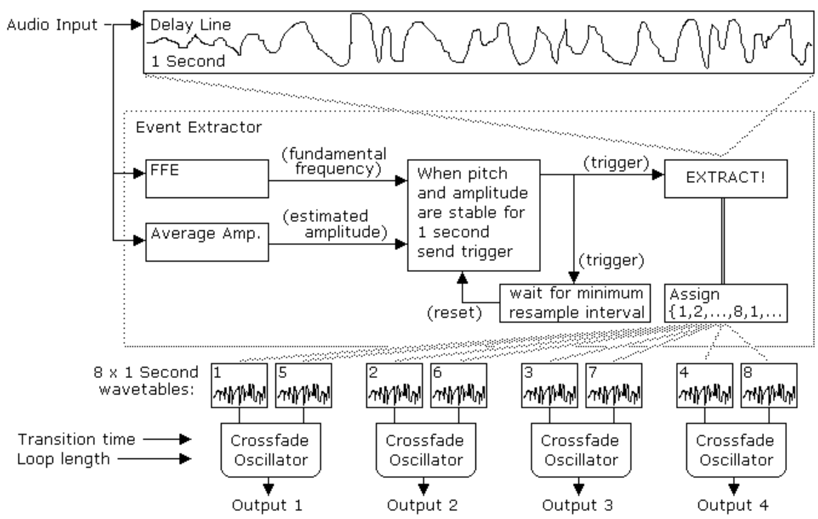

Figure 2. *QuadraDrone automatic sampling algorithm*

A crossfade oscillator generates a continuous signal by looping a sample and crossfading from end to start rather than performing a simple jump as is usually the case with wavetable oscillators. This implementation of brassage[5] or crossfade looping[6] allows the dynamic specification of the loop length. The crossfade oscillators used in QuadraDrone randomly select between a range of loop lengths to create non-periodic cycling to reduce the perceptibility of the loop points (Fig. 3a).

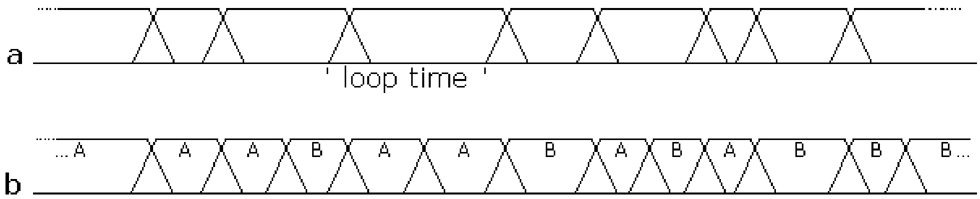

Figure 3. *Crossfade oscillator segment envelopes (a) variable loop time, (b) a probabilistic exchange between table A and table B.*

Each crossfade oscillator is notified when one of its tables is updated, it then performs a probabilistic exchange between the wavetable it is currently generating loops from, and the table that has most recently been updated. The transition time between tables is randomly selected from a range specified by the user, and can range from 0 to 30 seconds (Fig. 3b).

6.5 MandalaSpat

Mandala spat takes four input signals and separately spatialises each of them according to a symmetrical pattern involving four rotating ellipses (Fig 4.). Each of the following parameters is modulated by an independent sinusoidal LFO, the user can specify the rate of each LFO and the range of values across which the LFO sweeps:

Width and Height of ellipses
Rotational velocity of ellipses
Distance of circles from the center point
Velocity of ellipses around the center point.

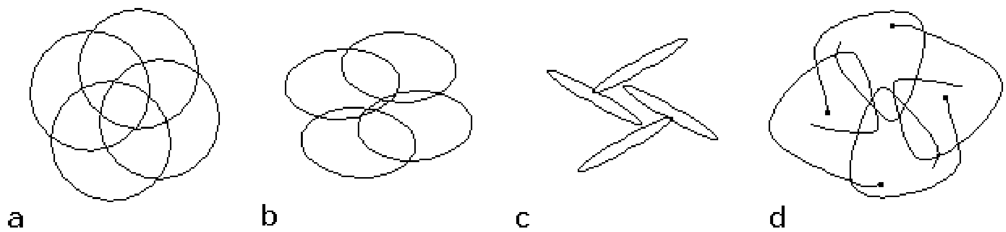

Figure 4. *MandalaSpat trajectories (a, b and c) elliptical paths, (d) source trajectories over 40 seconds.*

The position of each sound source is modelled as a point on its respective ellipse, the points rotate around their ellipses at a user controlled velocity. The spatialisation is implemented using a simple panning algorithm to pan each source to a point along a side of the quadraphonic square according to its angle from the center point, the distance from the center point is used to scale the amplitude. An amplitude factor is provided to scale the effect of distance on the output level.

6.7 Application of QuadraDrone and MandalaSpat to the Cor Anglais / Oboe

The QuadraDrone was fed directly by the Oboe input, the output of the QuadraDrone was fed into the four inputs of a MandalaSpat which was in turn patched to the four outputs. This patch created the effect of a chord of four notes slowly moving around the space, with intermittent transitions from one note to another. The chords slowly changed according to the pitch set played by the oboe. It was possible for the performer to vary the rate at which the chords unfolded by controlling the number of notes played for longer than one second.

The QuadraDrone distance / amplitude factor was used to control the proximity of the chords in performance. The QuadraDrone wavetables were manually silenced over the final 30 seconds of the piece.

7.0 Outcome

The two performances of *Oasis Rose* both went smoothly, in spite of the computer crashing during rehearsals, and a recompile of the entire AudioMulch project which finished as the performers were walking on stage. Audience feedback was positive, and my feeling was that I had achieved a greater degree of human-performer / computer-speaker integration than in any of my previous works.

Leakage between microphones caused some problems, and I suspect that this degraded the performance of the QuadraDrone pitch detector / event extractor algorithm. However, the cross-pollination of sonic materials created a nice shimmering effect that sounded great as the piece died away. If more extreme contrasts in processing had been employed microphone leakage may have interfered with the intent of the piece.

In hindsight, a longer rehearsal period and a more structured temporal plan may have benefited the performers, who were not particularly experienced in the kind of improvisation the work called for.

8.0 Future Directions

I intend to continue to develop works for vocalists and / or instrumentalists and live signal processing. From my experience in preparing *Oasis Rose* I believe the following additions to the performance system would prove useful:

- Controllers such as foot pedals to allow the performers to control parameters during the performance.
- The ability to predefine the motion of parameters over time, to lend some shape to the signal processing context.

This kind of music would benefit from performance by an ensemble established to perform improvised electroacoustic music.

AudioMulch is under development, details of planed features are available from the web site cited at the top of this paper.

9.0 Acknowledgements

Thanks to Lawrence Harvey for organizing the Next Wave concert series, to Michael Hewes for the superb live mix, to Next Wave for the amazing 38 channel sound system, and to Stephen Graham for getting my computer back in working order at the last minute.

10.0 References

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