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Music for Piano and Computer: A Description

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Abstract

Music for Piano and Computer was composed in 1996 using signal processing Max running on the IRCAM Signal Processing Workstation. The piece is divided into six sections, has a duration of approximately 18 minutes, and is part of a continuing series of interactive pieces by the author in which, via analysis of audio input, musicians regulate many aspects of algorithms for both control and digital signal processing in real-time performance situations. Analysis/resynthesis and convolution, both which require spectral domain analysis, are used extensively in the composition.

作品 "Music for Piano and Computer" の解説

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"Music for Piano and Computer"は、IRCAMシグナル・プロセッシング・ワークステーション上で稼働する信号処理プログラム「MAX」を用いて1996年に作曲された。演奏時間約18分のこの作品は6つのセクションに分かれており、オーディオ入力解析を介し、演奏者が実時間演奏において制御とデジタル信号処理のためのアルゴリズムの多様な局面を統制していく著者による一連のインタラクティブ作品の一つである。この作品ではスペクトラル領域における分析を必要とする解析/再合成とコンヴォリューションが広く用いられている。

Introduction

Music for Piano and Computer (*MPC*) was composed in 1996 using signal processing Max [Puckette, 1988, 1991] running on the IRCAM Signal Processing Workstation (ISPW) [Lindemann, *et al*, 1990]. The piece is divided into six sections, has a duration of approximately 18 minutes, and is part of a continuing series of interactive pieces by the author in which performers regulate many aspects of algorithms for both control and digital signal processing (DSP) in real-time performance situations [Lippe & Puckette, 1991]. Analysis/resynthesis and convolution, both which require spectral domain analysis, are used extensively in the composition.

1 The Digital Signal Processing Environment

Briefly, the DSP environment used in *MPC* consists of analysis, transformation, and sound synthesis modules. Some of these modules are kernel objects in Max, additionally there are external objects (written by Puckette and others) which are not part of the standard Max release, and some modules are algorithms or "patches" which have been developed in the Max programming environment by the author, Puckette, and other users of the ISPW.

1.1 Analysis

The analysis level of the DSP environment consists of an analog-to-digital convertor (ADC) for the piano audio input via a microphone, a pitch tracker, an amplitude follower (which includes attack, threshold, and rest detection), and a spectral analyzer. The spectral analysis is FFT-based, and is employed in three ways: (1) a continuous FFT/IFFT analysis is used for cross synthesis (also known as "convolution") [Settel & Lippe, 1994], (2) an FFT-based analysis of the piano's spectrum is used to control the amplitudes of a bank of oscillators, and (3) an FFT-based external object called *jack~*, developed in 1994 by Puckette, is used to control a bank of oscillators in an analysis/resynthesis application. The *jack~* object detects up to twenty peaks in a given signal for each FFT analysis window, and outputs frequency/amplitude pairs. The *jack~* object attempts to maintain continuity between analysis windows by keeping the same voice allocation for common frequencies in adjacent windows. (When controlling a bank of twenty oscillators with the twenty frequency/amplitude pairs, this attempt at maintaining continuity helps to keep the oscillators from jumping to extremely different frequencies when the spectrum of the input changes.)

1.2 Transformation

A DSP module which outputs an altered version of an incoming signal can be considered transformative, while a module that does not take an incoming signal, but only outputs a signal, can be considered generative. Thus, reverberation is transformative and synthesis via oscillators is generative. Furthermore, of the spectral analysis applications mentioned above, convolution is transformative, while analysis/resynthesis is generative. Other modules used for transformation in *MPC* include a bank of harmonizers, random amplitude modulation, flanging, ring modulation, filtering, a variant on the Karplus-Strong algorithm (using the piano as the noise impulse), a bank of variable delays, reverberation, and a phase-based spatialization algorithm.

1.3 Sampling

Sample playback falls, somewhat arbitrarily, in a grey area between transformative and generative. Depending on the application, sampling could be considered highly transformative (real-time recordings of input which are played back in non-standard ways), while normal playback of pre-recorded sounds might be considered generative. *MPC* employs both transformative and generative sampling techniques, making special use of the non-standard technique of granular sampling [Lippe, 1994].

1.4 Synthesis

As mentioned above, 20 oscillators are controlled using both *jack~* and FFT-based amplitude control. In the simplest case, sinusoidal waveforms are used, but *MPC* also makes use of other more complex waveforms (although this is a rather non-standard approach to analysis/resynthesis techniques). Frequency modulation (FM) is also employed in the piece, but only as a sound source for convolution.

1.5 Control versus DSP

The attempt to categorize techniques as either transformative or generative is difficult at times. This difficulty is somewhat similar to the blurring, in real-time contexts, of the traditionally separate computer music categories of "control" and DSP. This separation exists in order to economize CPU time, but in a real-time environment like Max, the intertwining of control and DSP can become quite complex. For example, the piano output from the ADC is DSP which is sent to the *jack~* object. The output of the *jack~* analysis is control information which is sent to the oscillator bank, and the output of the oscillators is, again, DSP. At the same time, compositional control over the choice of waveforms for the oscillator bank is influenced by an analysis of piano amplitude in the DSP domain which is, in turn, converted to control information, etc.

1.6 DSP Crossbar and Virtual DSP

A DSP cross-bar [Lippe, *et al*, 1991] exists to allow the output of all modules to become input for all modules that accept input. This kind of crossbar helps to maximizes DSP possibilities, although all of the DSP modules used in *MPC* cannot run simultaneously on a single ISPW. Modules are switched in and out of the DSP chain in Max using the *switch~* object, creating a kind of "virtual" DSP environment. Specific configurations of modules can be created, and DSP has both augmented potential and flexibility.

2 The Composition

As mentioned above, *MPC* is divided into an introduction and five sections, with an overall duration of approximately 18 minutes. The following description will deal predominantly with the computer part of the piece. Discussion of the instrumental part will be kept to general compositional concepts and procedures.

2.1 Electronic Introduction

MPC begins with an entirely electronic introduction of approximately two minutes duration. It is based on two very short pre-recorded piano passages which are time-stretched over the entire duration of the introduction using granular sampling techniques. Grain durations and envelopes are manipulated for timbral control, and the sampler output is transformed by the harmonizer bank with feedback, and by the spatializer.

2.2 Section I

The piano enters at the beginning of section I, which is approximately 4 minutes long. The primary compositional idea for this section is based on a rather simple metaphorical concept: the piano part consists of rising and falling arpeggios which cover a huge tessitura, much larger than the physical range of the piano. When notes are "sounding" outside the piano range, the score has rests. The initial arpeggios rise above the range of the piano (producing silence), turn and fall back into the range of the piano, and continue falling until they pass below the piano range (again producing silence), and eventually turn and rise back into the range of the piano again. This gestural motion is repeated continually, while the arpeggios' range is compressed gradually until they fall within the tessitura of the piano. At this point, the rests have almost totally disappeared. The compression continues until the arpeggios fall within the range of an octave somewhere in the middle of the keyboard. The concept of compression/expansion of tessitura is also exploited in the electronic part. Electronically, this section predominantly makes use of analysis/resynthesis using *jack~*. If the analysis of a piano note outputs 20 spectral peaks, we can safely guess that these peaks will reflect the harmonic makeup of the piano timbre (in other words, a harmonic series). If the twenty frequencies of the series are multiplied by a factor of 1.0 , the series is left unchanged. Multiplication by a factor of 0.5 will compress the harmonic series, producing a new fundamental (based on the psychoacoustic "missing fundamental" phenomena). Likewise, a factor of 2.0 will produce an expansion of the harmonic series. Very slight compression (e.g. 0.9999) or expansion (e.g. 1.0009) produces slight de-tunings of the harmonic series, and other compression/expansion factors can produce inharmonic sounds. The compression/expansion factor moves over the course of the section from extremely expanded to extremely compressed (following the tendency of the piano arpeggios), and is controlled within a minimum/maximum range by the piano dynamics. Throughout most of the section, the *jack~* object outputs new spectral peak values every 16 milliseconds. (In this way, the analysis windows are linked like the frames of a movie.) At other times, the *jack~* output is frozen in a kind of "spectral snapshot" of the piano timbre, the frequencies are left unchanged, and the amplitudes vary based on a continuing FFT analysis of the piano. During the course of this section, sinusoidal waveforms are replaced in the oscillators by other, more complex waveforms. This gradual replacement of waveforms takes place based on the frequency of piano attacks, which increases and then decreases arch-like over the course of the section. Other DSP, used sparingly in this section, includes: the Karplus-Strong module, granular sampling, reverberation, and spatialization.

2.3 Section II

The piano part of section II, approximately four minutes in duration, begins with the same material found at the end of section I. Within a very restrained tessitura, rhythmic syncopations and an imitative, contrapuntal texture develop. The electronic part is essentially an extension of this contrapuntal texture. Four-second samples of the piano part are recorded on-the-fly and played back using granular techniques. The playback respects the order of the samples and does not transpose the material, but the speed of traversal in the sample space is controlled by the piano dynamics. The playback read pointer moves back and forth through the samples (producing forwards and backwards playback of the sound) in a continuous fashion within a set of elastic barriers. This "scrubbing" technique keeps the material of the four-second samples sounding for a given length of time as the player moves forward in the piano score recording new material at specific intervals, thereby creating contrapuntal relationships between the electronic and instrumental material. Midway through this section a kind of call-and-response between the piano and the electronics begins making use of convolution of the piano with an FM pair. The FM modulation index is controlled by the piano dynamics and the pitch ratio of the FM pair is controlled by the pitch of the piano. The delay between call and response is compressed over time until the convolution is no longer a separate "voice", but acts as an immediate transformation of the piano timbre. Other DSP techniques used in this section include: standard sample playback of pre-recorded samples, and spatialization based on pitch and amplitude analysis of the piano input. For the most part, the piano writing continues in a restrained interval space, while this entire interval space is continuously transposed in the frequency range.

2.4 Section III

Section III lasts only one minute. The first piano notes of section III are analyzed by *jack~* using the spectral snapshot technique, so that the pitches of the 20 oscillators remain static during the entire section. The amplitudes of the oscillators are controlled by continuous FFT analysis of the piano. The oscillator waveforms are not sinusoidal, but instead, a complex waveform is employed, and the output of the oscillators is sent into a bank of variable delays along with sample playback of pre-recorded piano sounds. These pre-recorded sampled are recordings of sounds played inside the piano. The piano part begins on a single repeated note, expands pitch-wise while exploring irregular rhythmic accents, and ends with writing which is reminiscent of section I. Formally, section III falls exactly at the mid-point of *MPC* and acts as a divider between the two halves of the composition.

2.5 Section IV

In section IV, spectral snapshots using *jack~* are taken of the attacks of chordal and arpeggiated figures in the piano part. These snapshots produce a kind of frozen resonance of the piano voicings. For the most part, the piano writing is divided into higher-pitched, widely spaced chords and arpeggios, contrasted with lower-pitched, very closely spaced and accented chords, clusters, and dyads. The higher-pitched chords and arpeggios are echoed in the oscillators. There is no compression or expansion of the spectrum, just a continuation or extension of the piano resonances using sinusoidal waveforms. Conversely, the lower-pitched material is compressed or expanded so that the timbres of the oscillators are more complex and percussive. This alternation and juxtaposition of purer resonances versus inharmonic, percussive sounds is extended later in the section to an exploration of most of the transformative and generative modules used in the piece.. Harmonization, convolution, random amplitude modulation, flanging, ring modulation, filtering, variable delay, reverberation, spatialization, and sampling are all alternately juxtaposed. The section ends with continuous analysis/resynthesis which contrasts with the spectral snapshots exploited throughout most of the section.

2.6 Section V

The final section, approximately four minutes in duration, acts as a kind of instrumental and electronic finale. Instrumentally, the material of the final gesture of the piece (the last two measures) is the only material developed throughout the section. This final gesture covers almost five octaves. The electronics employed throughout the section are based on the pitch range sounding in the piano. The first half of the section employs convolution in two ways: convolution with an FM pair is used to transform the higher piano notes, while convolution with granular sampling is used to transform the lower notes. Piano samples are granulated, so that the convolution has a very strong response in the lower frequencies, corresponding to the resonant frequencies of a grand piano. In addition to convolution, the second part of section V employs granular sampling (reminiscent of the electronic introduction) and spatialization with doppler shift. The complexity of the electronic part grows as transformations of the granular sampling and convolution are increased over time.

Conclusion

Analysis/resynthesis and convolution, both which require spectral domain analysis, have a great deal to offer the musician in the domain of real-time DSP. As computer power increases, richer results in these domains can be exploited. Multi-voiced, real-time FFT-based vocoders, analysis/resynthesis with larger numbers of oscillators, algorithms like the FFT-1, and other spectral representations, like wavelets, can be exploited in a real-time context, Real-time is beginning to rival the richness and power that was available in the past on non-realtime systems. [Lippe, 1996].

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