Real-time Cross-Adaptive Audio Effects

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Abstract. This work focuses on solving the problems associated with the implementation of the system supporting real-time cross-adaptive audio effects, especially when low-latency audio streams processing is required, and complex adaptive audio effects are used over large number of channels.

Keywords: Adaptive audio affects, Low latency, multiple channels.

1 INTRODUCTION

In 2000, James A. Moorer in “Audio in the New Millennium” [1] described three stages in the music production development. In the past the profession audio concentrated on “**fidelity of reproduction**”. The main goal was to reproduce the sound as accurate as perceptible in live concert environments. At current, it is “**supernatural recording**” stage, in which recorded sound is extensively crafted artificially. Based on his extrapolation, in the future, it could be the era of “**intelligent assistant**”, that is audio engineer might deal with thousands channels with million points FFT effects in real time. The amount of information is beyond capability of any proficient audio engineer. Therefore the intelligent assistant is needed to automate the processes and enable the intelligent audio production.

The real-time intelligent audio production supported by underpinning adaptive audio effects (A-DAFX) based on audio feature extractions as shown in the Figure 1.

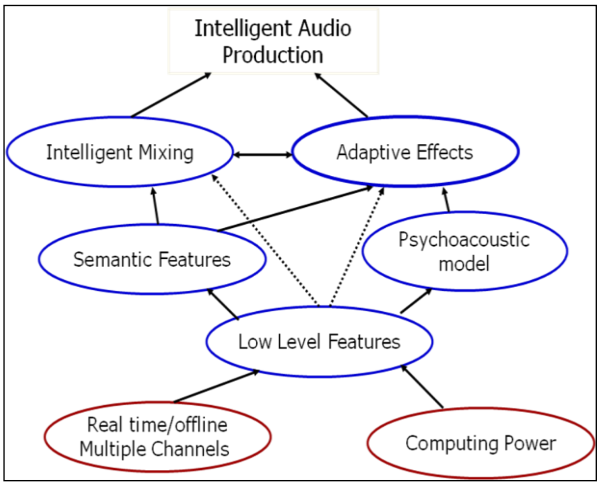

FIGURE 1. Intelligent audio production

Adaptive audio effects are the intelligent sound effects which are automatically controlled by the sound features [2]. Cross adaptive audio effects (XA-DAFX) are adaptive audio effects which extract and analyse features from multiple input channels or sound sources in order to generate the effects. In particular, XA-DAFX produces multi-dimensional audio effect control parameters which may be used to mimic the actions of an audio engineer when he or she accomplishes complex mixing tasks [3].

2 CHALLENGES

In some live performance or recording monitoring environments, the requirements of audio processing latency can be very low (for example less than 10ms) [4]. It could be the challenge to use real-time cross adaptive audio effects in such environment especially when audio feature extractions can be computational costly if large number of channels are applied and the feature updates at high rate, e.g. constant Q updating at sampling frequency.

Normally the high computational cost multi-channel feature extractions will not affect the real time audio processing path when appropriate side chain is adopted in implementation architecture. However it is not the case when feature extractions needs to be synchronised with audio processing chain for some applications such as audio effects based on beat tracking [5]. The conceptual system architecture to support XA-DAFX can be depicted as in Figure 2.

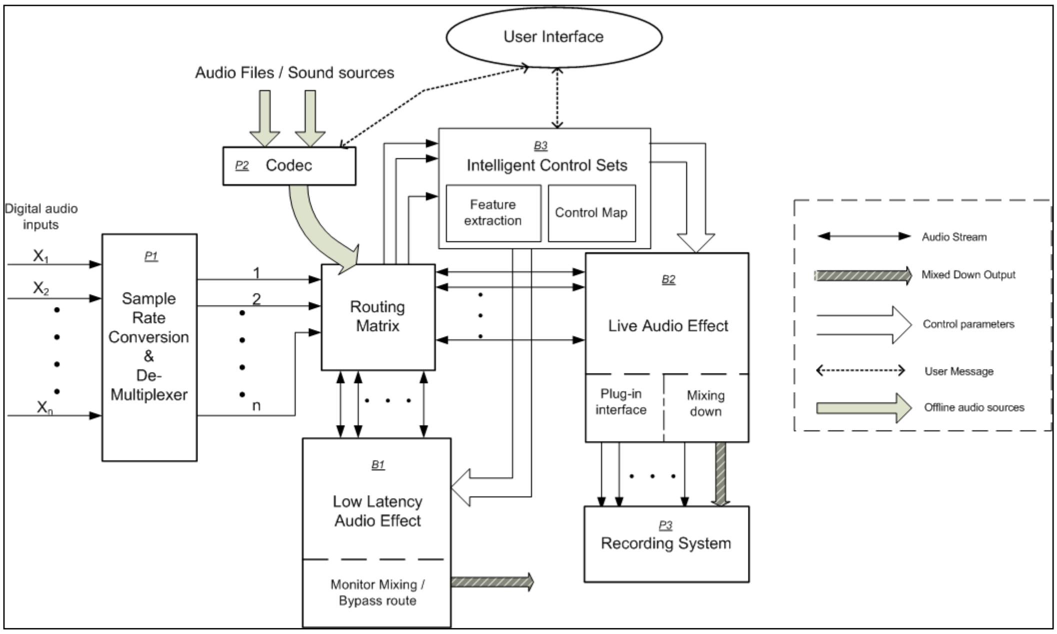

FIGURE 2. System architecture supporting XA-DAFX

The extrapolation made by James A Moorer in [1] in terms of the computing power is based on Moore's Law of raw computing power speeding up. However since 2006 the ride from faster CPU is over due to the physical limit of clock rate. And the multi-core parallel computing moved into personal computer arena in order to sustain the Moore's Law. But the increasing the number of CPU cores doesn't guarantee performance gain without extra efforts being made [6].

2 APPROACHES

Multi channel feature extractions are computational costly under real-time constrains especially when synchronisation with audio processing path is needed. In order to gain the maximum performance and ensure the future scalability, the multi-level parallelism needs to be considered such as hardware, operating system, low level DSP libraries and applications.

The research focuses on the framework of application layer in hardware / software implementation such as digital audio workstation (DAW) to support XA-DAFX with future scalability and considerations of developments of other layers. It includes the following main objectives:

- Evaluate the low latency algorithm sets for audio effects and features extractions.
- Develop new schemes for scalable real-time multichannel intelligent audio effects.
- Software architecture to support the stability of constant low latency in the audio signal path, especially when the flexibility of processing, routing, synchronising, and feature extraction over multiple channels is needed.

3 RESULTS

The hypothesis is made that with modern DAWs hosts, operating systems, audio APIs and recent audio hardware codecs, the intelligent subsystem and multiple audio processing paths should not affect the real time audio processing path, even when the CPU load is coming from the audio application itself.

If this hypothesis can be proved, that indicates the underpinning layers have provided the potential infrastructure for application layer to support real-time low latency cross-adaptive audio effects, especially when synchronisation of multi channel and feature extractions is not needed.

Lack of literature and outdated research results [7][8] in this particular area led us to conduct the experiments of the latency measurements on DAWs. The comprehensive discussion of the test results is presented in the paper “Audio latency measurement for desktop operating systems with onboard soundcards” for the 128th AES Convention.

In general the test results indicate the stability of latency path of audio processing can be maintained regardless the CPU load. However the tested DAW hosts still lack in supporting flexible feature extractions, routing and synchronisation over multiple channels [9].

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