Real-time Cross-Adaptive Audio effects

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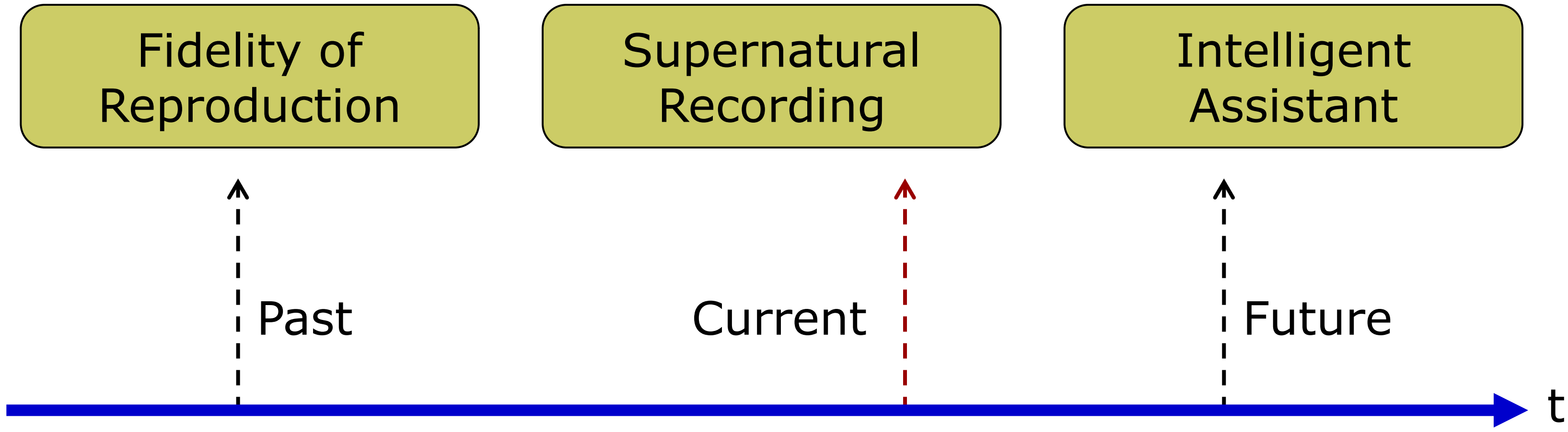
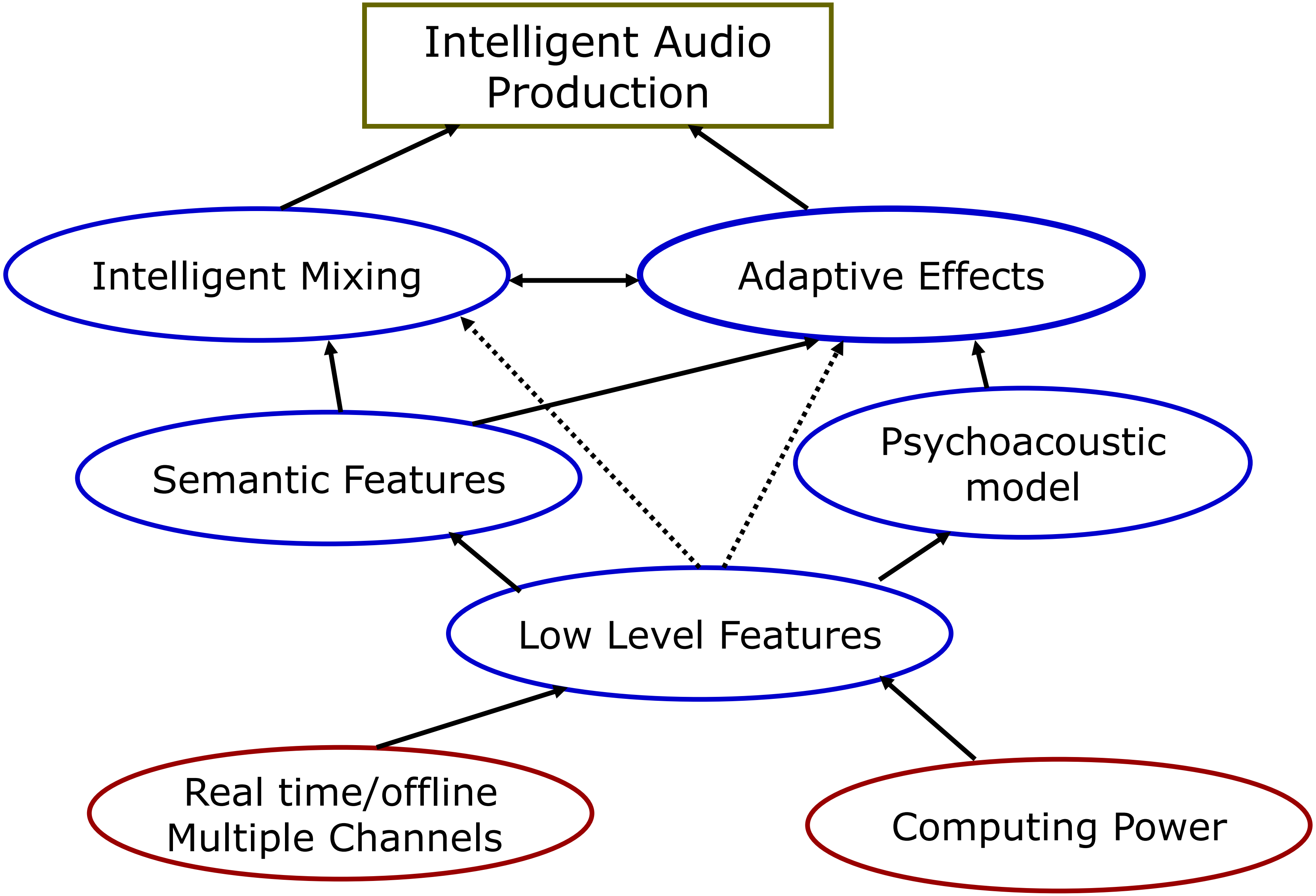
“Audio in New Millennium”



James A. Moorer’s extrapolation for future:

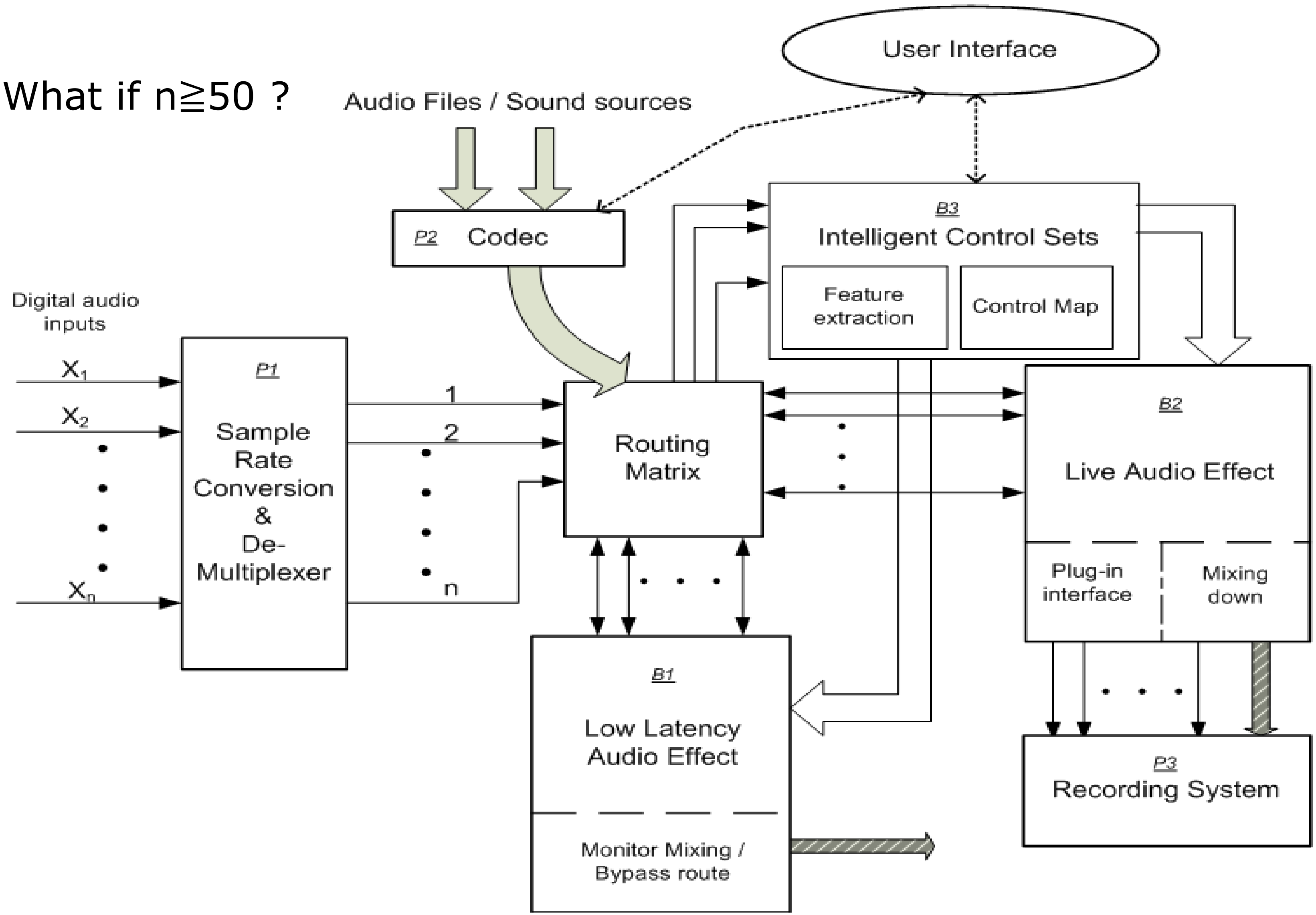
1. Processing 9000 Channel in Real-time.
2. Doing million points FFT.
3. Beyond capability of a proficient audio engineer.

What makes Intelligent Assistant in audio Production?



System supporting real-time XA-DAFX

What if n≧50 ?



Latency problem in real-time applications

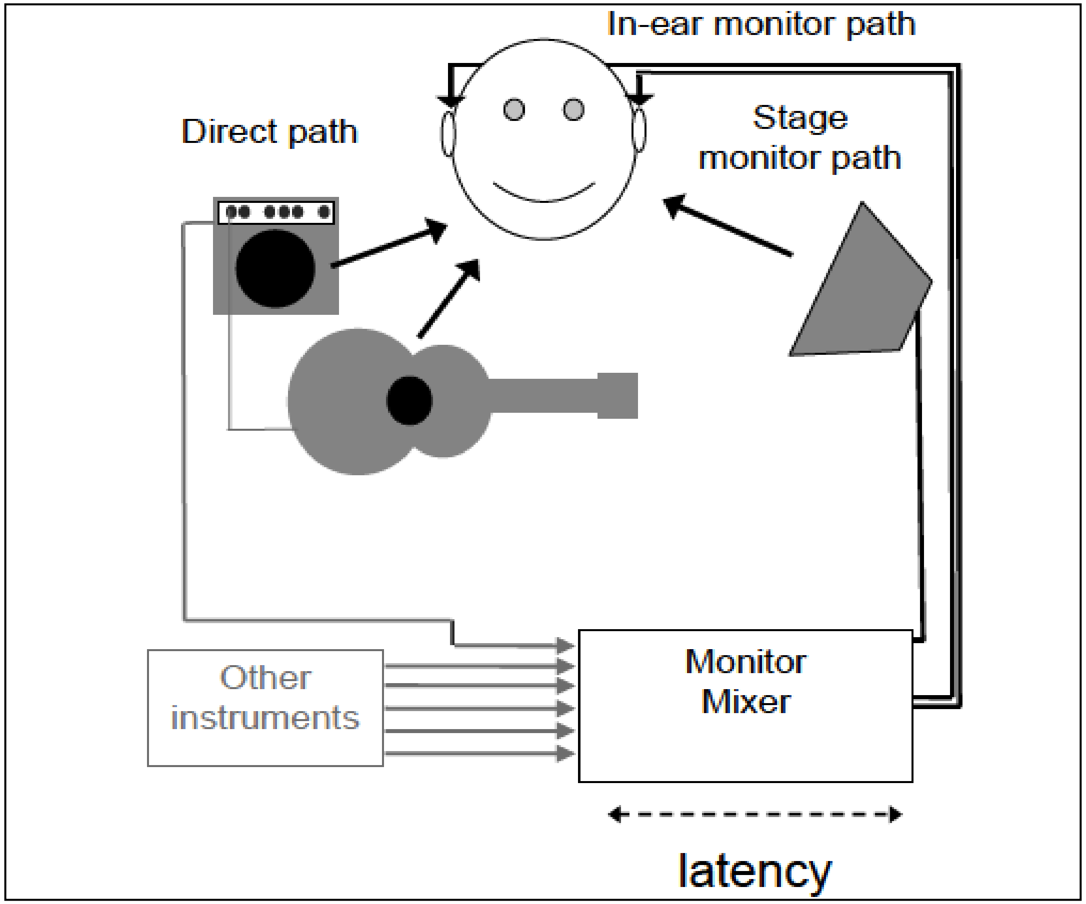


Figure produced by Rob Clark, Allen-Heath Ltd.

Live monitoring latency requirements

- Lester and Boley (2007) in the paper "The Effects of Latency on Live Sound Monitoring" indicates the latency requirements of different types of performers:
  - Keyboard player < 30ms
  - Vocalist <10ms
  - Saxophone player < 5-7ms
  - ……
- Imaging intelligent assistant used in this scenario.

Challenges

- Features need to be extracted at real-time, even at sampling rate to provide finest granularity.
- Audio effects need to be applied into audio processing chain without introducing great deal of latency.
- Multiple channels and processing stages need to be synchronised.

Inherent computational problem

- The Moorer’s extrapolation is based on sustainability of the Moore’s Law.
- In 2006, the CPU has the power wall and moved into multi-core era.
- Increasing the number of CPU cores doesn’t guarantee performance gain.
- The software architecture of multi-channel real-time audio processing with intelligent assistant needs to be looked.

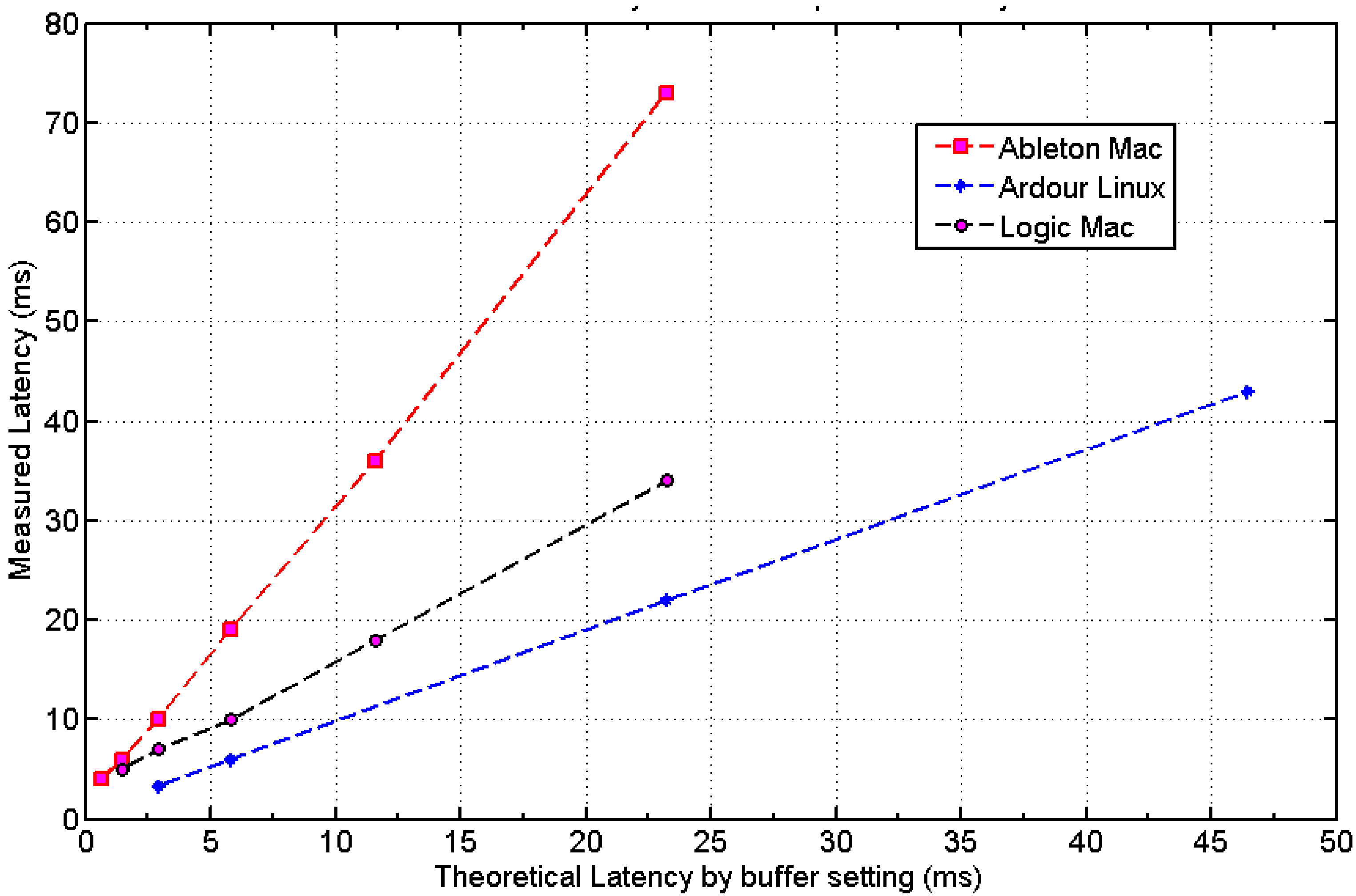
Hypothesis of DAW

- With modern DAWs hosts, OSes, Audio stacks and hardware codec, 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.
- Lack of literature and outdated research results in this particular area led us to conduct the experiments of the latency measurements on DAWs.

Results

Host	OS (API)	Without Load	With Outside load	With Audio load
2.b	WinXP (DirectX)	73ms (buffer 512)	81ms (buffer 512)	104ms (buffer 512)
2.a	OS X (CoreAudio)	4ms (buffer 14*2)	4 ms (buffer 14*2)	5.80ms (buffer 14*2)
4	Linux (ALSA)	3.31ms	3.31ms	error
		22ms (buffer 512*2)	22ms (buffer 512*2)	22ms (buffer 512*2)

Theory v.s. Measurement



Conclusion

- 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.

Future Work

- Measure latency of different digital audio effects and various implementations, sample level accuracy – "ChucK"?.
- Benchmarking real-time feature extractions: Sliding DFT in "libxtract" or "aubio"?
- Software and hardware architecture for effective audio processing parallel computing?

Thanks

- Questions?