Master's Department of Engineering Sciences

Name of the unit: Introduction to sound processing

Lecturer: Henri BOUTIN (henri.boutin@sorbonne-universite.fr)
Course code: UM4RBI11-Son1
Student workload: 12h of lectures, 6h of tutorial sessions, 10h of lab sessions@
Credits: 3 ECTS

Specialization tracks:				
	- **E3A** :	☐ CIMES	☐ Syscom	☐ IPS
	- **AR** :	☐ ROBOT	☒ SI	☐ MeDH
	- **MECA** :	☐ MS2	☐ MF2A	☐ CompMech
		☐ ACOU	☐ EE	☐ EE APP

Semester offered: ☒ S1 ☐ S2 ☐ S3 ☐ S4
Language of instruction: ☒ French ☐ English
Targeted audience: ☒ Engineering Sciences students ☐ Other (specify):
Localization : ☒ PMC Campus ☐ Other (specify):

Course Overview

This course aims to train students in the analysis and processing of an audio signal in time and frequency. The teaching focuses on the analysis, modeling and coding of speech and music signals.
Keywords: Signal Processing, Audio Signal, Music, Speech, Time-Frequency Analysis, Short-Term Fourier Transform, Overlap Add Algorithm, Source-Filter Model, LPC (Linear Predicting Coding), Cepstre, Perception, Coding, Compression.

Prerequisites

Students should have previously acquired the following prerequisites to follow this course:

- Analog signal processing: convolution, series and continuous-time Fourier transform;
- Discrete signal processing: Discrete Signal Fourier Transform, Discrete Fourier Transform;

Intended Learning Outcomes

By the end of this course, students will be able to:

1. Identify the characteristics of an audio signal in the time domain: periodicity / attacks / transients
2. Identify the characteristics of an audio signal in the frequency domain: timbre, spectral centroid, etc.
3. Master the theoretical tools of time-frequency analysis to make the link between these 2 types of characteristics: TF, TFSD, TFD and TFCT
4. Knowing how to analyze and synthesize a speech signal with a Source-Filter model
5. Distinguish between descriptors and parameters in speech signal models
6. Know and understand how an audio signal coding system works
7. Know how to optimize the encoding and compression, lossless or lossy, of an audio signal
8. Know how to process a human audio signal, production and perception devices in humans
9. Know how to analyze, characterize and model a musical signal using timbre descriptors
10. Knowing how to identify a musical structure by processing similarity matrices
11. Know how to evaluate the analysis algorithms implemented

Indicative Teaching Sequence and Methods

Week	C/TD/TP*	Content	Preparation	Learn. outc.
S1	C1	Production chain / acquisition / restitution / perception AUDIO		AAV1-2 / 8
S2	C2 / TD1	Frequency analysis of a digital audio signal	Review prerequisites: analog and discrete signal processing	AAV1-3
S3	C3 / TP1	Time-frequency analysis of an audio signal	Preparation TP1	AAV1-3 / 5 / 9
S4	C4 / TD2	Sound production and perception in humans		AAV4-5 / 8 / 1-2
S5	C5 / TP2	Application to the encoding of an audio signal	Preparation TP2	AAV6-7 / 1-2
S6	C6 / TP3	Automatic analysis of musical signals	review C3 + TP3 preparation	AAV9-11 / 3
S7	TD3		review C4 and C5	AAV4-5
S8	TP4		review C4 and C5 + preparation TP4	AAV6-8
S9	TP5		TP5 preparation	AAV1-3 / 9

* C/TD/TP respectively corresponds to lectures, tutorials and lab sessions.

Sequence of the unit:
The UE takes place in 3 sequences:
- Sequence 1: Time-frequency analysis of an audio signal (C1-3 + TD1/2 + TP1-3 + DC)
- Sequence 2: Production and perception of sound in humans and application to the coding of an audio signal (C4/5 + TD3 + TP4)
- Sequence 3: Automatic analysis of musical signals (C6 + TP5 + ER + Exam de TP)

Indicative Assessment of Intended Learning Outcomes (1st session)

Week	Individ./group	In-person/remote	Type of exam	Evaluated outcomes	Scale %
S4	Individual	In-person	Written	AAV 1-3	20 %
S8	Individual	In-person	Written	AAV 4-11	40 %
S10	Individual	In-person	Practical	AAV 1-3 / 6 - 9	40 %

2nde session of exam

Session	Individ./group	In-person/remote	Type of exam	Evaluated outcomes	Scale %
2	Individual	In-person	Written	AAV 1-11	60 %
1	Individual	In-person	Practical	AAV 1-3 / 6 - 9	40 %

Bibliographic references

- Blanchet & Charbit, 2001, "Signaux et Images sous Matlab". Hermes Sciences.
- Kahrs, 1998, "Applications of digital signal processing to audio and acoustics", Kluwer Academic Publishers.
- Hartmann, 1996, "signal, sound and sensation", Springer-Verlag
- Hayes, 1996, "Statistical Digital Signal Processing", John Wiley
- Imai & Abe, "Spectral Envelope Extraction by Improved Cepstral Method"

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