The Constant-Q Transform Spectral Envelope Coefficients: A Timbre Feature Designed for Music

I. Scope

TIMBRE is the attribute of sound which makes, for example, two musical instruments playing the same note sound different. It is generally associated with the spectral (but also temporal) envelope and is typically assumed to be independent from the pitch (but also the loudness) of the sound [1]. Typical attempts to characterize the timbre of musical data is to use to the mel-frequency cepstral coefficients (MFCC) [2], a feature original designed for speech recognition, while more recently, data-driven approach ...

In this article, we will show how to compute a timbre feature that is well-adapted to musical data. The feature will be derived from the constant-Q transform (CQT), a log-scaled frequency transform which matches the notes of the Western music scale [3], [4].

Decompose the CQT into a pitch-invariant spectral envelope and an energy-normalized pitch component.

CQT spectral envelope coefficients (CQT-SEC)

The CQT-SEC will compare with the , on the NSynth dataset, a publicly-available dataset of musical notes [5].

II. Relevance

III. Prerequisites

Basic knowledge of audio signal processing and music information retrieval is required to understand this article, in particular, concepts such as the Fourier transform, convolution, spectral envelope, pitch, CQT, and MFCC.

IV. Problem Statement

V. Solution

convolution theorem: [6].

A. Observations

Assumption: A log-spectrum, such as the CQT-spectrum, can be represented as the convolution of a pitch-invariant log-specrtal envelope component (= timbre) and a envelope-normalized pitch component.

- A pitch change in the audio translates to a linear shift in the log-spectrum.
- The Fourier transform (FT) of a convolution of two functions is equal to the point-wise product of their FTs (convolution theorem).
- The magnitude FT is shift-invariant.

VI. Numerical Example

VII. What We Have Learned

We have shown that ...

VIII. Author

Zafar Rafii (zafarrafii@gmail.com) received a PhD in Electrical Engineering and Computer Science from Northwestern University in 2014, and an MS in Electrical Engineering from both Ecole Nationale Superieure de l'Electronique et de ses Applications in France and Illinois Institute of Technology in the US in 2006. He is currently a senior research engineer at Gracenote in the US. He also worked as a research engineer at Audionamix in France. His research interests are centered on audio analysis, somewhere between signal processing, machine learning, and cognitive science, with a predilection for source separation and audio identification.

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