MIREX 2016:
VAMP PLUGINS FROM THE CENTRE FOR DIGITAL MUSIC

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ABSTRACT

In this submission we offer for evaluation several audio feature extraction plugins in Vamp format.

Most of these plugins were also submitted to the 2013, 2014, and 2015 editions of MIREX. The majority are unchanged here and may offer a useful baseline for comparison across years. One category (Audio Downbeat Estimation) sees a new submission, although it is not a new plugin. One plugin (Chordino) has been corrected after the side-effects of an earlier bug fix made it perform unexpectedly badly last year. The rest are unchanged from 2015.

Some of these plugins represent efficient implementations based on modern work, while others are no longer state-of-the-art and were developed a few years ago. The methods implemented in this set of plugins are described in the literature and are referenced throughout this paper. All of the plugins are written in C++ and have been published under open source licences, in most cases the GPL.

1. INTRODUCTION

The Vamp plugin format [1] was developed at the Centre for Digital Music (C4DM) at Queen Mary, University of London, during 2005-2006 in response to a desire to publish work in a form that would be immediately useful to people outside this research field. The Vamp plugin format was published with an open source SDK, alongside the Sonic Visualiser [3] audio analysis application which provided a useful host for Vamp plugins.

In subsequent years the Vamp format has become a moderately popular means of distributing methods from the Centre and other research groups. Some dozens of Vamp plugins are now available from groups such as the Music Technology Group at UPF in Barcelona, the Sound and Music Computing group at INESC in Porto, the BBC, and others, as well as from the Centre for Digital Music.

The plugins submitted for this evaluation are provided as a set of dynamic library files. Those with names starting "QM" are all provided in a single library file, the QM Vamp Plugins set, made available in binary form for Windows, OS/X, and Linux from the Centre for Digital Music's download page. [2] These plugins come from a number of authors who are credited in this abstract and in the plugins' accompanying documentation.

In addition to the QM Vamp Plugins set, this submission contains a number of separate plugins: the Chordino and Segmentino plugins from Matthias Mauch; the BeatRoot Vamp Plugin from Simon Dixon; OnsetsDS from Dan Stowell; and the Silvet note transcription plugin from Emmanouil Benetos and Chris Cannam.

The plugins are all provided as 64-bit Linux shared objects depending on GNU libc 2.15 or newer and GNU libstdc++ 3.4.15 or newer. Sonic Annotator v1.1 is also required [3] in order to run the task scripts.

For an overview of this submission across all of the tasks and plugins it covers, please see the relevant repository at the SoundSoftware site. [4]

2. SUBMISSIONS BY MIREX TASK

2.1 Audio Beat Tracking

2.1.1 QM Tempo and Beat Tracker

The QM Tempo and Beat Tracker [5] Vamp plugin analyses a single channel of audio and estimates the positions of metrical beats within the music.

This plugin uses the complex-domain onset detection method from [8] with a hybrid of the two-state beat tracking model proposed in [5] and a dynamic programming method based on [9].

To identify the tempo, the onset detection function is partitioned into 6-second frames with a 1.5-second increment. The autocorrelation function of each 6-second onset detection function is found and this is then passed through a perceptually weighted comb filterbank [5]. The successive comb filterbank output signals are grouped together into a matrix of observations of periodicity through time. The best path of periodicity through these observations is found using the Viterbi algorithm, where the transition matrix is defined as a diagonal Gaussian.

Given the estimates of periodicity, the beat locations are recovered by applying the dynamic programming algo-

[1] http://vamp-plugins.org/

[2] http://vamp-plugins.org/plugin-doc/qm-vamp-plugins.html
[3] http://code.soundsoftware.ac.uk/projects/sonic-annotator/
[4] http://code.soundsoftware.ac.uk/projects/mirex2013/

rithm [9]. This process involves the calculation of a recursive cumulative score function and backtrace signal. The cumulative score indicates the likelihood of a beat existing at each sample of the onset detection function input, and the backtrace gives the location of the best previous beat given this point in time. Once the cumulative score and backtrace have been calculated for the whole input signal, the best path through beat locations is found by recursively sampling the backtrace signal from the end of the input signal back to the beginning.

The QM Tempo and Beat Tracker plugin was written by Matthew Davies and Christian Landone.

2.1.2 BeatRoot

The BeatRoot Vamp plugin [5] is an open source Vamp plugin library that implements the BeatRoot beat-tracking method of Simon Dixon [6]. The BeatRoot algorithm has been submitted to MIREX evaluation in earlier years [7]; this plugin consists of the most recent BeatRoot code release, converted from Java to C++ and modified for plugin format.

The BeatRoot plugin was written by Simon Dixon and Chris Cannam.

2.2 Audio Key Detection

2.2.1 QM Key Detector

The QM Key Detector Vamp plugin continuously estimates the key of the music by comparing the degree to which a block-by-block chromagram correlates to stored key profiles for each major and minor key.

This plugin uses the correlation method described in [11] and [10], but using different tone profiles. The key profiles used in this implementation are drawn from analysis of Book I of the Well Tempered Klavier by J S Bach, recorded at A=440 equal temperament, as described in [17].

The QM Key Detector plugin was written by Katy Noland and Christian Landone.

2.3 Audio Chord Estimation

2.3.1 Chordino

The Chordino plugin [6] was developed following Mauch's 2010 work on chord extraction, submitted to MIREX in that year [15]. While that submission used a C++ chroma implementation with a MATLAB dynamic Bayesian network as a chord extraction front-end [14], Chordino is an entirely C++ implementation that was developed specifically to be made freely available as an open-source plugin for general use.

The method for the Chordino plugin has two parts:

NNLS Chroma — NNLS Chroma analyses a single channel of audio using frame-wise spectral input from the Vamp host. The spectrum is transformed to a log-frequency spectrum (constant-Q) with three bins per semitone. On this representation, two processing steps are performed: tuning, after which each centre bin (i.e. bin 2, 5, 8,) corresponds to a semitone, even if the tuning of the piece deviates from 440 Hz standard pitch; and running standardisation: subtraction of the running mean, division by the running standard deviation. This has a spectral whitening effect.

The processed log-frequency spectrum is then used as an input for NNLS approximate transcription using a dictionary of harmonic notes with geometrically decaying harmonics magnitudes. The output of the NNLS approximate transcription is semitone-spaced. To get the chroma, this semitone spectrum is multiplied (element-wise) with the desired profile (chroma or bass chroma) and then mapped to 12 bins.

Chord transcription — A fixed dictionary of chord profiles is used to calculate frame-wise chord similarities. A standard HMM/Viterbi approach is used to smooth these to provide a chord transcription.

Chordino was written by Matthias Mauch.

2.4 Audio Onset Detection

2.4.1 QM Note Onset Detector

The QM Note Onset Detector Vamp plugin estimates the onset times of notes within the music. It calculates an onset likelihood function for each spectral frame, and picks peaks in a smoothed version of this function.

Several onset detection functions are available in this plugin; this submission uses the complex-domain method described in [8].

The QM Note Onset Detector plugin was written by Chris Duxbury, Juan Pablo Bello and Christian Landone.

2.4.2 OnsetsDS

OnsetsDS [7] is an onset detector plugin wrapping Dan Stowell's OnsetsDS library [8], described in [18].

OnsetsDS was designed to provide an FFT-based onset detection that works very efficiently in real-time, with a fast reaction time. It is not tailored for non-real-time use or for any particular type of signal.

The OnsetsDS plugin was written by Dan Stowell and Chris Cannam.

2.5 Multiple Fundamental Frequency Estimation and Tracking

2.5.1 Silvet

Silvet (for Shift-Invariant Latent Variable Transcription) [9] is a Vamp plugin for automatic music transcription, using a method based on that of [2]. It produces a note transcription as output, and we have included a script to transform this into a framewise output as well, in order to make it available for framewise evaluation as well as note-tracking evaluation.

Silvet uses a probablistic latent-variable estimation method to decompose a Constant-Q time-frequency matrix into note

[5] http://code.soundsoftware.ac.uk/projects/beatroot-vamp/

[6] http://isophonics.net/nnls-chroma

[7] http://code.soundsoftware.ac.uk/projects/vamp-onsetsds-plugin/

[8] http://onsetsds.sourceforge.net/

[9] http://code.soundsoftware.ac.uk/projects/silvet/

activations using a set of spectral templates learned from recordings of solo instruments. The method is thought to perform quite well for clear recordings that contain only instruments with a good correspondence to the known templates. Silvet does not contain any vocal templates, or templates for typical rock or electronic instruments.

The method implemented in Silvet is very similar to that submitted to MIREX in 2012 as the BD1, BD2 and BD3 submissions in the Multiple F0 Tracking task of that year [1]. In common with that submission, and unlike the paper cited at [2], Silvet uses a simple thresholding method instead of an HMM for note identification. However, Silvet follows [2] rather than [1] in including a 5-bin-per-semitone pitch shifting parameter.

The Silvet plugin was written by Chris Cannam and Emmanouil Benetos.

2.5.2 Silvet Live

The Silvet Live submission uses the Silvet plugin in its "Live" mode. This has somewhat lower latency than the default mode, and is much faster to run. This is mainly a result of using a reduced 12-bin chromagram and corresponding instrument templates, making this conceptually a very simple method. Results are expected to be substantially poorer than those for the default Silvet parameters.

The Silvet plugin was written by Chris Cannam and Emmanouil Benetos.

2.6 Structural Segmentation

2.6.1 QM Segmenter

The QM Segmenter Vamp plugin divides a single channel of music up into structurally consistent segments.

The method, described in [12], relies upon timbral or pitch similarity to obtain the high-level song structure. This is based on the assumption that the distributions of timbre features are similar over corresponding structural elements of the music.

The input feature is a frequency-domain representation of the audio signal, in this case using a Constant-Q transform for the underlying features (though the plugin supports other timbral and pitch features). The extracted features are normalised in accordance with the MPEG-7 standard (NASE descriptor), and the value of this envelope is stored for each processing block of audio. This is followed by the extraction of 20 principal components per block using PCA, yielding a sequence of 21 dimensional feature vectors where the last element in each vector corresponds to the energy envelope.

A 40-state Hidden Markov Model is then trained on the whole sequence of features, with each state corresponding to a specific timbre type. This partitions the timbre-space of a given track into 40 possible types. After training and decoding the HMM, the song is assigned a sequence of timbre-features according to specific timbre-type distributions for each possible structural segment.

The segmentation itself is computed by clustering timbre-type histograms. A series of histograms are created over a sliding window which are grouped into M clusters by an adapted soft k-means algorithm. Reference histograms, iteratively updated during clustering, describe the timbre distribution for each segment. The segmentation arises from the final cluster assignments.

The QM Segmenter plugin was written by Mark Levy.

2.6.2 Segmentino

The Segmentino plugin is a C++ implementation of a segmentation method described in Matthias Mauch's paper on using musical structure to enhance chord transcription [16] and expanded on in Mauch's PhD thesis [13].

A beat-quantised chroma representation is used to calculate pair-wise similarities between beats (really: beat "shingles", i.e. multi-beat vectors). Based on this first similarity calculation, an exhaustive comparison of all possible segments of reasonable length in beats is executed, and segments are added to form segment families if they are sufficiently similar to another "family member". Having accumulated a lot of families, the families are rated, and the one with the highest score is used as the first segmentation group that gets annotated. This last step is repeated until no more families fit the remaining "holes" in the song that haven't already been assigned to a segment.

This method was developed for "classic rock" music, and therefore assumes a few characteristics that are not necessarily found in other music: repetition of harmonic sequences in the music that coincide with structural segments in a song; a steady beat; segments of a certain length; corresponding segments have the same length in beats.

Segmentino plugin was written by Matthias Mauch and Massimiliano Zanoni.

2.7 Audio Tempo Estimation

2.7.1 QM Tempo and Beat Tracker

For this task we submit the same plugin as that used in the Audio Beat Tracking task in section 2.1.1.

2.8 Audio Downbeat Estimation

2.8.1 QM Bar and Beat Tracker

The QM Bar and Beat Tracker [4] Vamp plugin estimates the positions of bar lines and metrical beat positions.

The plugin first uses the method of the QM Tempo and Beat Tracker (see section 2.1.1) to estimate beat locations. Once these have been identified, the plugin makes a second pass over the input audio signal, partitioning it into beat synchronous frames. The audio within each beat frame is down-sampled to give a new sampling frequency of 2.8kHz. A beat-synchronous spectral representation is then calculated within each frame, from which a measure of beat spectral difference is calculated using Jensen-Shannon divergence. The bar boundaries are identified as those beat transitions leading to most consistent spectral change given the specified number of beats per bar.

The plugin expects the number of beats per bar as a parameter rather than attempt to estimate this from the music. For the purposes of this submission the number of beats per bar is fixed to 4.

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