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Contents

1 Installation

The following lists the dependancies of audioguide1.2. For the moment audioguide1.2 only works on OS X. Note that, regarding numpy, recent versions of OS X (after 10.5) come with python2.7 that automatically has numpy installed. This distribution of audioguide1.2 comes with a precompiled version of pysdif for 64-bit python and, on more current machines, should run out of the box. Here is a complete list of the resources that audioguide1.2 requires on your computer:

pysdif To compile install download the sdif library and configure, make and install. Then download my patched version of the pysdif module. 'cd' into the directory when unzipped and run: 'python2.7 setup.py install'. You're now setup to use pysdif in python2.7.

numpy Most python2.7 installations come with numpy, a numerical computation module. Upgrading to the latest python2.7 should get you there. If you don't have it, you can download the source code or a binary installer here.

csound6 Needed only if you would like audioguide1.2 to automatically render concatenations (which you probably do). Download an installer from here.

matlibplot (optional) Install this python module to enable graphing of descriptors and soundfile segmentation. Get it from here.

supervp (optional) Used to stretch a target before concatenating.

2 Really Quick Start

```
$> python agSegmentSf.py examples/lachenmann.aiff
$> python agConcatenate.py examples/01-simplest.py
```

3 Quick Start

Using audioguide1.2 comes down to interacting with python scripts in the audioguide1.2 folder. For soundfile concatenation, one script does segmentation of corpus soundfiles. A second script performs concatenation based on a variety of variables found in a options file. While you do not need to know how to write Python code in order to use audioguide1.2, it is not a bad idea to know some of Python's basic syntax.

The reason that segmentation and concatenation are separated into discrete steps that I find is useful to fine-tune the segmentation of corpus sounds *before* using them in a concatenation. Soundfile segmentation is a difficult technical problem and should remain conceptually and aesthetically open-ended. I have yet to find an algorithm that does not require adjustments based on the nature of the sound in question and the intention of the user as to what a segment should be.

3.1 Segmenting Corpus Soundfiles

Note: If you only want to use folders of sounds that have been pre-segmented into individual files, you can skip[1] the *Segmenting Corpus Soundfiles* section and proceed to the concatenation section.

The script you use to segment your corpus files with a script called 'agSegmentSf.py'. AgSegmentSf.py automatically creates a textfile which denotes the start and stop times of sound segments in a continuous audiofile. Once you find a segmentation that you're happy with you're ready to use this sound file in the agConcatenate.py script. Keep in mind that you do not *need* to use agSegmentSf.py if you do not want to – alternatively you could:

1. use whole soundfiles as segments
2. create segmentation files by hand
3. create segmentation files with other software as long the textfile is written in the same format as audioguide1.2's.

To segment a corpus file, 'cd' into the audioguide1.2 folder and run the following command:

```
$ > python2.7 agSegmentSf.py examples/lachenmann.aiff
```

audioguide1.2 will think for a second, and then output the following data detailing the segmen-

[1] But make sure tell audioguide1.2 not to search for segmentation textfiles by setting the corpus attribute wholeFIle=True. See the Manipulating How Directories Are Read subsection of the CORPUS options section for more info.

tation of this audiofile:

```
------------------------ AUDIOGUIDE SEGMENT SOUNDFILE ------------------------

Evaluating /Users/ben/Documents/audioguide1.1/examples/lachenmann.aiff from 0.00-64.65

AN ONSET HAPPENS when
The amplitude crosses the Relative Onset Trigger Threshold: -40.00 (-t option)


AN OFFSET HAPPENS when
1. Offset Rise Ratio: when next-frame's-amplitude/this-frame's-amplitude >= 1.30 (-r
 option)
   ...or...
2. Offset dB above minimum: when this frame's absolute amplitude <= -80.00 (minimum
 found amplitude of -260.00 plus the offset dB boost of 12.00 (-d option))


Found 144 segments
Wrote file /Users/ben/Documents/audioguide1.1/examples/lachenmann.aiff.txt
```

As a result of running this python script, audioguide1.2 automatically writes a textfile with the exact same name and path as the soundfile, but adding the extension .txt – in this case: examples/lachenmann.aiff.txt. In this case audioguide1.2 found 144 segments obtained using a triggering threshold of -40 dB, a rise ratio of 1.3 and a offset dB value of -80 (you can read more about how to alter these values and their effect in section 4).

3.2 Concatenating

Once you have segmented corpus soundfiles to your satisfaction, you are ready to call the concatenation script agConcatenate.py with a special audioguide1.2 options file as the first (and only) argument.

To run one of the examples in the examples directory, run the following command inside the audioguide1.2 directory:

```
python2.7 agConcatenate.py examples/01-simplest.py
```

..which will use the options contained in 'examples/01-simplest.py' to parameterize the concatenative algorithm. In this options file you specify a target sound, the corpus sounds, and (if you like) lots of other options that parameterize the concatenative process. When run, the 'agConcatenate.py' script will perform the following summarised operations:

1. Run an ircamdescriptor analysis of the soundfile in the TARGET variable[2].

[2]An SDIF analysis is only done once – subsequent usages of this soundfile simply read SDIF data from disk. Analysis files are stored in a directory called 'audioguide1.2/data_json/' in the audioguide1.2 folder. This directory can become quite large since these SDIF files are quite substantial in size. Removing this folder will cause all SDIF files to be recomputed.

2. Segment the target sound according to your options file. An Audacity-style label file is created in a file called 'output/tgtlabels.txt' in the output directory.

3. Run an ircamdescriptor analysis of the soundfiles in the CORPUS variable (only the first time each of these files are used).

4. (If you've specified them) Remove corpus segment according to descriptor limitations (Nothing above a certain pitch, nothing below a certain dynamic, etc.).

5. Normalise target and corpus descriptor data according to your options file.

6. Go through each target segment one by one. Select corpus segment(s) to match each target segment according to the descriptors and search passes in the SEARCH variable of your options file. Control over the layering and superimposition of corpus sounds is specified in the SUPERIMPOSE variable.

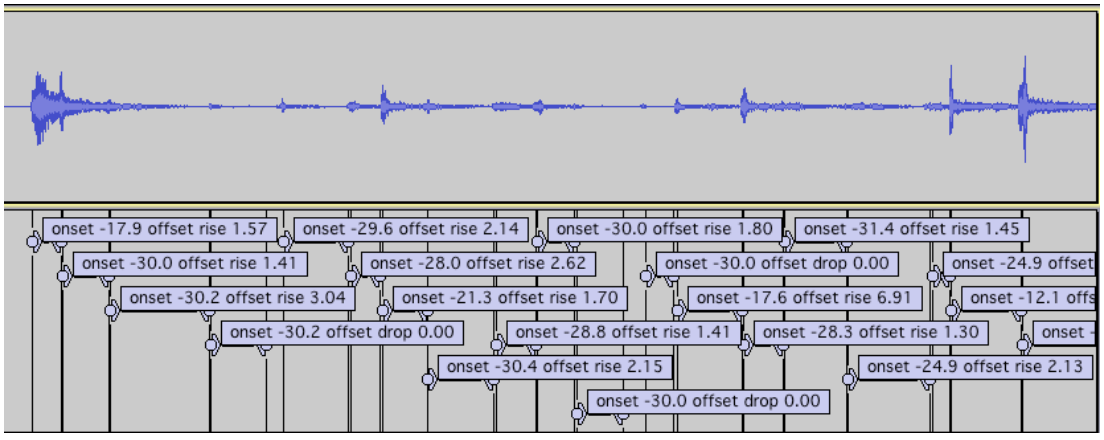
7. Write selected segments to a csound score called 'output/output.csd'. (In addition to the csd file there are many other types of outputs which you can read about in 11.3.

8. If you have csound, 'output/output.csd' is rendered with csound to create an audiofile called 'output/output.aiff'.

9. If you have csound, automatic playback of 'output/output.aiff' at the command line.

4 Corpus Segmentation

The textfiles created by agSegmentSf.py use a segmentation labeling format identical to that of the soundfile editor Audacity. So, to examine your segments, open lachenmann.aiff in Audacity, then import labels and select 'examples/lachenmann.aiff.txt'.



5 The Concatenation Options File

The options file used by the concatenate script is a python file that defines a bunch of variables. Most variables can be changed with simple assignments using the '=' symbol. For instance, to change the path of the csound output sound file, write the following in your myOptions.py file:

```
CSOUND_RENDER_FILEPATH = '/path/to/the/file/i/want.aiff' # sets the path of the csound
  output aiff file
DESCRIPTOR_HOP_SIZE_SEC = 0.02049 # change the analysis hop size
```

However, there are five custom objects that are written into the options file as well – tsf(), csf(), spass(), d() and si(). These objects take required parameters and also take keyword arguments. The following sections describe the object-style variables; Section 11 details variables changed with the '=' symbol.

6 The TARGET Variable and tsf() object

The TARGET variable is written as a tsf() object which requires a path to a soundfile and also takes the following optional keyword arguments:

```
tsf('pathtosoundfile', start=0, end=file-length, thresh=-40, offsetRise=1.5,
 offsetThreshAdd=+12, offsetThreshAbs=-80, scaleDb=0, minSegLen=0.05, maxSegLen=1000,
 midiPitchMethod='composite', stretch=1, segmentationFilepath=None)
```

start The time in seconds to start reading the soundfile.

end The time in seconds to stop reading the soundfile.

thresh Segmentation onset threshold: a value from -100 to 0. The lower the value the soft the target's amplitude can be in order to trigger a selection from the corpus. So, -12 yields fewer corpus selections that -24.

offsetRise Segmentation offset ratio: a number greater than 1. During segmentation the rise-ratio is the ratio of a frame's amplitude to the next frame's amplitude. In an active segment, if this ratio is greater than user-supplied rise-ratio, it will cause an offset.

offsetThreshAdd Segmentation offset relative threshold: a positive value in dB. This value is added to the soundfile's minimum amplitude. During segmentation, if a frame's amplitude is below this threshold, it causes a segment offset. Also see offsetThreshAbs.

offsetThreshAbs Segmentation offset absolute threshold: a negative value in dB. When segmenting the target, if a frame's amplitude is below this value it will cause an offset. This

variable is an absolute value while offsetThreshAdd is relative to the soundfile's minimum amplitude. Effectively, whichever of these two variable is closer to 0 will be the offset threshold.

minSegLen* Segmentation: the minimum duration in seconds of a target segment.

maxSegLen* Segmentation: the maximum duration in seconds of a target segment.

scaleDb* Applies an amplitude change to the whole target sound. By default, it is 0, yielding no change. -6 = twice as soft. The target's amplitude will usually affect concatenation: the louder the target, the more corpus sounds can be composited to approximate it's energy profile.

midiPitchMethod same as corpus method documented in 7.3.

stretch* Uses SuperVp to time stretch/compress the target sound before analysis and concatenation. Stretched sound files are saved and reused in the audioguide/data_stretched_sfs directory. A value of 1, the default, will cause no time stretching to be applied. 2 will double the duration of the soundfile. Note that, in order to use this option, you must set the SUPERVP_BIN variable in the audioguide/defaults.py file.

segmentationFilepath by default the Target sound is segmented at runtime. However, if you'd like to specify a user-defined segmentation, you may give a file path to this variable. Note that this file must be a textfile with the same format as corpus segmentation files.

```
TARGET = tsf('cage.aiff') # uses the whole soundfile at its given amplitude
TARGET = tsf('cage.aiff', start=5, end=7, scaleDb=6) # only use seconds 5-7 of cage.
 aiff at double the amplitude.
TARGET = tsf('cage.aiff', start=2, end=3, stretch=2) # only uses seconds 2-3, but
 stretches the sound with supervp to twice its duration before concatenation
```

7 The CORPUS Variable and csf() object

The CORPUS variable is defined as a list of csf() objects which require a path to a soundfile OR a directory. File paths and/or directory paths may be full paths or relative paths to the location of the options file you're using or a path found in the SEARCH_PATHS variable. A csf() object required the soundfile/directory name, then takes the following optional keyword arguments:

```
csf('pathToFileOrDirectoryOfFiles', start=None, end=None, includeTimes=[],
 excludeTimes=[], limit=[], wholeFile=False, recursive=True, includeStr=None,
 excludeStr=None, scaleDb=0.0, limitDur=None, onsetLen=0.01, offsetLen='30%',
 postSelectAmpBool=False, postSelectAmpMethod='power-mean-seg', postSelectAmpMin=-12,
 postSelectAmpMax=+12, midiPitchMethod='composite', transMethod=None,  transQuantize
 =0, allowRepetition=True, restrictRepetition=0.5, restrictOverlaps=None,
```

```
restrictInTime=0, maxPercentTargetSegments=None, scaleDistance=1, superimposeRule=
None, segmentationFile=None, segmentationExtension='.txt')
```

The simplest way to include a soundfile in your corpus is to use its path as the first argument of the csf() object:

```
CORPUS = [csf("lachenmann.aiff")] # will search for a segmentation file called
 lachenmann.aiff.txt and add all of its segments to the corpus
```

The simplest way to include a directory of soundfiles in your corpus is to use its path as the first argument of the csf() object:

```
CORPUS = [csf("lachenmann.aiff"), csf('piano')] # will use segments from lachenmann.
 aiff as well as all sounds in the directory called piano
```

However, as you can see above, each csf() object has *a lot* of optional arguments to give you better control over what segments are used, how directories are read and how segments are treated during concatenation.

Note: Each of these keyword arguments only *apply to the csf() object within which they are written. If you'd like to specify these parameters for the entire corpus, see 7.4.*

7.1 Manipulating which segments are added to the corpus

start Any segments which start before this time will be ignored.

end Any segments which start after this time will be ignored.

```
csf('lachenmann.aiff', start=20) # only use segments who start later than 20s.
csf('lachenmann.aiff', start=20, end=50) # only use segments who start between 20-50s.
```

includeTimes* A list of two-number lists which specify regions of segments to include from this file's list of segment times. See example below.

excludeTimes* Same as includeTimes but excludes segments in the identified regions.

```
csf('lachenmann.aiff', includeTimes=[(1, 4), (10, 12)]) # only use segments falling
 between 1-4 seconds and 10-12 seconds.
csf('lachenmann.aiff', excludeTimes=[(30, 55)]) # use all segments except those
 falling between 30-55s.
```

limit* A list of equation-like strings where segmented descriptor names are used to include/exclude segments from this file / directory.

```
csf('lachenmann.aiff', limit=['centroid-seg >= 1000']) # segments whose centroid-seg
 is equal to or above 1000.

csf('lachenmann.aiff', limit=['centroid-seg < 50%']) # only use 50% of segments with
 the lowest centroid-seg.

csf('lachenmann.aiff', limit=['power-seg < 50%', 'power-seg > 10%']) # only use
 segments whose power-seg falls between 10%-50% of the total range of power-seg's in
 this file/directory.
```

segmentationFile* Manually specify the segmentation text file. By default, audioguide1.2 automatically looks for a file with the same name as the soundfile plus the extension '.txt'. You may specify a path file (as a string), or a list of strings to include multiple segmentation files which all use the same soundfile.

segmentationExtension* Manually specify the segmentation text file extension. See above.

```
csf('lachenmann.aiff', segmentationFile='marmotTent.txt') # will use a segmentation
 file called marmotTent.txt, not the default lachenmann.aiff.txt.

csf('lachenmann.aiff', segmentationExtension='-gran.txt') # will use a segmentation
 file called lachenmann.aiff-gran.txt, not the default lachenmann.aiff.txt.
```

7.2 Manipulating How Directories Are Read

The following keyword arguments are useful when dealing with directories of files.

wholeFile* if True audioguide1.2 will use this soundfile as one single segment. If False, audioguide1.2 will search for a segmnetation file made with agSegmentSf.py.

recursive* if True audioguide1.2 will include sounds in all subfolders of a given directory.

```
csf('sliced/my-directory', wholeFile=True) # will not search for a segmentation txt
 file, but use whole soundfiles as single segments.

csf('/Users/ben/gravillons', recursive=False) # will only use soundfiles in the named
 folder, ignoring its subdirectories.
```

includeStr* A string which is matched against the filename (not full path) of each soundfile in a given directory. If part of the soundfile name matches this string, it is included. If not it is excluded. This is case sensitive. See example below.

excludeStr* Opposite of includeStr.

```
# includeStr/excludeStr have lots of uses. One to highlight here: working with sample
  databases which are normalized. Rather than having each corpus segment be at 0dbs,
 we apply a scaleDb value based on the presence of a `dynamic' written into the
 filename.
```

```
csf('Vienna-harpNotes/', includeStr=['_f_', '_ff_'], scaleDb=-6),
csf('Vienna-harpNotes/', includeStr='_mf_', scaleDb=-18),
csf('Vienna-harpNotes/', includeStr='_p_', scaleDb=-30),
# this will use all sounds from this folder which match one of the three dynamics.
```

7.3 Manipulating How Segments Will Be Concatenated

scaleDb* applies an amplitude change to each segment of this collection. by default, it is 0, yielding no change. -6 = twice as soft. Note that amplitude scaling affects both the concatenative algorithm and the csound rendering.

limitDur* limits the duration of each segment from this csf() entry. The duration of all segments over this value (in seconds) will be truncated.

onsetLen* if onsetLen is a float or integer, it is the fade-in time in seconds. If it is a string formed as '10%', it is interpreted as a percent of each segment's duration. So, onsetLen=0.1 yields a 100 ms. attack envelope while onsetLen='50%' yields a fade in over 50% of the segment's duration.

offsetLen* Same as onsetLen, but for the envelope fade out.

```
csf('lachenmann.aiff', onsetLen=0.1, offsetLen='50%') # will apply a 100ms fade in
 time and a fade out time lasting 50% of each segments' duration.
```

postSelectAmpBool* If True, audioguide1.2 will attempt to change the amplitude of selected segments to match the amplitude of the target. The default, False, makes no adjustment.

postSelectAmpMethod* Tells audioguide1.2 how to attempt to match corpus and target amplitudes. It is only meaningful if postSelectAmpMethod is True:

'power-mean-seg' will use the mean of segment powers' to change adjust selected segments amplitudes.

'power-seg' will adjust amplitudes according to peak amplitudes.

postSelectAmpMin* If postSelectAmpBool is True, this value in dB limits the lower threshold of the amplitude change. 0=no change, -6=half volume, etc.

postSelectAmpMax* If postSelectAmpBool is True, this value in dB limits the upper threshold of the amplitude change. 0=no change, +6=double volume, etc.

midiPitchMethod* this tells audioguide1.2 how got calculate a midipitch for each segment. There are several possibilities, detailed below. Note that if any of these methods do not find a result, -1 is returned. This attribute is used when writing midi output files. This

data is also accessible as the descriptor d('MIDIPitch-seg').

'f0-seg' returns the averaged f0 array as the midipitch. Note that it is not power weighted - it is actually the median of f0.

'centroid-seg' returns the power averaged centroid converted into a midipitch.

'filename' looks at the filename of the corpus segment to see if there is a midipitch indicated. Names like A1ShortSeq.wav, Fs1ShortSeq.wav and BP-flatter-f-A#3.wav work.

'composite' first tries to look for a midipitch in the filename. If not found, then tries 'f0-seg'.

transMethod* A string indicating how to transpose segments chosen from this corpus entry. You may chose from:

None yields no transposition.

'single-pitch n' transposes all selected segments from this csf() to midipitch n.

'random n m' transposes this segment randomally between midipitch n and m.

'f0' transposes a selected corpus segment to match the f0-seg of the corresponding target segment.

'f0-chroma' transposes a selected corpus segment to match the f0-seg of the corresponding target segment modulo 12 (i.e., matching its chroma).

transQuantize* Quantization interval for transposition of corpus sounds. 1 will quantize to semitones, 0.5 to quarter tones, 2 to whole tones, etc.

```
csf('piano/', transMethod='f0') # transpose corpus segments to match the target's f0.
csf('piano/', transMethod='f0-chroma', transQuantize=0.5) # transpose corpus segments
 to match the target's f0 mod 12.  Then quantize each resulting pitch to the newest
 quarter of tone.
```

allowRepetition* If False, any of the segments from this corpus entry may only be picked one time. If True there is no restriction.

restrictRepetition* A delay time in seconds where, once chosen, a segment from this corpus entry is invalid to be picked again. The default is 0.1, which the same corpus segment from being selected in quick succession.

```
csf('piano/', allowRepetition=False) # each individual segment found in this directory
  of files may only be deleted one time during concatenation.
```

```
csf('piano/', restrictRepetition=2.5) # Each segment is invalid to be picked if it has
  already been selected in the last 2.5 seconds.
```

restrictOverlaps* An integer specifying how many overlapping samples from this collection may be chosen by the concatenative algorithm at any given moment. So, restrictOverlaps=2 only permits 2 overlapping voices at a time.

restrictInTime* a time in seconds specifying how often a sample from this entry may be selected. – for example restrictInTime=0.5 would permit segments from this collection to be select a maximum of once every 0.5 seconds.

maxPercentTargetSegments* a float as a percentage value from 0-100. This number limits the number of target segments that this corpus entry may be selected. For example, a value of 50 means that this corpus entry is only valid for up to 50% of target segments; after this threshold has been crossed, further selections are not possible. None is the default, which has no effect.

scaleDistance* Scale the resulting distance when executing a multidimensional search using on segments. scaleDistance=2 will make these sounds twice as 'far', and thus less likely to be selected by the search algorithm. scaleDistance=0.25 makes 4 times more likely to be picked.

superimposeRule* This one is a little crazy. Basically, you can specify when this corpus's segments can be chosen based on the number of simultaneously selected samples. You do this by writing a little equation as a 2-item list. superimposeRule=('==', 0) says that this set of corpus segments may only be chosen is this is the first selection for this target segment (sim selection '0'). superimposeRule=('>', 2) say this corpus's segments are only valid to by picked if there are already more than 2 selections for this target segment. I know, right?

7.4 Specifying csf() keywords globally

csf() keywords may be specified globally using the variable CORPUS_GLOBAL_ATTRIBUTES. Note that they are specified in dictionary format rather than object/keyword format.

```
CORPUS = [csf('lachenmann.aiff', scaleDb=-6), csf('piano/', scaleDb=-6, wholeFile=True
 )]
# is equivalent to
CORPUS_GLOBAL_ATTRIBUTES = {'scaleDb': -6}
CORPUS = [csf('lachenmann.aiff'), csf('piano/', wholeFile=True)]
```

8 SEARCH variable and spass() object

The SEARCH variable specifies how audioguide1.2 picks corpus segments to match target segments. The idea here is make a *very* flexible searching structure where the user can create multiple search passes on different descriptor criteria.

The SEARCH variable is written as a list of spass() objects. Each spass() has the following parameters:

```
spass(search_type, descriptor1... descriptorN, percent=None, minratio=None, maxratio=
 None)
```

And the search_type string may be among the following methods:

'closest' Return the best matching segment.

'closest_percent' Return the top *percent* percent of the best matching segments.

'farthest' Return the worst matching segment.

'farthest_percent' Return the worst *percent* percent of segments.

'ratio_limit' Return segments where the ratio of the target descriptor value to the segment's value falls between minratio and maxratio. Only works for averaged descriptors.

Here is the most simple case of a SEARCH variable:

```
SEARCH = [spass('closest', d('centroid'))] # will search all corpus segments and
 select the one with the `closest' centroid to the target segment.
```

Note that the first argument is the type of search performed – in this case, selecting the closest sample. Following the arguments are a list of descriptor objects which specify which descriptors to use:

```
SEARCH = [spass('closest', d('centroid'), d('effDur-seg'))] # will search all corpus
 segments and select the one with the `closest' centroid and effective duration
 compared to the target segment.
```

Ok, great. As you can probably imagine, the first argument, 'closest', tells audioguide1.2 to pick the closest sound. But, there are also other possibilities:

```
SEARCH = [spass('farthest', d('centroid'))] # return the worst matching segment.

SEARCH = [spass('closest_percent', d('centroid'), percent=20)] # return the top 20
 percent best matches.

SEARCH = [spass('farthest_percent', d('centroid'), percent=20)] # return the worst 20
```

```
percent of matches.
```

If you use 'closest_percent' or 'farthest_percent' as the one and only spass object in the SEARCH variable, audioguide1.2 will select a corpus segment randomly among the final candidates. However, you can also chain spass objects together, essentially constructing a hierarchical search algorithm. So, for example, take the following SEARCH variable with two separate phases:

```
SEARCH = [
spass('closest_percent', d('effDur-seg'), percent=20), # take the best 20% of matches
 from the corpus
spass('closest', d('mfccs')), # now find the best matching segment from the 20 percent
  that remains.
]
```

I use the above example a lot when using audioguide1.2. It first matches effDur-seg, the effective duration of the target measured agains't the effective duration of each corpus segment. It retains the 20% closest matches, and throws away the worst 80%. Then, with the remaining 20%, the timbre of the sounds are matched according to mfccs.

```
SEARCH = [spass('ratio_limit', d('centroid-seg'), minratio=0.9, maxratio=1.1)] #
 reduce the number of samples in the corpus such
```

Remember, the order of the spass objects in the SEARCH variable is very important – it is essentially the order of operations.

9 The d() object

Use the d() object for parameterizing a descriptor in an spass() object. The d() object take 1 argument – the name of the desired descriptor – and then several optional keywords arguments, detailed below.

```
d('descriptor name', weight=1, norm=2, normmethod='stddev', distance='euclidean',
 energyWeight=False)
```

descriptor name possible descriptors may be: centroid, chromas, chroma0, chroma1, chroma10, chroma11, chroma2, chroma3, chroma4, chroma5, chroma6, chroma7, chroma8, chroma9, crests, crest0, crest1, crest2, crest3, decrease, effDur-seg, energyenvelope, f0, flatnesses, flatness0, flatness1, flatness2, flatness3, harmoniccentroid, harmonicdecrease, harmonicdeviation, harmonicenergy, harmonickurtosis, harmonicoddevenratio, harmonicrolloff, harmonicskewness, harmonicslope, harmonicspread, harmonictristimulus0, harmonictristimulus1, harmonictristimulus2, harmonicvariation, inharmonicity, kurtosis, loudness, noiseen-

ergy, noisiness, perceptualcentroid, perceptualdecrease, perceptualdeviation, perceptualkurtosis, perceptualoddtoevenratio, perceptualrolloff, perceptualskewness, perceptualslope, perceptualspread, perceptualtristimulus0, perceptualtristimulus1, perceptualtristimulus2, perceptualvariation, power, rolloff, sharpness, skewness, slope, spread, variation, zeroCross. Note that appending '-seg' to most descriptors yields an average.

weight* How to weight this descriptor in relation to other descriptors.

```
SEARCH= [spass('closest', d('centroid', weight=1), d('noisiness', weight=0.5))]
 # centroid is twice as important as noisiness.
```

norm* A value of 2 normalizes the target and corpus data separately. A value of 1 normalizes the target and corpus data together. 2 will yield a better rendering of the target's morphological contour. 1 will remain more faithful to concrete descriptor values. I recommend using 2 by default, only using 1 when dealing with very 'descriptive' descriptors like duration or pitch.

```
SEARCH= [spass('closest', d('centroid'), d('effDur-seg', norm=1))]
```

normmethod* How to normalize data – either 'stddev' or 'minmax'. minmax is more precise, stddev is more forgiving of 'outliers.'

distance* Only valid for time-varying descriptors. How to arithmetically measure distance between target and corpus arrays.

'euclidean' does a simple least squares distance.

'pearson' a pearson correlation measurement.

```
SEARCH= [spass('closest', d('centroid', distance='pearson'))] # uses a pearson
 correlation formula for determining distance between target and corpus centroid
 arrays.
```

energyWeight Only valid for time-varying descriptors. Weight distance calculations with the corpus segments' energy values. The means that softer frames will not affect distance as much as louder frames. Only works if distance='euclidean'.

10 The SUPERIMPOSE variable and si() object

Use the si() object for specifying how corpus segments may be superimposed during concatenation.

```
SUPERIMPOSE = si(minSegment=None, maxSegment=None, minOnset=None, maxOnset=8,
 minOverlap=None, maxOverlap=None, searchOrder='power', calcMethod='mixture',
 peakAlign=False)
```

minSegment The minimum number of corpus segments that must be chosen to match a target segment.

maxSegment The maximum number of corpus segments that must be chosen to match a target segment.

minOnset The minimum number of corpus segments that must be chosen to begin at any single moment in time.

maxOnset The maximum number of corpus segments that must be chosen to begin at any single moment in time.

minOverlap The minimum number of overlapping corpus segments at any single moment in time. Note that an 'overlap' is determined according to an amplitude threshold – see overlapAmpThresh.

maxOverlap The maximum number of overlapping corpus segments at any single moment in time. Note that an 'overlap' is determined according to an amplitude threshold – see overlapAmpThresh.

searchOrder ('power' or 'time') The default is 'time', which indicated to match corpus segments to target segments in the temporal order of the target (i.e., first searched segment is the first segment in time). 'power' indicates to first sort the target segments from loudest to softest, then search for corpus matches.

calcMethod A None/string which denotes how to calculate overlapping corpus sounds. None does nothing – each corpus selection is unaware of previous selections. 'subtract' subtracts the energy of a selected corpus sound from the target's amplitude so that future selections factor in the amplitude of past selections. 'mixture' subtracts the amplitude and then attempts to mix the descriptors of simultaneous sounds together. Note that some descriptors are not algorithmically mixable, such as f0, zeroCross.

11 Other Options

11.1 Descriptor Computation Parameters

DESCRIPTOR_FORCE_ANALYSIS (type=bool, default=False) if True, audioguide1.2 is forced to remake all SDIF analysis, even if previously made.

DESCRIPTOR_WIN_SIZE_SEC (type=float, default=0.04096) the FFT window size of descriptor analysis in seconds. 0.04096 seconds = 512 @ 12.5kHz (the default resample rate).

DESCRIPTOR_HOP_SIZE_SEC (type=float, default=0.01024) the FFT window overlaps of descriptor analysis in seconds. Important, as it effectively sets of temporal resolution of audioguide1.2.

IRCAMDESCRIPTOR_RESAMPLE_RATE (type=int, default=25000) The internal resample rate of the IRCAM analysis binary. Important, as it sets the frequency resolution of spectral sound descriptors.

IRCAMDESCRIPTOR_WINDOW_TYPE (type=string, default='blackman') see ircamdescriptor documentation for details.

IRCAMDESCRIPTOR_NUMB_MFCCS (type=int, default=13) sets the number of MFCCs to make. *Doesn't seem to work* in this release, as 13 is all I can ever seem to get out of the library.

IRCAMDESCRIPTOR_F0_MAX_ANALYSIS_FREQ (type=float, default=5000) see ircamdescriptor documentation for details.

IRCAMDESCRIPTOR_F0_MIN_FREQUENCY (type=float, default=20) minimum possible f0 frequency.

IRCAMDESCRIPTOR_F0_MAX_FREQUENCY (type=float, default=5000) maximum possible f0 frequency.

IRCAMDESCRIPTOR_F0_QUALITY (type=float, default=0.1) see ircamdescriptor documentation for details.

SUPERVP_BIN* (type=string/None, default=None) Optionally specify a path to the supervp analysis binary. Used for target pre-concatenation time stretching.

11.2 Concatenation

ROTATE_VOICES* (type=bool, default=False]) if True, audioguide1.2 will rotate through the list of corpus entries during concatenation. This means that, when selecting corpus segment one, audioguide1.2 will only search sound segments from the first item of the CORPUS variable. Selection 2 will only search the second, and so on. Corpus rotation is modular around the length of the CORPUS variable. If the corpus only has one item, True will have no effect.

VOICE_PATTERN* (type=list, default=[]) if an empty list, this does nothing. However, if the user gives a list of strings, audioguide1.2 will rotate through this list of each concatenative selection and only use corpus segments who's filepath match this string. Matching can use parts of the filename, not necessarily the whole path and it is not case sensitive.

OUTPUT_QUANTIZE_TIME_METHOD (type=string/None, default=None) controls the quantisation of the start times of events selected during concatenation. Note that any quantisation takes place after the application of OUTPUT_TIME_STRETCH and OUTPUT_TIME_ADD, as detailed below. This variable has the following possible settings:

None no quantisation takes place (the default).

'snapToGrid' conform the start times of events to a grid spaced in OUTPUT_QUANTIZE_TIME_INTERVAL second slices.

'medianAggregate' change each event's start time to the median start time of events in slices of OUTPUT_QUANTIZE_TIME_INTERVAL seconds.

OUTPUT_QUANTIZE_TIME_INTERVAL (type=float, default=0.25) defines the temporal interval in seconds for quantisation. If OUTPUT_QUANTIZE_TIME_METHOD = None, this doesn't do anything.

OUTPUT_GAIN_DB* (type=int/None, default=None) adds a uniform gain in dB to all selected corpus units. Affects the subtractive envelope calculations and descriptor mixtures as well as csound rendering.

OUTPUT_TIME_STRETCH* (type=float, default=1.) stretch the temporality of selected units. A value of 2 will stretch all events offsets by a factor of 2.

OUTPUT_TIME_ADD* (type=float, default=0.) offset the start time of selected events by a value in seconds.

RANDOM_SEED (type=int/None, default=None) sets the pseudo-random seed for random unit selection. By default a value of None will use the system's timestamp. Setting an integer will create repeatable random results.

11.3 Concatenation Output Files

For each of the following *_FILEPATH variables, a value of None tells the agConcatenate.py NOT to create an output file. Otherwise a string tells agConcatenate.py to create this output file and also indicates the path of the file to create. Strings may be absolute paths. If a relative path is given, audioguide1.2 will create the file relative to the location o the agConcatenate.py script.

CSOUND_CSD_FILEPATH (type=string/None, default='output/output.csd') creates an output csd file for rendering the resulting concatenation with csound.

CSOUND_RENDER_FILEPATH (type=string/None, default='output/output.aiff') sets the sound output file in the CSOUND_CSD_FILEPATH file. This is the name of csound's output soundfile and will be created at the end of concatenation.

MIDI_FILEPATH (type=string/None, default='output/output.mid') a midi file of the concatenation with pitches chosen according to midiPitchMethod from each corpus entry.

OUTPUT_LABEL_FILEPATH (type=string/None, default='output/outputlabels.txt') Audacity-style labels showing the selected corpus sounds and how they overlap.

LISP_OUTPUT_FILEPATH (type=string/None, default='output/output.lisp.txt') a textfile containing selected corpus events as a lisp-style list.

DATA_FROM_SEGMENTATION_FILEPATH (type=string/None, default=None) This file lists all of the extra data of selected events during concatenation. This data is taken from corpus segmentation files, and includes everything after the startTime and endTime of each segment. This is useful if you want to tag each corpus segment with text based information for

use later.

DICT_OUTPUT_FILEPATH (type=string/None, default='output/output.json') a textfile containing selected corpus events in json format.

MAXMSP_OUTPUT_FILEPATH (type=string/None, default='output/output.maxmsp.json') a textfile containing a list of selected corpus events. Data includes starttime in MS, duration in MS, filename, transposition, amplitude, etc.

11.4 Other Output Files

For each of the following *_FILEPATH variables, a value of None tells the agConcatenate.py NOT to create an output file. Otherwise a string tells agConcatenate.py to create this output file and also indicates the path of the file to create. Strings may be absolute paths. If a relative path is given, audioguide1.2 will create the file relative to the location o the agConcatenate.py script.

LOG_FILEPATH (type=string/None, default='output/log.txt') a log file with lots of information from the concatenation algorithm.

TARGET_SEGMENT_LABELS_FILEPATH (type=string/None, default='output/targetlabels.txt') Audacity-style labels showing how the target sound was segmented.

TARGET_SEGMENTATION_GRAPH_FILEPATH (type=string/None, default=None) like TARGET_SEGMENT_LABELS_FILEPATH, this variable creates a file to show information about target segmentation. Here however, the output is a jpg graph of the onset and offset times and the target's power. This output requires you to install python's module matplotlib.

TARGET_DESCRIPTORS_FILEPATH (type=string/None, default=None) saves the loaded target descriptors to a json dictionary.

TARGET_PLOT_DESCRIPTORS_FILEPATH (type=string/None, default=None) creates a plot of each target descriptor used in concatenation. Doesn't create plots for averaged descriptors ("-seg"), only time varying descriptors.

11.5 Printing/Interaction

SEARCH_PATHS (type=list, default=[]) a list of strings, each of which is a path to a directory where soundfile are located. These paths extend the list of search paths that audioguide1.2 examines when searching for target and corpus soundfiles. The default is an empty list, which doesn't do anything.

VERBOSITY (type=int, default=2) affects the amount of information audioguide1.2 prints to the terminal. A value of 0 yields nothing. A value of 1 prints a minimal amount of information. A value of 2 (the default) prints refreshing progress bars to indicate the progress of the algorithms.

PRINT_SELECTION_HISTO (type=bool, default=False) if True will print robust information about corpus selection after concatenation. If false (the default) will add this information to the log file, if used.

PRINT_SIM_SELECTION_HISTO (type=bool, default=False) if True will print robust information about corpus overlapping selection after concatenation. If false (the default) will add this information to the log file, if used.

11.6 Csound Rendering

CSOUND_SR (type=int, default=48000) The sample rate used for csound rendering. Csound will interpolate the sample rates of all corpus files to this rate. It will be the sr of csound's output soundfile.

CSOUND_KSMPS (type=int, default=128) The ksmps value used for csound rendering. See csound's documentation for more information.

CSOUND_CHANNEL_RENDER_METHOD (type=string, default='mix') Tells audioguide1.2 how deal with corpus segments distribution in the output soundfile By default''mix" creates a 2 channel csound file and puts mono corpus sounds in the middle of the stereo field. The string "oneChannelPerVoice" tells audioguide to put selected sounds from each item of the CORPUS list into a separate channel. The number of output channels will therefore equal the length of the CORPUS list variable.

CSOUND_STRETCH_CORPUS_TO_TARGET_DUR (type=string/None, default=None) Affects the durations of concatenated sound events rendered by csound. By default None doesn't do anything – csound plays back each corpus sound according to its duration. "pv" uses a phase vocoder to stretch corpus sounds to match the duration of the corresponding target segment. "transpose" does the same, but using the speed of playback to change duration rather than a phase vocoder. Note that, in this case, any other transposition information generated by the selection algorithm is overwritten.

CSOUND_PLAY_RENDERED_FILE (type=bool, default=True) if True, audioguide1.2 will play the rendered csound file at the command line at the end of the concatenative algorithm.

MIDIFILE_TEMPO (type=int/float, default=60) sets the tempo of the midi file output.