AudioSegment Documentation

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Feb 02, 2019

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CHAPTER 1

algorithms package

1.1 Submodules

1.2 algorithms.asa module

This module extracts out a bunch of the Auditory Scene Analysis (ASA) logic, which has grown to be a little unwieldy in the AudioSegment class.

`algorithms.asa.`**`visualize`**(*spect*, *frequencies*, *title=''*)

`algorithms.asa.`**`visualize_fronts`**(*onsets*, *offsets*, *spect*, *frequencies*)

`algorithms.asa.`**`visualize_peaks_and_valleys`**(*peaks*, *valleys*, *spect*, *frequencies*)

`algorithms.asa.`**`visualize_segmentation_mask`**(*segmentation*, *spect*, *frequencies*, *mode='new'*)

`algorithms.asa.`**`visualize_time_domain`**(*seg*, *title=''*)

1.3 algorithms.eventdetection module

This module contains a bunch of functions that are integral to the auditory event detection algorithm used by AudioSegment. We refactored them to here because they aren't really useful on their own, and they take up brainspace by being in the AudioSegment class.

1.4 algorithms.filters module

Convenience functions for using Numpy/Scipy filters in the audio domain.

`algorithms.filters.`**`bandpass_filter`**(*data*, *low*, *high*, *fs*, *order=5*)

Does a bandpass filter over the given data.

Parameters

- **data** – The data (numpy array) to be filtered.
- **low** – The low cutoff in Hz.
- **high** – The high cutoff in Hz.
- **fs** – The sample rate (in Hz) of the data.
- **order** – The order of the filter. The higher the order, the tighter the roll-off.

Returns Filtered data (numpy array).

algorithms.filters.**lowpass_filter**(*data*, *cutoff*, *fs*, *order=5*)
Does a lowpass filter over the given data.

Parameters

- **data** – The data (numpy array) to be filtered.
- **cutoff** – The high cutoff in Hz.
- **fs** – The sample rate in Hz of the data.
- **order** – The order of the filter. The higher the order, the tighter the roll-off.

Returns Filtered data (numpy array).

1.5 algorithms.util module

Utility module for miscellaneous stuff

algorithms.util.**isclose**(*a*, *b*, *, *rel_tol=1e-09*, *abs_tol=0.0*)
Python 3.4 does not have math.isclose, so we need to steal it and add it here.

1.6 Module contents

CHAPTER 2

audiosegment module

This module simply exposes a wrapper of a pydub.AudioSegment object.

class audiosegment.**AudioSegment**(*pydubseg*, *name*)

Bases: `object`

This class is a wrapper for a pydub.AudioSegment that provides additional methods.

auditory_scene_analysis(*debug=False*, *debugplot=False*)

Algorithm based on paper: Auditory Segmentation Based on Onset and Offset Analysis, by Hu and Wang, 2007.

Returns a list of AudioSegments, each of which is all the sound during this AudioSegment's duration from a particular source. That is, if there are several overlapping sounds in this AudioSegment, this method will return one AudioSegment object for each of those sounds. At least, that's the idea.

Current version is very much in alpha, and while it shows promise, will require quite a bit more tuning before it can really claim to work.

Parameters

- **debug** – If *True* will print out debug outputs along the way. Useful if you want to see why it is taking so long.
- **debugplot** – If *True* will use Matplotlib to plot the resulting spectrogram masks in Mel frequency scale.

Returns List of AudioSegment objects, each of which is from a particular sound source.

detect_event(*model*, *ms_per_input*, *transition_matrix*, *model_stats*, *event_length_s*, *start_as_yes=False*, *prob_raw_yes=0.5*)

A list of tuples of the form [('n', AudioSegment), ('y', AudioSegment), etc.] is returned, where tuples of the form ('n', AudioSegment) are the segments of sound where the event was not detected, while ('y', AudioSegment) tuples were the segments of sound where the event was detected.

```
# Example usage
import audiosegment
import keras
```

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```
import keras.models
import numpy as np
import sys

class Model:
    def __init__(self, modelpath):
        self.model = keras.models.load_model(modelpath)

    def predict(self, seg):
        _bins, fft_vals = seg.fft()
        fft_vals = np.abs(fft_vals) / len(fft_vals)
        predicted_np_form = self.model.predict(np.array([fft_vals]), batch_
↪size=1)
        prediction_as_int = int(round(predicted_np_form[0][0]))
        return prediction_as_int

modelpath = sys.argv[1]
wavpath = sys.argv[2]
model = Model(modelpath)
seg = audiosegment.from_file(wavpath).resample(sample_rate_Hz=32000, sample_
↪width=2, channels=1)
pyes_to_no = 0.3  # The probability of one 30 ms sample being an event, and␣
↪the next one not
pno_to_yes = 0.2  # The probability of one 30 ms sample not being an event,␣
↪and the next one yes
ptrue_pos_rate = 0.8  # The true positive rate (probability of a predicted␣
↪yes being right)
pfalse_neg_rate = 0.3  # The false negative rate (probability of a predicted␣
↪no being wrong)
raw_prob = 0.7  # The raw probability of seeing the event in any random 30 ms␣
↪slice of this file
events = seg.detect_event(model, ms_per_input=30, transition_matrix=[pyes_to_
↪no, pno_to_yes],
                          model_stats=[ptrue_pos_rate, pfalse_neg_rate],␣
↪event_length_s=0.25,
                          prob_raw_yes=raw_prob)
nos = [event[1] for event in events if event[0] == 'n']
yeses = [event[1] for event in events if event[0] == 'y']
if len(nos) > 1:
    notdetected = nos[0].reduce(nos[1:])
    notdetected.export("notdetected.wav", format="WAV")
if len(yeses) > 1:
    detected = yeses[0].reduce(yeses[1:])
    detected.export("detected.wav", format="WAV")
```

Parameters

- **model** – The model. The model must have a predict() function which takes an AudioSegment of *ms_per_input* number of ms and which outputs 1 if the audio event is detected in that input, and 0 if not. Make sure to resample the AudioSegment to the right values before calling this function on it.
- **ms_per_input** – The number of ms of AudioSegment to be fed into the model at a time. If this does not come out even, the last AudioSegment will be zero-padded.
- **transition_matrix** – An iterable of the form: [p(yes->no), p(no->yes)]. That is, the probability of moving from a 'yes' state to a 'no' state and the probability of vice versa.

- **model_stats** – An iterable of the form: [p(reality=1|output=1), p(reality=1|output=0)]. That is, the probability of the ground truth really being a 1, given that the model output a 1, and the probability of the ground truth being a 1, given that the model output a 0.
- **event_length_s** – The typical duration of the event you are looking for in seconds (can be a float).
- **start_as_yes** – If True, the first *ms_per_input* will be in the 'y' category. Otherwise it will be in the 'n' category.
- **prob_raw_yes** – The raw probability of finding the event in any given *ms_per_input* vector.

Returns A list of tuples of the form [('n', AudioSegment), ('y', AudioSegment), etc.], where over the course of the list, the AudioSegment in tuple 3 picks up where the one in tuple 2 left off.

Raises ValueError if *ms_per_input* is negative or larger than the number of ms in this AudioSegment; if *transition_matrix* or *model_stats* do not have a __len__ attribute or are not length 2; if the values in *transition_matrix* or *model_stats* are not in the closed interval [0.0, 1.0].

detect_voice(*prob_detect_voice=0.5*)

Returns self as a list of tuples: [('v', voiced segment), ('u', unvoiced segment), (etc.)]

The overall order of the AudioSegment is preserved.

Parameters **prob_detect_voice** – The raw probability that any random 20ms window of the audio file contains voice.

Returns The described list.

dice(*seconds*, *zero_pad=False*)

Cuts the AudioSegment into *seconds* segments (at most). So for example, if seconds=10, this will return a list of AudioSegments, in order, where each one is at most 10 seconds long. If *zero_pad* is True, the last item AudioSegment object will be zero padded to result in *seconds* seconds.

Parameters

- **seconds** – The length of each segment in seconds. Can be either a float/int, in which case *self.duration_seconds* / *seconds* are made, each of *seconds* length, or a list-like can be given, in which case the given list must sum to *self.duration_seconds* and each segment is specified by the list - e.g. the 9th AudioSegment in the returned list will be *seconds[8]* seconds long.
- **zero_pad** – Whether to zero_pad the final segment if necessary. Ignored if *seconds* is a list-like.

Returns A list of AudioSegments, each of which is the appropriate number of seconds long.

Raises ValueError if a list-like is given for *seconds* and the list's durations do not sum to *self.duration_seconds*.

fft(*start_s=None*, *duration_s=None*, *start_sample=None*, *num_samples=None*, *zero_pad=False*)

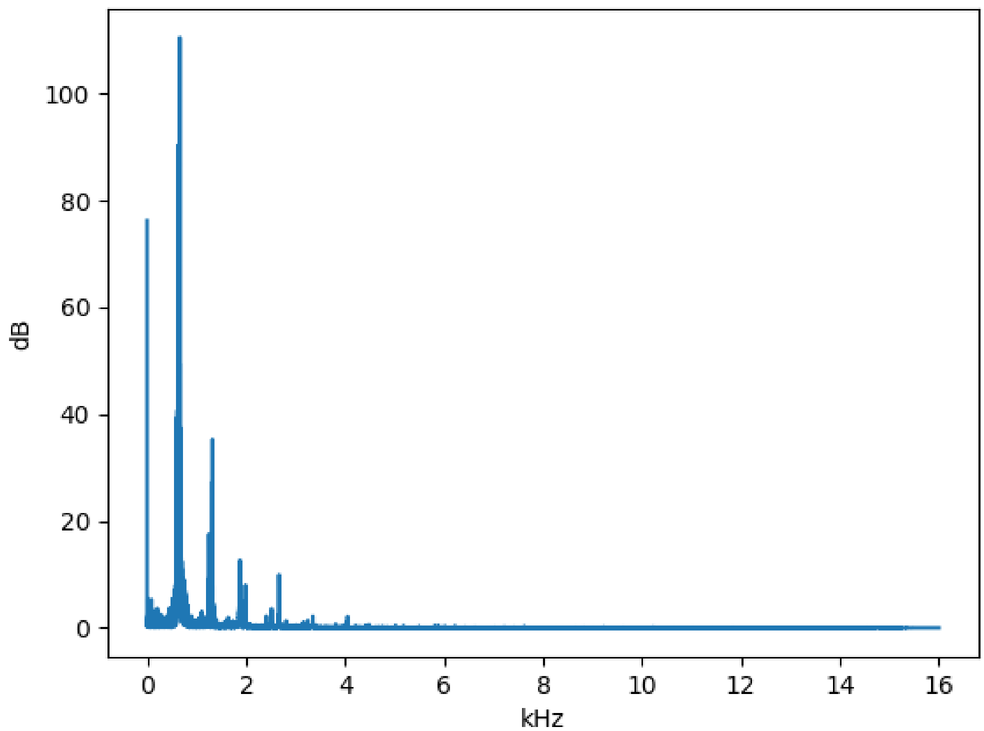
Transforms the indicated slice of the AudioSegment into the frequency domain and returns the bins and the values.

If neither *start_s* or *start_sample* is specified, the first sample of the slice will be the first sample of the AudioSegment.

If neither *duration_s* or *num_samples* is specified, the slice will be from the specified start to the end of the segment.

```
# Example for plotting the FFT using this function
import matplotlib.pyplot as plt
import numpy as np

seg = audiosegment.from_file("furelise.wav")
# Just take the first 3 seconds
hist_bins, hist_vals = seg[1:3000].fft()
hist_vals_real_normed = np.abs(hist_vals) / len(hist_vals)
plt.plot(hist_bins / 1000, hist_vals_real_normed)
plt.xlabel("kHz")
plt.ylabel("dB")
plt.show()
```



Parameters

- **start_s** – The start time in seconds. If this is specified, you cannot specify *start_sample*.
- **duration_s** – The duration of the slice in seconds. If this is specified, you cannot specify *num_samples*.
- **start_sample** – The zero-based index of the first sample to include in the slice. If this is specified, you cannot specify *start_s*.
- **num_samples** – The number of samples to include in the slice. If this is specified, you cannot specify *duration_s*.
- **zero_pad** – If True and the combination of start and duration result in running off the end of the AudioSegment, the end is zero padded to prevent this.

Returns np.ndarray of frequencies, np.ndarray of amount of each frequency

Raises ValueError If *start_s* and *start_sample* are both specified and/or if both *duration_s* and *num_samples* are specified.

filter_bank (*lower_bound_hz=50*, *upper_bound_hz=8000.0*, *nfilters=128*, *mode='mel'*)

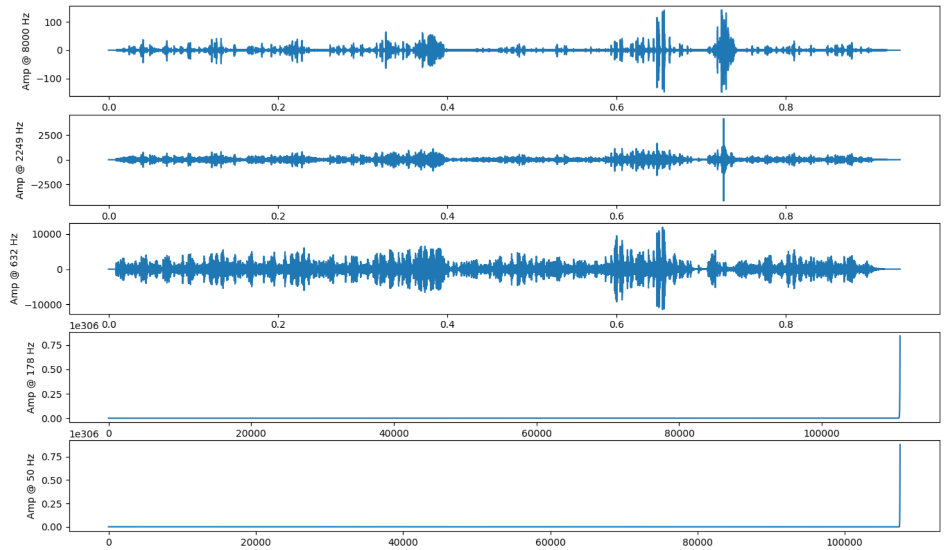
Returns a numpy array of shape (nfilters, nsamples), where each row of data is the result of bandpass filtering the audiosegment around a particular frequency. The frequencies are spaced from *lower_bound_hz* to *upper_bound_hz* and are returned with the np array. The particular spacing of the frequencies depends on *mode*, which can be either: 'linear', 'mel', or 'log'.

Note: This method is an approximation of a gammatone filterbank until I get around to writing an actual gammatone filterbank function.

```
# Example usage
import audiosegment
import matplotlib.pyplot as plt
import numpy as np

def visualize(spect, frequencies, title=""):
    # Visualize the result of calling seg.filter_bank() for any number of
↪filters
    i = 0
    for freq, (index, row) in zip(frequencies[::-1], enumerate(spect[::-1,
↪:])):
        plt.subplot(spect.shape[0], 1, index + 1)
        if i == 0:
            plt.title(title)
            i += 1
        plt.ylabel("{0:.0f}".format(freq))
        plt.plot(row)
    plt.show()

seg = audiosegment.from_file("some_audio.wav").resample(sample_rate_Hz=24000,
↪sample_width=2, channels=1)
spec, frequencies = seg.filter_bank(nfilters=5)
visualize(spec, frequencies)
```



Parameters

- **lower_bound_hz** – The lower bound of the frequencies to use in the bandpass filters.
- **upper_bound_hz** – The upper bound of the frequencies to use in the bandpass filters.
- **nfilters** – The number of filters to apply. This will determine which frequencies are used as well, as they are interpolated between *lower_bound_hz* and *upper_bound_hz* based on *mode*.
- **mode** – The way the frequencies are spaced. Options are: *linear*, in which case the frequencies are linearly interpolated between *lower_bound_hz* and *upper_bound_hz*, *mel*, in which case the mel frequencies are used, or *log*, in which case they are log-10 spaced.

Returns A numpy array of the form (nfilters, nsamples), where each row is the audiosegment, bandpass-filtered around a particular frequency, and the list of frequencies. I.e., returns (spec, freqs).

filter_silence(*duration_s=1*, *threshold_percentage=1*, *console_output=False*)

Returns a copy of this AudioSegment, but whose silence has been removed.

Note: This method requires that you have the program 'sox' installed.

Warning: This method uses the program 'sox' to perform the task. While this is very fast for a single function call, the IO may add up for large numbers of AudioSegment objects.

Parameters

- **duration_s** – The number of seconds of "silence" that must be present in a row to be stripped.
- **threshold_percentage** – Silence is defined as any samples whose absolute value is below *threshold_percentage * max(abs(samples in this segment))*.

- **console_output** – If True, will pipe all sox output to the console.

Returns A copy of this AudioSegment, but whose silence has been removed.

generate_frames (*frame_duration_ms*, *zero_pad=True*)

Yields self's data in chunks of frame_duration_ms.

This function adapted from pywebrtc's example [https://github.com/wiseman/py-webrtcvad/blob/master/example.py].

Parameters

- **frame_duration_ms** – The length of each frame in ms.
- **zero_pad** – Whether or not to zero pad the end of the AudioSegment object to get all the audio data out as frames. If not, there may be a part at the end of the Segment that is cut off (the part will be <= *frame_duration_ms* in length).

Returns A Frame object with properties 'bytes (the data)', 'timestamp (start time)', and 'duration'.

generate_frames_as_segments (*frame_duration_ms*, *zero_pad=True*)

Does the same thing as *generate_frames*, but yields tuples of (AudioSegment, timestamp) instead of Frames.

normalize_spl_by_average (*db*)

Normalize the values in the AudioSegment so that its *spl* property gives *db*.

Note: This method is currently broken - it returns an AudioSegment whose values are much smaller than reasonable, yet which yield an SPL value that equals the given *db*. Such an AudioSegment will not be serializable as a WAV file, which will also break any method that relies on SOX. I may remove this method in the future, since the SPL of an AudioSegment is pretty questionable to begin with.

Parameters **db** – The decibels to normalize average to.

Returns A new AudioSegment object whose values are changed so that their average is *db*.

Raises ValueError if there are no samples in this AudioSegment.

reduce (*others*)

Reduces others into this one by concatenating all the others onto this one and returning the result. Does not modify self, instead, makes a copy and returns that.

Parameters **others** – The other AudioSegment objects to append to this one.

Returns The concatenated result.

resample (*sample_rate_Hz=None*, *sample_width=None*, *channels=None*, *console_output=False*)

Returns a new AudioSegment whose data is the same as this one, but which has been resampled to the specified characteristics. Any parameter left None will be unchanged.

Note: This method requires that you have the program 'sox' installed.

Warning: This method uses the program 'sox' to perform the task. While this is very fast for a single function call, the IO may add up for large numbers of AudioSegment objects.

Parameters

- **sample_rate_Hz** – The new sample rate in Hz.
- **sample_width** – The new sample width in bytes, so sample_width=2 would correspond to 16 bit (2 byte) width.
- **channels** – The new number of channels.
- **console_output** – Will print the output of sox to the console if True.

Returns The newly sampled AudioSegment.

serialize()

Serializes into a bytestring.

Returns An object of type Bytes.

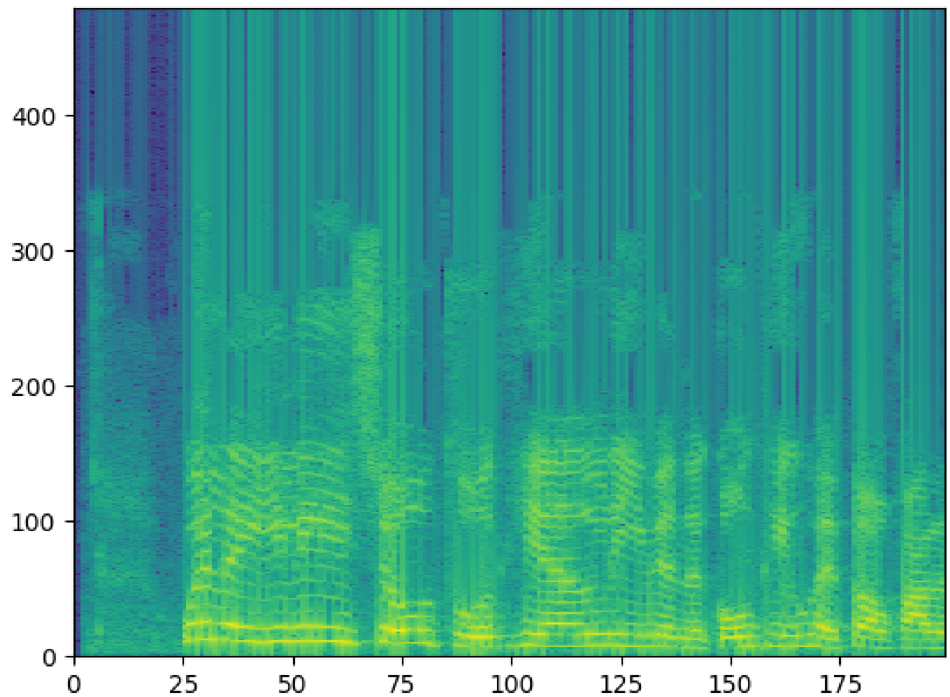
spectrogram(*start_s=None, duration_s=None, start_sample=None, num_samples=None, window_length_s=None, window_length_samples=None, overlap=0.5*)

Does a series of FFTs from *start_s* or *start_sample* for *duration_s* or *num_samples*. Effectively, transforms a slice of the AudioSegment into the frequency domain across different time bins.

```
# Example for plotting a spectrogram using this function
import audiosegment
import matplotlib.pyplot as plt

#...
seg = audiosegment.from_file("somebodytalking.wav")
freqs, times, amplitudes = seg.spectrogram(window_length_s=0.03, overlap=0.5)
amplitudes = 10 * np.log10(amplitudes + 1e-9)

# Plot
plt.pcolormesh(times, freqs, amplitudes)
plt.xlabel("Time in Seconds")
plt.ylabel("Frequency in Hz")
plt.show()
```



Parameters

- **start_s** – The start time. Starts at the beginning if neither this nor *start_sample* is specified.
- **duration_s** – The duration of the spectrogram in seconds. Goes to the end if neither this nor *num_samples* is specified.
- **start_sample** – The index of the first sample to use. Starts at the beginning if neither this nor *start_s* is specified.
- **num_samples** – The number of samples in the spectrogram. Goes to the end if neither this nor *duration_s* is specified.
- **window_length_s** – The length of each FFT in seconds. If the total number of samples in the spectrogram is not a multiple of the window length in samples, the last window will be zero-padded.
- **window_length_samples** – The length of each FFT in number of samples. If the total number of samples in the spectrogram is not a multiple of the window length in samples, the last window will be zero-padded.
- **overlap** – The fraction of each window to overlap.

Returns Three np.ndarrays: The frequency values in Hz (the y-axis in a spectrogram), the time values starting at start time and then increasing by *duration_s* each step (the x-axis in a spectrogram), and the dB of each time/frequency bin as a 2D array of shape [len(frequency values), len(duration)].

Raises **ValueError** – If *start_s* and *start_sample* are both specified, if *duration_s* and *num_samples* are both specified, if the first window's duration plus start time lead to running off the end of the AudioSegment, or if *window_length_s* and *window_length_samples* are either both specified or if they are both not specified.

spl

Sound Pressure Level - defined as 20 * log10(p/p0), where p is the RMS of the sound wave in Pascals and p0 is 20 micro Pascals.

Since we would need to know calibration information about the microphone used to record the sound in order to transform the PCM values of this audiosegment into Pascals, we can't really give an accurate SPL measurement.

However, we can give a reasonable guess that can certainly be used to compare two sounds taken from the same microphone set up.

Be wary about using this to compare sounds taken under different recording conditions however, except as a simple approximation.

Returns a scalar float representing the dB SPL of this audiosegment.

to_numpy_array()

Convenience function for *np.array(self.get_array_of_samples())* while keeping the appropriate dtype.

zero_extend(*duration_s=None*, *num_samples=None*)

Adds a number of zeros (digital silence) to the AudioSegment (returning a new one).

Parameters

- **duration_s** – The number of seconds of zeros to add. If this is specified, *num_samples* must be None.
- **num_samples** – The number of zeros to add. If this is specified, *duration_s* must be None.

Returns A new AudioSegment object that has been zero extended.

Raises ValueError if duration_s and num_samples are both specified.

audiosegment.**deprecated**(*func*)

Deprecator decorator.

audiosegment.**deserialize**(*bstr*)

Attempts to deserialize a bytestring into an audiosegment.

Parameters **bstr** – The bytestring serialized via an audiosegment's serialize() method.

Returns An AudioSegment object deserialized from *bstr*.

audiosegment.**empty**()

Creates a zero-duration AudioSegment object.

Returns An empty AudioSegment object.

audiosegment.**from_file**(*path*)

Returns an AudioSegment object from the given file based on its file extension. If the extension is wrong, this will throw some sort of error.

Parameters **path** – The path to the file, including the file extension.

Returns An AudioSegment instance from the file.

audiosegment.**from_mono_audiosegments**(**args*)

Creates a multi-channel AudioSegment out of multiple mono AudioSegments (two or more). Each mono AudioSegment passed in should be exactly the same number of samples.

Returns An AudioSegment of multiple channels formed from the given mono AudioSegments.

`audiosegment.`**`from_numpy_array`**`(`*nparr*, *framerate*`)`

Returns an AudioSegment created from the given numpy array.

The numpy array must have shape = (num_samples, num_channels).

Parameters **`nparr`** – The numpy array to create an AudioSegment from.

Returns An AudioSegment created from the given array.

`audiosegment.`**`silent`**`(`*duration=1000*, *frame_rate=11025*`)`

Creates an AudioSegment object of the specified duration/frame_rate filled with digital silence.

Parameters

- **`duration`** – The duration of the returned object in ms.
- **`frame_rate`** – The samples per second of the returned object.

Returns AudioSegment object filled with pure digital silence.

CHAPTER 3

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