DiViMe Documentation

Release 1.0

ACLEW

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CHAPTER 1

Before starting

1.1 What is the ACLEW DiViMe?

It is a collection of speech processing tools allowing users to automatically add annotations onto a raw audio recording. At present, we have tools to do the following types of annotation:

1. Speech activity detection (*when is someone talking?*)
2. Talker diarization (*who is talking?*)
3. Role diarization (*what kind of person is talking?*)
4. Vocal type classification (*what kind of vocalization is this one?*)

We are hoping to add more tools in the future, including register detection, and syllable quantification.

1.2 Who is the ACLEW DiViMe for?

ACLEW DiViMe is for researchers dealing with speech recorded in naturalistic environments (speech in the wild), typically in daylong recording settings. The population recorded may be sensitive, therefore researchers may not be able to share their audio recordings. Our primary test case involves language acquisition in children 0-3 years of age.

We are trying to make the use of these tools as easy as possible, but some command line programming/scripting is unavoidable. If you are worried when reading this, we can recommend the Software Carpentry programming courses for researchers, and particularly their unix bash and version control bootcamps.

1.3 What exactly is inside the ACLEW DiViMe?

A virtual machine (VM) is actually a mini-computer that gets set up inside your computer. This creates a virtual environment within which we can be sure that our tools run, and run in the same way across all computers (Windows, Mac, Linux).

Inside this mini-computer, we have tried to put several tools for each one of our three questions. Please note that some of the tools are developed by fellow researchers and programmers, and since we do not control them, we cannot be absolutely certain they will work. Therefore, we provide a general introduction to the contents in the usage section, and a specific list of tools in dedicated Detailed instructions sections.

1.4 How should I cite ACLEW DiViMe?

The main citation is this paper, which explains the structure and idea, and provides some evaluation:

Adrien Le Franc, Eric Riebling, Julien Karadayi, Yun Wang, Camila Scaff, Florian Metze, and Alejandrina Cristia. The ACLEW DiViMe: An easy-to-use diarization tool. In Proc. INTERSPEECH, Hyderabad; India, September 2018.

The idea of using virtual machines to package speech tools comes from this work:

Florian Metze, Eric Fosler-Lussier, and Rebecca Bates. The speech recognition virtual kitchen. In Proc. INTERSPEECH, Lyon; France, August 2013. https://github.org/srvk.

Depending on the particular tool that you are using, you should potentially cite additional papers that describe the underlying software or methods - please check.

CHAPTER 2

Installing DiViMe

2.1 Requirements

DiViMe can be installed in any operating system and computer with at least 2 CPUs, 8GB of RAM, and 25GB of available disc space. You may need more of everything to actually use the tools, specifically when running on large files. *Before following the instructions under "First Installation"*, you must follow the instructions in the relevant subsection of the Troubleshooting section, at the end of this page, in the following cases:

- your computer has only one core (or you don't know)
- your computer has 25 GB or less of disc space
- your computer has 6 GB or less of RAM
- your computer is running Ubuntu (e.g., 16.04)

2.2 First Installation

1. Install Vagrant: Click on the download link for your operating system and follow the prompted instructions
2. Install VirtualBox: When we last checked, the links for download for all operating systems were under the header "VirtualBox 5.2.18 platform packages", so look for a title like that one (picking the latest version, most likely).
3. Clone the present repository: To do this, you must use a terminal. If you don't know what this means, we recommend that you first follow the Software Carpentry Shell Tutorial (up to 00:45, namely "Introducing the shell", and "Navigating files and directories"). Next, navigate to the directory in which you want the VM to be hosted and type in: `$ git clone https://github.com/srvk/DiViMe` - or use another way or tool to "clone" a Git repository
4. Change into this folder: `$ cd DiViMe`
5. Install several Vagrant plugins: `$ vagrant plugin install vagrant-aws vagrant-sshfs vagrant-vbguest`

Depending on how you want to use DiViMe, and which provider you will be using, you may or may not need to install the above plugins (or you may need to install additional plugins), but this quickly turns into an advanced topic, because not all plugins will work equally well on all host platforms ...

1. Type `$ vagrant up`

The first time you do this, it will take at least 15 minutes to install all the packages that are needed to build the virtual machine. You are done when you see something like this:

```
FloriansMBP2019:DiViMe metze$ tail -n 8 example-logs/vagrant-up.log
    default: build succeeded, 4 warnings.
    default: The HTML pages are in build/html.
    default: INFO: You can remove Anaconda2-2019.03-Linux-x86_64.sh, if you don't␣
↪plan on re-provisioning DiViMe any time soon.
    default: INFO: You can remove MCR_R2017b_glnxa64_installer.zip, if you don't plan␣
↪on re-provisioning DiViMe any time soon.
    default: ---- Done bootstrapping DiViMe @ Wed Jul  3 03:30:01 UTC 2019 ----
    default: root@vagrant-ubuntu-trusty-64:/home/vagrant#
    default: exit
```

The instructions above make the simplest assumptions as to your environment. If you have Amazon Web Services, an Ubuntu system, or you do not have admin rights in your computer, you might need to read the instructions to the eesen-transcriber for fancier options. Or you can just open an issue here, describing your situation.

We are working on Installing With Docker, but this option is not yet fully functional.

Please note that there is a large amount of documentation on Vagrant and Virtualbox online, explaining on how to fix assorted errors. It is often a good idea to simply "google" errors that these tools throw; you may find a solution to your specific problem quickly, because the problem may not lie with DiViMe, but the VM, before DiViMe gets involved.

2.3 Checking your installation

The very first time you use DiViMe, it is a good idea to run a quick start test, which will be performed using data from the VanDam Public Daylong HomeBank corpus (VanDam et al., 2015):

1. Open a terminal
2. Navigate inside the DiViMe folder
3. Do `$ vagrant up` (if you haven't done it already)
4. Do `$ vagrant ssh -c "launcher/test.sh"`

This should produce the following output:

```
FloriansMBP2019:DiViMe metze$ vagrant ssh -c "launcher/test.sh"
Starting tests
Checking for HTK...
   HTK missing. You can probably ignore this warning, HTK is no longer needed.
Testing noisemes...
Noisemes passed the test.
Testing OpenSmile SAD...
OpenSmile SAD passed the test.
[...]
Congratulations, everything is OK!
[...]
```

2.4 When you are done with DiViMe, Teardown

After working with DiViMe, you can shut down the virtual machine, which will free up CPU and RAM resources on your computer (but not disc space). To do this, type `$ vagrant halt` or `$ vagrant suspend`. To continue working with the VM at a later point, simply issue another `$ vagrant up` command.

2.5 Updating DiViMe

If you want to install a new release of DiViMe, you will need to perform the following 3 steps from within the DiViME folder on your terminal:

```
$ vagrant destroy
$ git pull
$ vagrant up
```

2.6 Uninstallation

If you want to get rid of the files completely, you should perform the following 3 steps from within the DiViME folder on your terminal (assuming you are on Unix):

```
$ vagrant destroy
$ cd ..
$ rm -rf DiViMe
```

2.7 Troubleshooting

2.7.1 If your computer only has one core

Before doing `vagrant up`, open the file called `DiViMe/Vagrantfile` in a text editor. Change the following line:

```
> vbox.cpus = 2
```

into:

```
> vbox.cpus = 1
```

Then proceed with the installation. Also, if you have more than one CPU, and you do not want DiViMe to take over your entire computer, you can set it to any value >= 2, and you should be fine. DiViMe uses multiple processors, but we have not yet fully optimized for many cores.

2.7.2 If your computer has 25 GB or less of storage space

If your computer has less than 25 GB of storage space, then *you cannot build a fully working DiViMe*. In this case, clean up your files to free up space.

2.7.3 If your computer has 6 GB or less of RAM

If your computer has less than about 8 GB of RAM, then you may or may not be able to build and use DiViMe. You probably need to change the space allocated to the virtual machine. Before doing `vagrant up`, open the file called `DiViMe/Vagrantfile` in a text editor. Change the following line:

```
> vbox.memory = 4096
```

into:

```
> vbox.memory = 2048
```

Then proceed with the Installation. Also, if you have more RAM, and you experience issues during installation (or use), you may benefit from increasing this value, which should normally not exceed half of your total installed RAM (as a rule of thumb).

2.7.4 If your computer is running Ubuntu (16.04)

There is a known incompatibility between VirtualBox and the 4.13 Linux kernel on Ubuntu 16.04. What you may do is to install a previous version of the kernel, for example the 4.10, following these instructions, or install the latest version of VirtualBox, which should fix the problem.

Again, there is often a lot of information available online, because Vagrant and VirtualBox are widely used tools.

2.7.5 If something else fails

Please open an issue here. Please paste the complete output of the failing run there, so we can better provide you with a solution. Also, please provide detailed information on your host system (which OS, RAM, CPU, HDD), which changes you made to the Vagrantfile, and also provide access to the data the system chokes on (if any).

2.7.6 References

VanDam, M., De Palma, P., Strong, W. E. (2015, May). Fundamental frequency of speech directed to children who have hearing loss. Poster presented at the 169th Meeting of the Acoustical Society of America, Pittsburgh, PA.

CHAPTER 3

Using DiViMe

3.1 Overview

This is an overview of the full tool presentation found in the next sextion, recapping the main steps:

1. Put your data in the `data` shared directory.
2. Do `$ vagrant up` to "wake the machine up"

Next we provide instructions for all tools. More detailed information about each tool can be found in separate ReadMe files.

Assuming the installation of the virtual machine is complete and some of the tests have passed, you can now use at least some of the tools in the virtual machine. We explain more about each step below, but in a nutshell, the steps to use DiViMe are always the same:

1. Put the data you want to process in the `data/` directory (or any subdirectory within `data/`)
2. Go to the DiViMe folder
3. Do `$ vagrant up` to "wake the machine up"
4. Use tools on data, typically by doing `vagrant ssh -c "script.sh [arguments]"`. You can also run a recipe.
5. Finally, remember to put DiViMe back to sleep with `$ vagrant halt`

3.2 Further information on Step 1, putting your data into the `data/` directory

Put the sound files that you want analyzed (and annotations, if you have any) inside the shared `data` folder. It is probably safer to make a copy of your files (rather than moving them), in case you later decide to delete the whole folder. Also, for greater security, DiViMe (as a VM) can only see data within the `DiViMe` folder, so soft links to files outside of that fodler will not work.

You can drop a whole folder into `data`. You will provide the path to the specific folder to be analyzed when running the tools (as per instructions below). All `.wav` files in that folder will be analyzed.

If your files aren't .wav some of the tools may not work. Please consider converting them into wav with some other program, such as ffmpeg. It is probably safer to make a copy (rather than moving your files into the data folder), in case you later decide to delete the whole folder.

If you have any annotations, put them also in the same `data` folder. Annotations must be in .rttm format, and *they should be named exactly as your wav files*. If you have annotations in .cha, .eaf, .textgrid, or .its, see the Format section for instructions on converting them into .rttm.

IMPORTANT: If you already analyzed a file with a given tool, re-running the tool will result in the previous analysis being overwritten.

3.3 Further information on Step 2, going to the DiViMe folder

To interact with the virtual machine, you must use a terminal. If you don't know what this means, we recommend that you first follow the Software Carpentry Shell Tutorial (up to 00:45, namely "Introducing the shell", and "Navigating files and directories").

Next, navigate in the terminal window to the DiViMe directory that was created when you did `git clone https://github.com/srvk/DiViMe` when installing DiViMe.

3.4 Further information on Step 3, Waking the machine up

Remember that you will be using a mini-computer within your computer. Typically, the machine will be down - i.e., it will not be running. This is good, because when it is running, it will use memory and other resources from your computer (which we call "the host", because it is hosting the other computer). With this step, you launch the virtual machine:

```
$ vagrant up
```

3.5 Further information on Step 4, Using tools on data

3.5.1 Overview of tools

If all tools passed the test, then you'll be able to automatically add the following types of annotation to your audio files:

1. Speech activity detection (*when is someone talking?*): The tools available for this task are the following: noisemesSad, tocomboSad, opensmileSad
2. Talker diarization (*who is talking?*) The tools available for this task are the following: diartk
3. Role diarization (*what kind of person is talking?*) The tools available for this task are the following: yunitator
4. Vocal type classification (*what kind of vocalization is this one?*) The tools available for this task are the following: vcm
5. Evaluation (*how good is the automatic annotation?*) There is an evaluation available for the following tools: noisemesSad, tocomboSad, opensmileSad, diartk, yunitator

3.5.2 The concept of "pipelines"

DiViMe is a platform for tools to analyze naturalistic, unannotated audio recordings. We consider this process to involve three kinds of processes:

- speech activity detection and voice activity detection = "detecting vocalizations",
- diarization = "deciding to whom the vocalizations belong", and
- "additional annotations"

Some tools actually combine two of these stages (e.g. a tool may do both speech activity detection and role attribution in one fell swoop). This flowchart may help.

We call a *pipeline* a sequence of those processes; i.e., it involves using one tool after another. For example, you may do *speech activity detection* + *talker diarization* + *vocal type classification*

Starting from an audio file with no annotation, typically, you may want to run a *speech activity detection* tool followed by a *talker diarization* tool; then you will end up with an annotation showing who spoke when. However, you may not know who "talker0" and "talker1" are. (You could decide this by listening to some samples of each, and mapping them to different roles. However, we do not provide tools to do this.)

Alternatively, we provide a *role diarization* tool that directly segments recordings into 3 main roles, namely child, male adult, female adult; and these separated from silence.

In both cases, you may want to classify each vocalizations into different types with the *vocal type classification* tool.

3.5.3 How to run a Speech or Voice activity detection tool

For these tools, type a command like this one:

```
$ vagrant ssh -c "noisemesSad.sh data/mydata/"
```

You can read that command as follows:

vagrant ssh -c: This tells DiViMe that it needs to run a tool.

noisemesSad.sh: This first argument tells DiViMe which tool to run. The options are: noisemesSad.sh, tocomboSad.sh, opensmileSad.sh

data/mydata/: This second argument tells DiViMe where are the sound files to analyze. Note that the directory containing the input files should be located in the `data/` directory (or it can be `data/` itself). The directory does not need to be called `mydata` - you can choose any name.

For each input wav file, there will be one rttm file created in the same directory, with the name of the tool added at the beginning. For example, imagine you have put a single file called participant23.wav into `data/`, and you decided to run two SADs:

```
$ vagrant ssh -c "opensmileSad.sh data/"
$ vagrant ssh -c "noisemesSad.sh data/"
```

This will result in your having the following three files in your `data/` folder:

- participant23.wav
- opensmileSad_participant23.rttm
- noisemesSad_participant23.rttm

If you look inside one of these .rttm's, say the opensmileSad one, it will look as follows:

```
SPEAKER        participant23        1        0.00        0.77        <NA>        <NA>
↪        speech        <NA>
SPEAKER        participant23        1        1.38        2.14        <NA>        <NA>
↪        speech        <NA>
```

This means that opensmileSad considered that the first 770 milliseconds of the audio were speech; followed by 610 milliseconds of non-speech, followed by 2.14 seconds of speech; etc.

3.5.4 How to run a Talker diarization tool

For these tools, type a command like this one:

```
$ vagrant ssh -c "diartk.sh data/mydata/ noisemesSad"
```

You can read that command as follows:

vagrant ssh -c: This tells DiViMe that it needs to run a tool.

diartk.sh: This first argument tells DiViMe which tool to run. The options are: diartk.sh.

data/mydata/: This second argument tells DiViMe where are the sound files to analyze. Note that the directory containing the input files should be located in the `data/` directory (or it can be `data/` itself). The directory does not need to be called `mydata` - you can choose any name.

noisemesSad: Remember that this tool does "talker diarization": Given some speech, attribute it to a speaker. Therefore, this type of tool necessitates speech/voice activity detection. This third argument tells DiViMe what file contains information about which sections of the sound file contain speech.

You can only use one of the following options: rttm, opensmileSad, tocomboSad, noisemesSad. We explain each of these options next.

You can provide annotations done by a human or in some other way, and encoded as rttms. If you have a different format, see the Format section. *What is crucial for this procedure to work is that your rttm's reflection your human-annotation are called exactly like your sound files.* Notice that all annotations that say "speech" in the eighth column count as such.

Alternatively, you can use automatic annotations generated by DiViMe's speech/voice activity detection systems, encoded in rttm files. In this case, you would pass one of the following options:

- noisemesSad: this means you want the system to use the output of the noisemesSad system. If you have not run noisemesSad, the system will fail.
- opensmileSad: this means you want the system to use the output of the opensmile system. If you have not run this system before, the system will fail.
- tocomboSad: this means you want the system to use the output of the tocomboSad system. If you have not ran this system before, the system will fail.

If the third parameter is not provided, the system will give an error.

If all three parameters are provided, then the system will first find all the annotation files matching the third parameter (e.g., all the human-annotated files *.rttm; or all the tocomboSad_*.rttm files), and then find the corresponding sound files. For example, imagine you have put into your `data/mydata/` folder the following files:

- participant23.wav
- opensmileSad_participant23.rttm
- participant24.wav
- participant24.rttm

If you run:

```
$ vagrant ssh -c "diartk.sh data/mydata/ opensmileSad"
```

then only participant23.wav will be analyzed.

If you run:

```
$ vagrant ssh -c "diartk.sh data/mydata/ rttm"
```

then only participant24.wav will be analyzed.

At the end of the process, there will be an added rttm file for each analyzed file. For instance, if you have just one sound file (participant23.wav) at the beginning and you run opensmileSad followed by diartk, then you will end up with the following three files:

- `participant23.wav`: your original sound file
- `opensmileSad_participant23.rttm`: the output of opensmileSad, which states where there is speech
- `diartk_opensmileSad_participant23.rttm`: the output of opensmileSad followed by diartk, which states which speech sections belong to which speakers.

See Format section for explanation on how to read the resulting rttm.

3.5.5 How to run a talker type tool

For these tools, type a command like this one:

```
$ vagrant ssh -c "yunitator.sh data/mydata/"
```

You can read that command as follows:

vagrant ssh -c: This tells DiViMe that it needs to run a tool.

yunitator.sh: This first argument tells DiViMe which tool to run. The options are: yunitator.

data/mydata/: This second argument tells DiViMe where are the sound files to analyze. Note that the directory containing the input files should be located in the `data/` directory (or it can be `data/` itself). The directory does not need to be called `mydata` - you can choose any name.

It returns one rttm per sound file, with an estimation of where there are vocalizations by children, female adults, and male adults. See Format section for explanation on how to read the resulting rttm.

3.5.6 How to run a Vocalization classification tool

For these tools, type a command like this one:

```
$ vagrant ssh -c "vcm.sh data/mydata/"
```

You can read that command as follows:

vagrant ssh -c: This tells DiViMe that it needs to run a tool.

vcm.sh: This first argument tells DiViMe which tool to run. The options are: vcm.

data/mydata/: This second argument tells DiViMe where are the sound files to analyze. Note that the directory containing the input files should be located in the `data/` directory (or it can be `data/` itself). The directory does not need to be called `mydata` - you can choose any name.

The vocalization classification tool depends on the output of the talker type tool yunitator. Therefore, the directory where you put your clips to analyze must contain files called yunitator_*.rttm (e.g., yunitator_participant23.wav).

The vocalization classification tool returns one rttm per sound file, with an estimation for each CHI vocalzsation to be a canonical syllable (CNS), non-canonical syllable (NCS), crying (CRY), and others (OTH, normally refer to laughing).

See Format section for explanation on how to read the resulting rttm.

3.5.7 How to run an Evaluation

If you have some annotations that you have made, you probably want to know how well our tools did - how close they were to your hard-earned human annotations.

Type a command like the one below:

‘vagrant ssh -c “eval.sh data/ tocomboSad accuracy”‘‘

You can read that command as follows:

vagrant ssh -c: This tells DiViMe that it needs to run a tool.

eval.sh: This first argument tells DiViMe that we want to perform an evaluation.

data/: This second argument tells DiViMe where are the sound files that need to be evaluated. This directory must contain both the annotations generated by the model and the human-made ones.

noisemesSad: The third argument indicates which tool’s output to evaluate.

accuracy: The fourth argument tells DiViMe which metric need to be used to assess the model’s performances. Here, we want to use the well-known accuracy measure.

The output should look like this :

```
accuracy report
                          detection accuracy true positive true negative false␣
↪positive false negative
                                           %
item
my_file1.rttm                          48.73         30.49         27.16          44.
↪47          16.18
my_file2.rttm                          57.00         12.32         55.13          40.
↪36          10.53
TOTAL                                  52.86         42.81         82.29          84.
↪83          26.71
```

It generates a table showing the scores obtained for each file. Since it is usually not enough to look at the final metric (the detection accuracy here), the table also shows intermediate metrics, therefore allowing the user to have a better insight of model’s performances. Note that this table will be saved in the .csv format in the *data/* folder.

Here, all the metrics that are implemented :

Note that the identification task is the same as the diarization task when the one-to-one mapping between hypothesis classes and reference classes share the same labels. To assess diarization model’s performances in the identification mode, you need to type the following command :

```
vagrant ssh -c "eval.sh data/ diartk_tocomboSad completeness --identification"
```

If the flag *–identification* is not passed, the script will run in the diarization mode.

Note that you can ask to compute several metrics at once by typing :

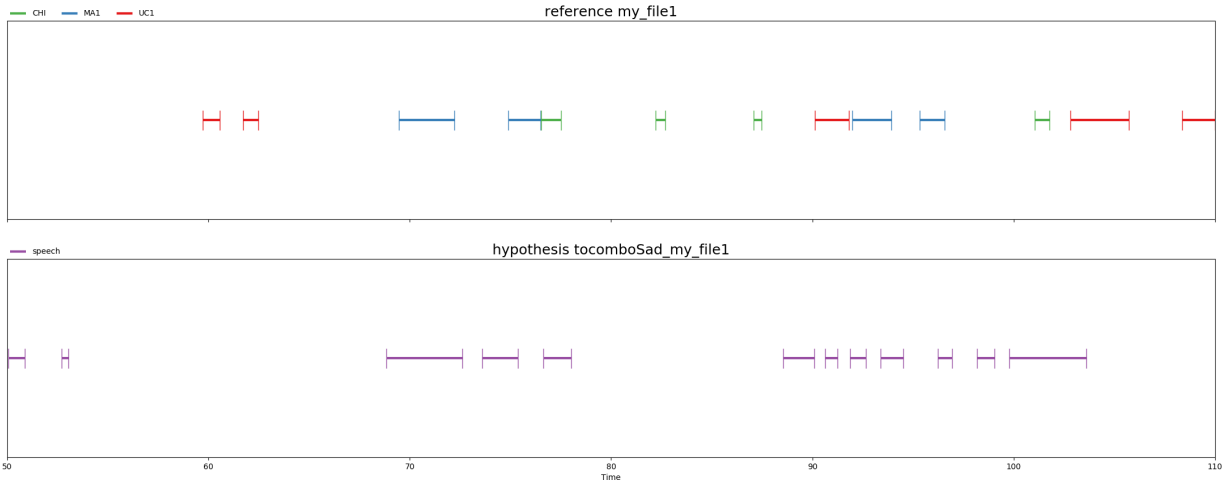
```
vagrant ssh -c "eval.sh data/ tocomboSad accuracy precision recall"
```

It will generated a report for each metric.

If you’re not a math person, you can add the –visualization flag by typing :

```
vagrant ssh -c "eval.sh data/ tocomboSad accuracy precision recall
--visualization"
```

It will extract the minute that contains the most speech for each file and align the reference and the hypothesis segments :



One minute alignement that has been obtained by adding the flag –visualization

Note that the process of calcuting the minute that contains the most speech can be time-consuming.

3.6 An alternative for Step 4: using recipes

It is possible to combine multiple steps into one program, which can then execute an entire complex analysis all by itself. Feel free to experiment. If you have good examples, **feel free to share**.

3.7 Further information on Step 5, putting DiViMe back to sleep

Last but not least, you should **remember to halt the virtual machine**. If you don't, it will continue running in the background, taking up useful resources! To do so, simply navigate to the DiViMe folder on your terminal and type in:

```
$ vagrant halt
```

CHAPTER 4

More information about DiViMe

DiViMe is a virtual machine, whose characteristics and contents are almost completely determined by provisioning scripts found in the DiViMe folder created when you did `git clone https://github.com/srvk/DiViMe/`. This section provides information to help you understand the DiViMe structure in conceptual terms (the "pipeline"). We also explain the process of "bootstrapping" or creation of this virtual machine. Finally, we describe the folder structure. If you just want to use the tools in DiViMe, you can probably skip this whole section, or read just the `Pipeline structure` section.

4.1 Pipeline Structure

DiViMe is a platform for tools to analyze naturalistic, unannotated audiorecordings. We consider this process to involve three kinds of processes:

- speech activity detection and voice activity detection = "detecting vocalizations",
- diarization = "deciding to whom the vocalizations belong", and
- "additional annotations"

Some tools actually combine two of these stages (e.g. a tool may do both speech activity detection and role attribution in one fell swoop).

4.2 Building a virtual machine

The structures and contents of the VM are actually built from reproducible directives which are executed when you do `vagrant up`. In technical terms, this is called VM is *provisioning*. In a nutshell, a file called "Vagrantfile" builds the machine (e.g., creates a virtual machine, allocates memory, installs an operating systems). Another file (conf/bootstrap.sh) installs tools within this virtual machine. For detailed documentation, see https://www.vagrantup.com/docs/vagrantfile/.

4.3 Folder Structure

By virtue of how DiViMe is provisioned, there is some information that is accessible both from the host machine (i.e., your computer) and the virtual machine (i.e., the minicomputer built inside your computer). These are called `shared folders`.

The following folders are shared between the host and the virtual machine, and thus accessible from both:

- *utils/* contains ancillary files, typically used across more than one tool
- *launcher/* contains files that target users will use to launch different tools
- *conf/* is for configuration files for two potential reasons: they are shared across more than one tool, or to make them more easily editable from outside the VM
- *data/* folder where users put the data to be analyzed, and recover the automatically annotated files

From within the virtual machine, these are inside /vagrant: /vagrant/utils, /vagrant/launcher, etc. Some of these are also accessible from the home of the default user, through links, ie, ~/utils and ~/launcher. (the default user is vagrant for virtualbox)

The following folders can only be found within the virtual machine:

- *repos/* contains any repository that is cloned into the machine, typically tools for the user, but also tools used by other tools to do e.g. feature extraction.

From within the virtual machine, ~/repos is inside the vagrant user's home folder, so you can access it as ~/repos/ or /home/vagrant/repos.

CHAPTER 5

Formats

This section explains the input and output formats.

5.1 Overview

The basic file format within DiViMe is a modified form of rttm, which is standard in key diarization tasks, and which allows us to evaluate most tools using a standardized evaluation routine. The different ways of using this same general format are explained below.

Many users, however, will be interested in knowing how to convert into and out of this format into something that is more commonly used for annotation. DiViMe includes routines to convert from any .TextGrid file (produced by the program Praat); from any .cha file (produced by the program CLAN); and from .eaf files (produced by the program ELAN) that have been generated using the ACLEW Annotation Scheme template. Users who rely on a different program (or for .eaf, on a different template) are advised to use the tools in their program or similar others to convert into one of these, or directly into the rttm format explained further below. We also provide code to convert the .its files outputted by the LENA(R) software, so you can evaluate them against your own coding and/or DiViMe's output.

The following subsections explain how to convert from each of these input formats into a basic reference rttm.

5.1.1 Input: TextGrid

Our process assumes that you will have the speech of each talker diarized in a different tier, using filled intervals when the person is talking, and empty intervals when they are not talking.

Notice that all tiers count, so if you have some tiers that are non-speech, you should remove them from your textgrids before you start. For example, if you have three tiers associated with a talker, coding e.g., what they say, how they say it, and to whom, you should remove 2 of these three tiers, because otherwise each of these tiers will be treated as a different talker.

Furthermore, any interval that is empty will be seen as *not* containing vocalizations from that speaker. Thus, if your coding is sparse (i.e., if you have a day-long recording, but have only coded some clips here and there), then you

should extract the audio clips and annotations for the sections that have been coded, and not process the whole day-long recording. (If you do, then your evaluation will be off, because all the speech found in sections you have not coded count towards false positives, as if the system had found speech when none was there.)

Additionally, the name of the tier is what will be taken to be the speaker's name. Therefore, if you have tiers that code speech by a speaker but are named differently, change the tier's name before starting. In fact, some of the tools assume a specific set of names, and thus the tool's output can only be properly evaluated if you use those names. In particular, the child wearing the recording device should be called "CHI". Other children should be called "XC0", where the X is the child's sex (F for Female, M for Male, U for uncertain/undecided/unknown) and 0 is a number 0-9 to identify a unique child. Similarly, adults should be called "XA0" where the X is the child's sex (F for Female, M for Male, U for uncertain/undecided/unknown) and 0 is a number 0-9 to identify a unique adult. Further, you can also define "XU0", a person of unknown age; and "EE0," a voice from a non-human source, such as a toy, radio, or TV.

Once you have removed all tiers that do not pertain to speakers, made sure that all the empty intervals really are non-vocalizations, renamed the tiers with the speaker's name, and ideally used this set of names, you are ready to convert your .TextGrid files into rttm. After you have put all the files you want to convert inside the folder data/mydata/, you would next run:

```
vagrant ssh -c "textgrid2rttm_folder.sh $j"
```

5.1.2 Input: cha

Our process extracts time stamps from bullet points, assuming that all vocalizations are coded. Therefore, if some of your entries do not have bullet points (e.g., "*CHI: 0 [=! crying]."), they will be treated as if there was no speech/vocalizations at that point. Furthermore, additional pre-processing steps are necessary if your coding is sparse (i.e., if you have a day-long recording, but have only coded some clips here and there) since our automatic extraction method has no way of knowing that you have skipped sections. If this is the case, then you should extract the audio clips and annotations for the sections that have been coded, and not process the whole day-long recording. (If you do, then your evaluation will be off, because all the speech found in sections you have not coded count towards false positives, as if the system had found speech when none was there.)

Additionally, some of the tools assume a specific set of names, and thus the tool's output can only be properly evaluated if you use those names. In particular, the child wearing the recording device should be called "CHI". Other children should be called "XC0", where the X is the child's sex (F for Female, M for Male, U for uncertain/undecided/unknown) and 0 is a number 0-9 to identify a unique child. Similarly, adults should be called "XA0" where the X is the child's sex (F for Female, M for Male, U for uncertain/undecided/unknown) and 0 is a number 0-9 to identify a unique adult. Further, you can also define "XU0", a person of unknown age; and "EE0," a voice from a non-human source, such as a toy, radio, or TV.

Once you have made sure that all lines of interest has bullet points, that there are no regions of the recording that have been skipped, and (if you want to use all tools) that your speakers follow this naming convention, you are ready to convert your .cha files into rttm. After you have put all the files you want to convert inside the folder data/mydata/, you would next run:

```
vagrant ssh -c "chat2rttm_folder.sh data/mydata/"
```

5.1.3 Input: Eaf

Since .eaf files can vary a lot in structure, we only provide tools to properly process .eaf files that follow the ACLEW Annotation Scheme template. One of the perks of using this format is that you can make full use of all tools in DiViMe, including a phonologization of your orthographic transcriptions into phonemic transcriptions, which will allow you to evaluate WCE in your data. For the phonologization stage, you need to provide the language, which can be: spanish, english, tzeltal.

The fourth column indicates the onset of a speech region; the forth column the duration of that speech region. All other columns may be ignored. Regions of non-speech are all the others (e.g., in this example, between .77 and 1.38).

5.2 Diarization style (diarization or role assignment) output

RTTM is an annotation format for audio files well designed for diarization. Explanations about how to write and read .rttm files can be found here This format is used by the DiViMe.

Diarization-type tools return one rttm per audio file, named toolnameDiar_filename.rttm, which looks like this:

```
SPEAKER file17  1       4.2     0.4  <NA>   talker0        <NA>
SPEAKER file17  1       4.6     1.2  <NA>   talker0        <NA>
SPEAKER file17  1       5.8     1.1  <NA>   talker1        <NA>
SPEAKER file17  1       6.9     1.2  <NA>   talker0        <NA>
SPEAKER file17  1       8.1     0.7  <NA>   talker1        <NA>
```

The fourth column indicates the onset of a region, the identity being indicated in ; the forth column the duration of that speech region. All other columns may be ignored. Regions of non-speech are all the others (e.g., in this example, between .77 and 1.38).

Tools that are of the speech/voice activity detection type return one rttm per audio file, named toolname-Sad_filename.rttm, which looks like this:

```
SPEAKER file17 1 0.00 0.77 <NA> <NA> speech <NA>
SPEAKER file17 1 1.38 2.14 <NA> <NA> speech <NA>
```

The third column indicates the onset of a region; the forth column the duration of that region. All other columns may be ignored. Regions of non-speech are all the others (e.g., in this example, between .77 and 1.38).

5.3 Output: rttm's from diarization tools

Diarization-type tools return one rttm per audio file, named toolnameDiar_filename.rttm, which looks like this:

```
SPEAKER file17 1 4.2 0.4 <NA> <NA> talker0 <NA>
SPEAKER file17 1 5.8 1.1 <NA> <NA> talker1 <NA>
SPEAKER file17 1 6.9 1.2 <NA> <NA> talker2 <NA>
SPEAKER file17 1 8.1 0.7 <NA> <NA> talker1 <NA>
```

The third column indicates the onset of a region, the forth column the duration of that region; and the identity of the speaker being indicated in the eighth column. All other columns may be ignored. Regions of non-speech are all the others (e.g., in this example, between 4.6 and 5.8).

5.4 Output: rttm's from talker type tools

Talker type tools return one rttm per audio file, named toolname_filename.rttm, which looks like this:

```
SPEAKER file17 1 4.2 0.4 <NA> <NA> CHI <NA>
SPEAKER file17 1 5.8 1.1 <NA> <NA> FA <NA>
SPEAKER file17 1 6.9 1.2 <NA> <NA> MA <NA>
SPEAKER file17 1 8.1 0.7 <NA> <NA> FA <NA>
```

The third column indicates the onset of a region, the forth column the duration of that region; and the speaker type being indicated in the eighth column: CHI is the key child, C is a child (target or other), FA is female adult, MA is male adult. All other columns may be ignored. Regions of non-speech are all the others (e.g., in this example, between 4.6 and 5.8).

5.5 Output: rttm's from vocal maturity tools

Vocal maturity tools return one rttm per audio file, named toolname_filename.rttm, which looks like this:

```
SPEAKER FILENAME 1 31.4 1.6 <NA> <NA> CNS 0.71 <NA>
SPEAKER FILENAME 1 34.6 1.1 <NA> <NA> NCS 0.81 <NA>
SPEAKER FILENAME 1 39.0 0.8 <NA> <NA> CRY 0.80 <NA>
SPEAKER FILENAME 1 47.9 0.5 <NA> <NA> NCS 0.62 <NA>
```

The third column indicates the onset of a region, the forth column the duration of that region; and the vocalization type being indicated in the eighth column: canonical syllable (CNS), non-cannoical syllable (NCS), crying (CRY), and others (OTH, normally refer to laughing); followed by the likelihood of that class (higher means the system was more "certain"). All other columns may be ignored. Regions of non-vocalization as well as regions where people other than the child vocalize are not marked (e.g., in this example, between 33 and 34.6).

CHAPTER 6

Speech or Voice activity detection tools

This section contains documentation from the different Speech or Voice activity detection tools.

6.1 NoisemesSad

6.1.1 General intro

Noiseme SAD was actually not specifically built as a SAD but rather as a broader "noiseme classifier". It is a neural network that can predict frame-level probabilities of 17 types of sound events (called "noisemes"), including speech, singing, engine noise, etc. The network consists of one single bidirectional LSTM layer with 400 hidden units in each direction. It was trained on 10h of basically web videos data (Strassel et al., 2012), with the Theano toolkit. The OpenSMILE toolkit (Eyben et al., 2013) is used to extract 6,669 low-level acoustic features, which are reduced to 50 dimensions with PCA. For our purposes, we summed the probabilities of the classes "speech" and "speech non-english" and labeled a region as speech if this probability was higher than all others. The original method in Wang et al. (2016) used 983 features selected using information gain criterion, but we used an updated version from authors Y. Wang and F. Metze in this paper.

6.1.2 Instructions for direct use (ATTENTION, MAY BE OUTDATED)

You can analyze just one file as follows. Imagine that <$MYFILE> is the name of the file you want to analyze, which you've put inside the `data/` folder in the current working directory.

```
$ vagrant ssh -c "OpenSAT/runOpenSAT.sh data/<$MYFILE>"
```

You can also analyze a group of files as follows:

```
$ vagrant ssh -c "OpenSAT/runDiarNoisemes.sh data/"
```

This will analyze all .wav's inside the "data" folder.

Created annotations will be stored inside the same "data" folder.

This system will classify slices of the audio recording into one of 17 noiseme classes:

- background
- speech
- speech non English
- mumble
- singing alone
- music + singing
- music alone
- human sounds
- cheer
- crowd sounds
- animal sounds
- engine
- noise_ongoing
- noise_pulse
- noise_tone
- noise_nature
- white_noise
- radio

Some more technical details

For more fine grained control, you can log into the VM and from a command line, and play around from inside the "Diarization with noisemes" directory, called "OpenSAT":

```
$ vagrant ssh
$ cd OpenSAT
```

The main script is runOpenSAT.sh and takes one argument: an audio file in .wav format. Upon successful completion, output will be in the folder (relative to ~/OpenSAT) `SSSF/data/hyp/<input audiofile basename>/confidence.pkl.gz`

The system will grind first creating features for all the .wav files it found, then will place those features in a subfolder `feature`. Then it will load a model, and process all the features generated, producing output in a subfolder `hyp/` two files per input: `<inputfile>.confidence.mat` and `<inputfile>.confidence.pkl.gz` - a confidence matrix in Matlab v5 mat-file format, and a Python compressed data 'pickle' file. Now, as well, in the `hyp/` folder, `<inputfile>.rttm` with labels found from a config file noisemeclasses.txt

-More details on output format-

The 18 classes are as follows:

```
0        background
1        speech
2        speech_ne
3        mumble
```

(continues on next page)

(continued from previous page)

```
4         singing
5         musicSing
6         music
7         human
8         cheer
9         crowd
10         animal
11         engine
12         noise_ongoing
13         noise_pulse
14         noise_tone
15         noise_nature
16         white_noise
17         radio
```

The frame length is 0.1s. The system also uses a 2-second window, so the i-th frame starts at (0.1 * i - 2) seconds and finishes at (0.1 * i) seconds. That's why 60 seconds become 620 frames. 'speech_ne' means non-English speech

-Sample RTTM output snippet-

```
SPEAKER family  1       4.2     0.4     noise_ongoing <NA>    <NA>    0.37730383873
SPEAKER family  1       4.6     1.2     background    <NA>    <NA>    0.327808111906
SPEAKER family  1       5.8     1.1     speech        <NA>    <NA>    0.430758684874
SPEAKER family  1       6.9     1.2     background    <NA>    <NA>    0.401730179787
SPEAKER family  1       8.1     0.7     speech        <NA>    <NA>    0.407463937998
SPEAKER family  1       8.8     1.1     background    <NA>    <NA>    0.37258502841
SPEAKER family  1       9.9     1.7     noise_ongoing <NA>    <NA>    0.315185159445
```

The script `runClasses.sh` works like `runDiarNoisemes.sh`, but produces the more detailed results as seen above.

6.1.3 Main references:

Wang, Y., Neves, L., & Metze, F. (2016, March). Audio-based multimedia event detection using deep recurrent neural networks. In Acoustics, Speech and Signal Processing (ICASSP), 2016 IEEE International Conference on (pp. 2742-2746). IEEE. pdf

6.1.4 Associated references:

S.Burger,Q.Jin,P.F.Schulam,andF.Metze,"Noisemes:Man- ual annotation of environmental noise in audio streams," Carnegie Mellon University, Pittsburgh, PA; U.S.A., Tech. Rep. CMU-LTI- 12-07, 2012. S.Strassel,A.Morris,J.G.Fiscus,C.Caruso,H.Lee,P.D.Over, J. Fiumara, B. L. Shaw, B. Antonishek, and M. Michel, "Creating havic: Heterogeneous audio visual internet collection," in Proc. LREC. Istanbul, Turkey: ELRA, May 2012. F. Eyben, F. Weninger, F. Gross, and B. Schuller, "Recent developments in opensmile, the munich open-source multimedia fea- ture extractor," in Proceedings of the 21st ACM international con- ference on Multimedia. ACM, 2013, pp. 835–838.

6.1.5 Questions and bug reports

http://github.com/srvk/OpenSAT/issues

6.2 OpenSmile SAD

6.2.1 General intro

openSMILE SAD relies on openSMILE (Eyben et al., 2013a) to generate an 18-coefficient RASTA-PLP plus first order delta features. It then uses a long short-term memory recurrent neural network (see details in Eyben et al., 2013b) that has been pre-trained on two corpora of read and spontaneous speech by adults recorded in laboratory conditions, augmented with various noise types.

6.2.2 Some more technical details

These are the parameters that are being used with values that depart from the openSMILE default settings (quoted material comes from either of the openSMILE manuals):

monoMixdown = 1, means "mix down all recorded channels to 1 mono channel" noHeader = 0, means read the RIFF header (don't need to specify the parameters 'sampleRate', 'channels', and possibly 'sampleSize') preSil = 0.1 "Specifies the amount of silence at the turn beginning in seconds, i.e. the lag of the turn detector. This is the length of the data that will be added to the current segment prior to the turn start time received in the message from the turn detector component"; we use a tighter criterion than the default (.2) postSil = 0.1 "Specifies the amount of silence at the turn end in seconds. This is the length of the data that will be added to the current segment after to the turn end time received in the message from the turn detector component."; we use a tighter criterion than the default (.3)

You can change these parameters locally by doing:

```
$ vagrant ssh
$ nano /vagrant/conf/vad/vadSegmenter_aclew.conf
```

openSMILE manuals consulted:

- Eyben, F., Woellmer, M., & Schuller, B. (2013). openSMILE: The Munich open Speech and Music Interpretation by Large space Extraction toolkit. Institute for Human-Machine Communication, version 2.0. http://download2.nust.na/pub4/sourceforge/o/project/op/opensmile/openSMILE_book_2.0-rc1.pdf
- Eyben, F., Woellmer, M., & Schuller, B. (2016). openSMILE: open Source Media Interpretation by Large feture-space Extraction toolkit. Institute for Human-Machine Communication, version 2.3. https://www.audeering.com/research-and-open-source/files/openSMILE-book-latest.pdf

6.2.3 Main references for this tool:

Eyben, F. Weninger, F. Gross, F., &1 Schuller, B. (2013a). Recent developments in OpenSmile, the Munich open-source multimedia feature extractor. Proceedings of the 21st ACM international conference on Multimedia, 835–838.

Eyben, F., Weninger, F., Squartini, S., & Schuller, B. (2013b). Real-life voice activity detection with lstm recurrent neural networks and an application to hollywood movies. In Acoustics, Speech and Signal Processing (ICASSP), 2013 IEEE International Conference on (pp. 483-487). IEEE.

6.2.4 Questions and bug reports

https://www.audeering.com/technology/opensmile/#support

6.3 TOCombo SAD

6.3.1 General intro

This tool's name stands for "Threshold-Optimized" "combo" SAD; we explain each part in turn. It is a SAD because the goal is to extract speech activity. It is called "combo" because it combines linearly 4 different aspects of voicing (harmonicity , clarity, prediction gain, periodicity) in addition to one perceptual spectral flux feature (see details in Sadjadi & Hansen, 2013). These are extracted in 32-ms frames (with a 10 ms stride). The specific version included here corresponds to the Combo SAD introduced in Ziaei et al. (2014) and used further in Ziaei et al (2016). In this work, a threshold was optimized for daylong recordings, which typically have long silent periods, in order to avoid the usual overly large false alarm rates found in typical SAD systems provided with these data.

6.3.2 Main references:

Ziaei, A., Sangwan, A., & Hansen, J. H. (2014). A speech system for estimating daily word counts. In Fifteenth Annual Conference of the International Speech Communication Association. http://193.6.4.39/~czap/letoltes/IS14/IS2014/PDF/AUTHOR/IS141028.PDF A. Ziaei, A. Sangwan, J.H.L. Hansen, "Effective word count estimation for long duration daily naturalistic audio recordings," Speech Communication, vol. 84, pp. 15-23, Nov. 2016. S.O. Sadjadi, J.H.L. Hansen, "Unsupervised Speech Activity Detection using Voicing Measures and Perceptual Spectral Flux," IEEE Signal Processing Letters, vol. 20, no. 3, pp. 197-200, March 2013.

6.3.3 Questions and bug reports

Not available

CHAPTER 7

Talker diarization tools

This section contains documentation from the different Talker diarization tools (i.e., given a speech segment, decide who speaks).

7.1 DiarTK

7.1.1 General intro

This tool performs diarization, requiring as input not only .wav audio, but also speech/nonspeech in .rttm format, from human annotation, or potentially from one of the SAD or VAD tools included in this VM.

The DiarTK model imported in the VM is a C++ open source toolkit by Vijayasenan & Valente (2012). The algorithm first extracts MFCC features, then performs non-parametric clustering of the frames using agglomerative information bottleneck clustering. At the end of the process, the resulting clusters correspond to a set of speakers. The most likely Diarization sequence between those speakers is computed by Viterbi realignement.

We use this tool with the following parameter values:

- weight MFCC = 1 (default)
- Maximum Segment Duration 250 (default)
- Maximum number of clusters possible: 10 (default)
- Normalized Mutual Information threshold: 0.5 (default)
- Beta value: 10 (passed as parameter)
- Number of threads: 3 (passed as parameter)

7.1.2 Main references:

D. Vijayasenan and F. Valente, “Diartk: An open source toolkit for research in multistream speaker diarization and its application to meetings recordings,” in Thirteenth Annual Conference of the International Speech Communication

Association, 2012. https://pdfs.semanticscholar.org/71e3/9d42aadd9ec44a42aa5cd21202fedb5eaec5.pdf

7.1.3 Questions and bug reports

http://htk.eng.cam.ac.uk/bugs/buglist.shtml

CHAPTER 8

Other tools

This section contains documentation from other tools.

8.1 Yunitator

8.1.1 General intro

Given that there is no reference for this tool, we provide a more extensive introduction based on a presentation Florian Metze gave on 2018-08-13 in an ACLEW Meeting.

The data used for training were:

- ACLEW Starter dataset
- VanDam public 5-min dataset (about 13h; https://homebank.talkbank.org/access/Public/VanDam-5minute.html); noiseme-sad used to detect and remove intraturn silences

Talker identity annotations collapsed into the following 4 types:

- children (including both the child wearing the device and other children; class prior: .13)
- female adults (class prior .09)
- male adults (class prior .03)
- non-vocalizations (class prior .75)

The features were MED (multimedia event detection) feature, extracted with OpenSMILE. They were extracted in 2s windows moving 100ms each step. There were 6,669 dims at first, PCA-ed down to 50 dims

The model was a RNN, with 1 bidirectional GRU layer and 200 units in each direction. There was a softmax output layer, which therefore doesn't predict overlaps..

The training regime used 5-fold cross-validation, with 5 models trained on 4/5 of the data and tested on the remainder. The outputs are poooled together to measure performance. The final model was trained on all the data.

The loss function was cross entropy with classes weighted by 1/prior. The batch size was 5 sequences of 500 frames. The optimizer was SGD with Nesterov momentum=.9, the inital LR was .01 and the LR schedule was *=0.8 if frame accuracy doesn't reach new best in 4 epochs

The resulting F1 for the key classes were:

- Child .55 (Precision .55, recall .55)
- Male adult .43 (P .31, R .61)
- Female adult .55 (P .5, R .62)

8.1.2 Main references:

There is no official reference for this tool.

8.1.3 Questions and bug reports

Not available

8.2 VCM

8.2.1 General intro

Two independent models: one (modelLing) to predicts linguistic vs. non-linguistic infant vocalisations; the other one (modelSyll) predicts canonical vs. non-canonical syllables if given a linguistic infant vocalization.

Specifically, the modelLing was trained on an infant linguistic dataset (refer to this paper: https://static1.squarespace.com/static/591f756559cc68d09fc2e308/t/5b3a94cb758d4645603085db/1530565836736/ZhangEtAl_2018.pd and modelSyll was trained on another infant syllable vocalisation dataset (refer to this paper: https://pdfs.semanticscholar.org/2b25/bc84d2c4668e6d17f4f9343106f726198cd0.pdf).

Feature set: 88 eGeMAPS extracted by openSMILE-2.3.0 on the segment level.

Model: two hidden layers feed-forward neural networks with 1024 hidden nodes per each hidden layer. A log_softmax layer is stacked as an output layer. The optimiser was set to SGD with a learning rate 0.01, and the batch size is 64.

Setups: Both the infant linguistic and syllable vocalisation datasets were split into train, development, and test partitions following a speaker independent strategy.

Results: The results are 67.5% UAR and 76.6% WAR on the test set for the lingustic voc classification; and are 70.4% UAR and 69.2% WAR for the syllable voc classification.

8.2.2 Main references:

There is no official reference for this tool.

8.2.3 Questions and bug reports

https://github.com/srvk/vcm/issues/

CHAPTER 9

Evaluation

9.1 Speech/voice activity detection and diarization evaluation

For SAD, we employ an evaluation, which returns the false alarm (FA) rate (proportion of frames labeled as speech that were non-speech in the gold annotation) and missed speech rate (proportion of frames labeled as non-speech that were speech in the gold annotation). For TD, we employ the evaluation developed for the DiHARD Challenge, which returns a Diarization error rate (DER), which sums percentage of speaker error (mismatch in speaker IDs), false alarm speech (non-speech segments assigned to a speaker) and missed speech (unassigned speech).

One important consideration is in order: What to do with files that have no speech to begin with, or where the system does not return any speech at the SAD stage or any labels at the TD stage. This is not a case that is often discussed in the literature because recordings are typically targeted at moments where there is speech. However, in naturalistic recordings, some extracts may not contain any speech activity, and thus one must adopt a coherent framework for the evaluation of such instances. We opted for the following decisions.

If the gold annotation was empty, and the SAD system returned no speech labels, then the FA = 0 and M = 0; but if the SAD system returned some speech labels, then FA = 100 and M = 0. Also, if the gold annotation was not empty and the system did not find any speech, then this was treated as FA = 0 and M=100.

As for the TD evaluation, the same decisions were used above for FA and M, and the following decisions were made for mismatch. If the gold annotation was empty, regardless of what the system returned, the mismatch rate was treated as 0. If the gold annotation was empty but a pipeline returned no TD labels (either because the SAD in that system did not detect any speech, or because the diarization failed), then this was penalized via a miss of 100 (as above), but not further penalized in terms of talker mismatch, which was set at 0.

CHAPTER 10

Word count estimation (WCE) tool

This section contains documentation for word count estimation tool.

10.1 Basic description

The purpose of the WCE tool is to provide an estimate of the number of words in an utterance that is provided as an input to the algorithm. It is based on automatic syllabification of speech, followed by mapping of syllable counts + a number of other acoustic features to word counts. The basic functionality is described in Räsänen et al. (submitted).

There are two basic ways to use the current tool:

1. An out-of-the-box version that can be used directly on any speech data , and
2. An adapted version, where the WCE system is first adapted to data provided by the user, and then performs more accurately on similar data.

In practice, out-of-the-box variant provides only syllable count estimates for the input data, while the adaptation mode can be used to re-train the system to provide meaningful word count estimates in any language or other data domain. These two operation modes are detailed further below.

Note that the tool assumes that the incoming signals are already segmented into utterances (e.g., using a SAD and/or a diarization tool), as it does not have an internal module for separating speech from non-speech contents.

By default, the WCE tool is using a bi-directional long short-term memory (LSTM) -based syllabifier, trained on four different languages. The package also contains a syllabifier stage from speaking-rate estimator published by Wang & Narayanan (2007), as implemented for MATLAB in Räsänen et al. (2018), and an oscillator-based syllabifier described in Räsänen et al. (2018). The default syllabifier can be changed from configuration files (see "Changing configuration").

All research or other use utilizing this WCE tool should cite the following paper:

Räsänen, O., Seshadri, S., Karadayi, J., Riebling, E., Bunce, J., Cristia, A., Metze, F., Casillas, M., Rosemberg, C., Bergelson, E., & Soderstrom, M. (submitted). Automatic word count estimation from daylong child-centered recordings in various language environments using language-independent syllabification of speech. In review.

10.2 Instructions for direct use (out-of-the-box version)

To get syllable count estimates on your audio files, run

vagrant ssh -c "~/launcher/estimateWCE.sh data/my_audiofolder/ data/WCE_output.txt"

which will run WCE on all .wav files in data/my_audiofolder/ and output results to data/WCE_output.txt.

10.3 Instructions for adapting the WCE to new language

NOTE: DOCUMENTATION INCOMPLETE

NOTE: This requires audio files and .eaf annotation files in ACLEW DAS annotation format.

10.4 Changing configuration

NOTE: DOCUMENTATION INCOMPLETE

10.5 Main references for this tool:

Räsänen, O., Seshadri, S. & Casillas, M. (2018). Comparison of Syllabification Algorithms and Training Strategies for Robust Word Count Estimation across Different Languages and Recording Conditions. Proc. Interspeech-201, Hyderabad, India, pp. 1200–1204.

Räsänen, O., Seshadri, S., Karadayi, J., Riebling, E., Bunce, J., Cristia, A., Metze, F., Casillas, M., Rosemberg, C., Bergelson, E., & Soderstrom, M. (submitted). Automatic word count estimation from daylong child-centered recordings in various language environments using language-independent syllabification of speech. In review.

Räsänen, O., Doyle, G., & Frank, M. C. (2018). Pre-linguistic segmentation of speech into syllable-like units. Cognition, 171, 130–150.

Wang, D., & Narayanan, S. (2007). Robust speech rate estimation for spontaneous speech. IEEE Transactions on Audio, Speech, and Language Processing, 15, 2190–2201.

10.6 Questions and bug reports

Send questions & Bug reports to Okko Räsänen (firstname.surname @ aalto.fi)

CHAPTER 11

Extra tools

11.1 Getting sample data

11.1.1 ACLEW Starter Dataset

The ACLEW Starter dataset is freely available, and can be downloaded in order to test the tools. To download it, using your terminal, as explained before, go in the DiViMe folder and do: `$ vagrant ssh -c "launcher/get_aclewStarter.sh data/aclewStarter/"`

This will create a folder called aclewStarter inside data, in which you will find the audio files from the public dataset and their corresponding .rttm annotations. At the present time, there are only two matched annotation-wav files, totalling 10 minutes of running recordings

You can then use the tools mentioned before, by replacing the "data/" folder in the command given in the previous paragraph by "aclewStarter/", E.G for noisemes:

```
$ vagrant ssh -c "launcher/noisemesSad.sh /"
```

Reference for the ACLEW Starter dataset:

Bergelson, E., Warlaumont, A., Cristia, A., Casillas, M., Rosemberg, C., Soderstrom, M., Rowland, C., Durrant, S. & Bunce, J. (2017). Starter-ACLEW. Databrary. Retrieved August 15, 2018 from http://doi.org/10.17910/B7.390.

11.2 Using scripts in the Utilities

11.2.1 elan2rttm.py

Convert annotations made using the ELAN tool, in .eaf format, into rttm transcriptions. Please note that, using this script, some information is lost, notably the vocal maturity annotation (coded in tier vcm@CHI), does not appear in the RTTM format. These information could be retrieved and put in the rttm. If you need this information in the RTTM, please raise an issue on github.

11.2.2 textgrid2rttm.py

Convert annotations made using Praat, in .TextGrid format, into rttm transcriptions. Requires:

- pympi
- tgt

We provide code to translate annotations from other formats into RTTM:

ELAN .eaf format

WARNING: the current version does not handle subtypes when parsing annotations e.g. TIER_ID 'CHI' would be written in the RTTM output file but 'vmc@CHI' would not. This is due to the time references being based on other TIER_ID's annotations for subtypes.

From within the machine, you would run the script as follows:

```
python utils/elan2rttm.py -i my_file.eaf -o my_output_folder
```

Praat TextGrid format

From within the machine, you would run the script as follows:

```
python utils/textgrid2rttm.py my_input_folder
```

11.2.3 adjust_timestamps.py

This script is specific to the data in ACLEW, with the ACLEW annotations conventions. It takes as input a daylong recording in wav format (available on databrary), and a transcription in .eaf format that contains annotated segment coded in an "on_off" tier (or "code" tier for some corpora that were annotated before the new convention). It then takes each annotated segment of 2 minutes, extract it from the daylong recording to output a small wav file of 2 minutes, with the name: corpus_id_onset_offset.wav where corpus is the name of the original corpus, id is the name of the daylong recording (which is itself the id given to the recorded child), onset is where the segment starts in the daylong recording (in seconds, with 6 digits padded with 0's if necessary), offset is where the segment ends in the daylong recording (with the same convention). For each of these segments extracted, it also writes the annotations in rttm format, with the timestamps adapted to correspond to the small wav, and with the same name as the small wav previously written.

11.2.4 remove_overlap_rttm.py

Take a transcription in RTTM format, and convert it to a SAD annotation in RTTM format. The SAD annotation contains less information, as it only indicated "speech" segment (i.e. the talker is written as "speech", no matter who the original talker is), and there are no overlap between speech segments.

make_big_corpus.sh

This script is called to treat all the daylong recording with their annotations, using the previous adjust_timestamps.py script. It also creates gold SAD rttm using the remove_overlap_rttm.py script previously described.

CHAPTER 12

Troubleshooting

12.1 Installation issues

12.1.1 Virtual Machine creation

If your computer freezes after `vagrant up`, it may be due to several things. If your OS is ubuntu 16.04, there's a known incompatibility between VirtualBox and the 4.13 Linux kernel on ubuntu 16.04. What you may do is to install a previous version of the kernel, for example the 4.10, following these instructions, or install the latest version of virtualbox which should fix the problem. If you are not on ubuntu 16.04, or if the previous fix didn't work, it may also be due to the fact that Vagrant is trying to create a Virtual Machine that asks for too much resources. Please ensure that you have enough space on your computer (you should have at least 15Gb of free space) and check that the memory asked for is okay. If not, you can lower the memory of the VM by changing line 25 of the VagrantFile,

```
vbox.memory = 3072
```

to a lower number, such as

```
vbox.memory = 2048
```

12.1.2 Resuming the Virtual Machine

If you already used the VM once, shut down your computer, turned it back on and can't seem to be able to do `vagrant up` again, you can simply do

```
vagrant destroy
```

and recreate the VM using

```
vagrant up
```

If you don't want to destroy it, you can try opening the VirtualBox GUI, go to `File -> Settings` or `Preferences -> Network`, click on the `Host-only Networks` tab, then click the network card icon with

the green plus sign in the right, if there are no networks yet listed. The resulting new default network should appear with the name 'vboxnet0'. You can now try again with `vagrant up`

12.2 Problems with some of the Tools

12.2.1 OpenSmile, DiarTK

If OpenSmile, DiarTK don't seem to work after `vagrant up`, first, please check that you indeed have the htk archive in your folder. If you don't, please put it there and launch:

```
vagrant up --provision
```

This step will install HTK inside the VM, which is used by several tools.

If you use the noisemesSad or the noisemes_full tool, one problem you may encounter is that it doesn't treat all of your files and gives you an error that looks like this:

```
Traceback (most recent call last):
  File "SSSF/code/predict/1-confidence-vm5.py", line 59, in <module>
    feature = pca(readHtk(os.path.join(INPUT_DIR, filename))).astype('float32')
  File "/home/vagrant/G/coconut/fileutils/htk.py", line 16, in readHtk
    data = struct.unpack(">%df" % (nSamples * sampSize / 4), f.read(nSamples *␣
↪sampSize))
MemoryError
```

If this happens to you, it's because you are trying to treat more data than the system/your computer can handle. What you can do is simply put the remaining files that weren't treated in a separate folder and treat this folder separately (and do this until all of your files are treated if it happens again on very big datasets). After that, you can put back all of your data in the same folder.

CHAPTER 13

Instructions For Contributors

Temporary instructions: We have reorganized DiViMe to try to facilitate tool incorporation and VM use, but these changes have not yet made it to the main branch. Therefore, all of the following instructions are NOT in the master branch, but in the major_reorganization branch. This should make no difference to you, except that you need to check out the right branch to view the VM fitting with these instructions. We provide the code for doing this below.

13.1 Overview

This detailed guide provides you with step-by-step, specific instructions for adapting a tool into the DiViMe environment. The following is a Summary of the steps:

1. Install the VM for yourself
2. Adapt & test your tools in the VM environment, and build the necessary links to other modules.
3. Put your tools and all your custom scripts somewhere where they can be downloaded (e.g., GitHub repo(s))
4. Create a bash script to download and install your tools on the VM (the same steps will be added to a Vagrantfile that controls the entire VM installation)
5. ...

13.2 Before You Start

Before you start, you'll need to get the VM up and running to establish the environment in which your tool will run.

1. Install DiViMe as per the installation instructions, including the `vagrant up` that wakes the machine up.

Temporary instructions: IMPORTANT: After you've cloned DiViMe, you should check out the major_reorganization branch, as follows:

```
$ git checkout remotes/origin/major_reorganization
```

If ever you want to go back to the master branch, you'd do:

```
$ git checkout master
```

If you've already built one version of the VM, you'll need to do:

```
vagrant destroy
vagrant up
```

1. Run the test.sh routine routine. It downloads audio recordings and annotations that you will need to make sure your tool is compatible with the rest of the workflow. Additionally, it will help you spot any issues there may be in your local environment.

13.3 Understanding the general structure of DiViMe

DiViMe is a virtual machine, whose characteristics and contents are almost completely determined by provisioning scripts found at the top level when you clone DiViMe. That means that if you want to contribute to DiViMe, you need to understand its structure, and bootstrapping of the virtual machine.

In the next section, we will walk you through some stages of modifying the contents of the VM, which you will do initially "by hand". However, remember the contents of the VM are actually built from reproducible directives ran when you do `vagrant up`. (In technical terms, when the VM is *provisioned*.) Therefore, any changes you make by hand when you are logged into the VM will be lost when you run `vagrant destroy`. Please bear that in mind when trying out the steps in the next section.

Additionally, by the end of these instructions you will be ready to propose a revised version of the VM - i.e., a new recipe to *provision* the VM. Therefore, any changes made by hand that you wish to make permanent in the VM should eventually be added to a file called `util/bootstraph.sh`. So you may want to have a copy of bootstraph.sh open to make notes in there of the steps you take, which should be reproduced when provisioning the VM.

13.4 Adapting Your Tool to the VM Environment

1. For this section, you'll be working almost entirely within the virtual machine. So start by doing `vagrant ssh` to log in. This logs you in as the `vagrant` user, which also has sudo privileges to do things like install software, change permissions, etc. in the virtual machine.
2. Decide where your tool is made available so that it can be copied into the VM. Ideally, it will be somewhere public, such as GitHub, so that anyone rebuilding the machine will get access to it. Alternatively, your tool might be stored on a server in a location you control, and then pulled into the virtual machine. Please note that the latter solution only preserves the anonymity of your code temporarily; if you get to the end of this document, when you are proposing a new version of the VM including your tool, then the URL needs to be known to the world, and accessible to everyone. Please note that neither alternative forces you to make your code open source. You can also share a binary executable (ask us for instructions if you need help).
3. Import your tool into the VM. Once you have decided where to put your tool, install your code by hand in the location where it will be within the machine: `/home/vagrant/repos`. For example if your code is in a public GitHub repository `https://github.com/srvk/OpenSAT`, you would type into the terminal:

```
cd /home/vagrant/repos
git clone http://github.com/srvk/OpenSAT
```

If your tool is accessible via URL `https://toolnameurl.com/tool.zip`, you would type into the terminal:

```
cd /home/vagrant/repos
wget -q -N https://toolnameurl.com/toolname.zip
unzip -q -o toolname.zip
```

In bootstrap.sh, add the same code to make this step permanently reproducible.

1. If your code requires certain linux packages that are not installed, first install them 'by hand' in the VM with a command like `sudo apt-get install <packagename>`. Any packages installed this way should be added similarly to one of the `apt-get` lines in bootstraph.sh like:

```
sudo apt-get install -y libxt-dev libx11-xcb1
```

1. The next step is to install additional dependencies, if any are needed. The VM already includes code to install OpenSmile 2.1, matlab v93, python 3, and anaconda 2.3.0. As in the two previous steps, you can do this by hand within the terminal, but you need to add the step to bootstraph.sh to make it permanent. For instance, if you need the python package `pympi-ling`, you would type `pip install pympi-ling` by hand into the terminal. Additionally, to make this change permanent (and have the dependencies installed when you reprovision the VM or when someone else rebuilds the VM), you need to add it to the Vagrantfile section where python packages are installed, with code like this:

```
su ${user} -c "/home/${user}/anaconda/bin/pip install pympi-ling"
```

13.5 Write a Wrapper

In this section, we provide detailed instructions for writing a wrapper that allows users to run your tool in a standardized way. The wrapper should be written in bash. Please look at the other launcher wrappers to have an idea of how to structure it, and read on for important considerations.

13.5.1 Naming Conventions

- You choose your own tool's name. Use anything you want except the other names already in use. We refer to your tool name later as 'TOOLNAME'
- It may be useful to you to decide at what "stage" of diarization your tool operates. A few things will be clearer if you have a good notion of when this is, such as the number of arguments and whether there is already an evaluation/scoring tool that can be re-used with your tool. We explain these things in more detail below.
- To facilitate end users' understanding of what tool they are using, we have systematically added the stage name to the wrapper's name, but you don't have to follow this procedure if you think it will not be useful.
- We have identified three stages with a fixed name: Sad (for both speech activity detection and voice activity detection), Diar (for speaker diarization and role assignment), and Add (for adding annotation dependent on role assignment).
- Other stages or stage combinations do not have fixed names. But please feel free to use these stage names. For instance, if your tool only requires audio files as input, then you can use Sad; if it operates on both audio and speech activity detection, then use Diar; and if it is specific to one talker role as input, use Add.

13.5.2 Tool Autonomy

Tools should be self-aware of where they have been installed, and they should use this awareness to find their dependencies. Said differently, a tool should run "in place" independent of the absolute path it's installed in. Tool dependency paths should be relative to the tool home folder, which should serve as the working directory. Again, please look at the

other launcher wrappers to reuse the code present at the top of the wrappers, which correctly reconstructs the absolute path of this folder.

13.5.3 Input, Output, and Parameters

Your wrapper should take at least one argument, namely the name of a folder containing data. This folder is appended to "/vagrant" so from your script and the VM's perspective, data appears in /vagrant/data/. This is actually a shared folder, coming from the host computer working directory. Everything in data/ on the host will in /vagrant/data in the VM, and vice-versa. The default wrapper argument, then, is typically "data/", but users could also supply "data/mystudy/" or "data/mystudy/baby1/" as the data folder argument. This supports the notion of having multiple datasets in different folders on the host. You can see how other wrappers use this, typically setting the variable `$audio_dir` to /vagrant/data. For the rest of this explanation, we'll be referring to this folder as DATAFOLDER.

- The wrapper should process all .wav files inside DATAFOLDER and, optionally, associated annotation files, which are in rttm format. (For more information on the rttm output, read NIST's 2009 eval plan)
- Your wrapper should support processing long sound files. If you have no better way of achieving this, look to utils/chunk.sh as an example; it breaks up long files into smaller 5 minute chunks, then iteratively calls a tool (in this case, yours), and concatenates the results.
- Your tool must process many sound files. This may require some optimization, for example loading very large model files into memory for each sound file is less optimal than loading once, then iterating over many files.
- The wrapper should write its *main output* into DATAFOLDER. Typically, this will be one annotation file for each .wav file. If your tool does VAD, SAD, or diarization, this annotation file should respect the rttm format. Additionally, the output file it should be named according to the pattern: toolname_file_original_name.rttm. If your tool does something other than VAD, SAD, or diarization, use a format that seems reasonable to you. If your tool returns results on individual files (and not e.g., summary statistics over multiple files), we still ask you to name the file toolnameStage_file_original_name.ext – where "ext" is any extension that is reasonable for the annotation format you have chosen.
- You probably also generate two types of ancillary files: Intermediary representation files, such as features extracted from the wav files; and log files, with detailed information of what was done and how. Both should be initially stored in the DATAFOLDER/temp/TOOLNAME folder.
- Intermediary representation files should be deleted – with one exception, introduced below.
- It is a good idea to print out a short log accompanying each batch of files analyzed, stating the key parameters, version number, and preferred citation. You may want to write out a lot more information. However, our target user may not be technically advanced, and thus including long technical logs may do more to confuse than to help. Log files should also be deleted if they are large (>5MB) – with one exception, introduced next.
- Your wrapper should expect an optional argument "–keep-temp", which is optionally passed in final position. If the last argument to the wrapper is not "–keep-temp", then ancillary files should be deleted. Code snippet example:

```
KEEPTEMP=false
if [ $BASH_ARGV == "--keep-temp" ]; then
        KEEPTEMP=true
fi
...
if ! $KEEPTEMP; then
        rm -rf $MYTEMP
fi
```

See any script in `launcher/` for examples.

13.6 Document Your Tool

Add a documentation file to the `DiViMe/docs/` folder on your host, in markdown format, containing at least the following three pieces of information: A one-paragraph explanation of what your tool does A reference or citation that people using your tool must employ A short section explaining how to use it. This will typically include a description of input & output formats, which can be replaced with references to the Format section of the docs. You should also include an example command line of how to run the tool. Often, this will be `vagrant ssh 'toolname.sh data/`. Please include any other information that would be useful to users, such as what parameters are available, how to access them, a tiny example input and output, further technical information on how the tool was built, additional references for work using this tool or on which the tool was built, etc.

For an example, see the tocomboSad section in the tools documentation.

13.7 Create a Reproducible Test for Your Tool

DiViMe comes equipped with a test script that downloads a publicly available daylong audio file and transcript, which all tools within DiViMe can process (and many can be evaluated). In this section, we provide instructions for you to add your tool to the test.sh routine.

By default, all launchers are read-only to avoid accidental editing by newbie users. For the next step, you need to change file permissions so as to be able to edit test.sh. From the host machine, type into the terminal:

```
chmod +rw test.sh
```

Open the file test.sh, which is inside the launcher folder, and add a section testing your tool, modeling it on the other tests present there. Typically, you will need these lines:

Under "# Paths to Tools" add a line with the path to your tool, eg: TOOLNAMEDIR=$REPOS/TOOLNAME

b) Before "# test finished", add a section like the following:

Example for a Sad type tool:

```
echo "Testing TOOLNAME..."
cd $TOOLNAMEDIR
TESTDIR=$WORKDIR/TOOLNAME-test
rm -rf $TESTDIR; mkdir -p $TESTDIR
ln -fs $TEST_WAV $TESTDIR
$LAUNCHERS/toolnameStage.sh $DATADIR/TOOLNAME-test >$TESTDIR/TOOLNAME-test.log || {␣
↪echo "   TOOLNAME failed - dependencies"; FAILURES=true;}

if [ -s $TESTDIR/toolnameStage_$BASETEST.rttm ]; then
    echo "TOOLNAME passed the test."
else
    FAILURES=true
    echo "   TOOLNAME failed - no RTTM output"
fi
```

In the above example: `DATADIR` is predefined as the test 5 minute data folder `data/VanDam-Daylong/BN32` `TOOLNAME` is whatever human readable name you have given your tool`toolnameStage` is the pattern for the system/launcher name for your tool, for example `opensmileSadBASETEST` is the basename of a test input file e.g. `BN32_010007_test` for the 5 minute input file `BN32_010007_test.wav`

Example for a Diar type tool: ** todo: complete**

Example for an Add type tool: ** todo: complete**

Run test.sh. Only proceed to the next phase if your tool passes the test.

13.8 Check reproducibility of your version of the VM by reprovisioning

Throughout the steps above, you have modified `Vagrantfile/bootstrap.sh` to automatically install your code, any required packages, and any required dependencies. If you have been keeping your Vagrantfile/bootstrap.sh in a good state, you should be able to rebuild your version of the virtual machine from scratch.

If necessary, log out from the virtual machine with control+D. Then, from the host machine, run the following code to destroy, re-build, and re-run the test:

```
vagrant destroy
vagrant up
vagrant ssh -c "test.sh"
```

WARNING: any changes you made by hand when you were logged into the VM will be lost when you run `vagrant destroy`: to make sure they show up automatically with `vagrant up`, all such dependencies need to be automated (added to Vagrantfile/bootstrap.sh). If your tool passes the test in this condition, you are ready to integrate your tool to DiViMe for real.

13.9 Integrate Your Tool Into the Public Version of DiViMe

1. Fork the DiViMe repo
2. Feplace Vagrantfile/bootstrap.sh with your version of Vagrantfile/bootstrap.sh
3. Add in the docs/ your tool's doc
4. Add your wrapper to launcher/
5. Replace test.sh with your version containing an additional test case specific to your tool
6. Create a pull request to DiViMe requesting your additions be incoporated

CHAPTER 14

References

Our work builds directly on that of others. The main references for tools currently included and/or data currently used to perform tests are:

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CHAPTER 15

Indices and tables

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