# **Research Design and Pilot Study Report**

# 2. Research Design

## 2.1 Research goal

Evaluation the effectiveness and accuracy of phrase detection using speech recognition models.

## 2.2 Research gap

While previous previous Systematic Literature Review showed the evaluations of overall accuracy of speech recognition systems – focusing on metrics like Word Error Rate (WER) and Character Error Rate (CER) - there is limited research on the detection of specific phrases or keywords in audio, especially when these phrases are defined by user. Many apps require efficient identification of some phrases within audio with real-time performance demands, which can be important for example in voice-activated systems.

# 2.3 Research Questions

- 1. How to effectively detect user-defined phrases within audio recording?
- 2. What are the challenges in detecting user-defined phrases with different models of speech recognition?
- 3. How does the accuracy of user-defined phrases vary across models of different sizes (whisper tiny, base, small, medium, large)?
- 4. How do different types of phrases (example clear, noisy) affect the success rate of phrase detection?
- 5. How do emotions and accent in speaking affect the success rate of phrase detection?

## 2.4 Research Hypotheses

- 1. User-defined phrases can be effectively detected within audio recordings using modern speech recognition models.
- 2. The challenges in detecting user-defined phrases depend on the speech recognition model, where smaller models face bigger difficulties in detecting phrases in noisy recordings.
- 3. The accuracy of user-defined detection varies between different speech recognition models, larger models outperform smaller models.
- 4. The detection accuracy of user-defined phrases is affected by the quality of the audio, with clear audio having higher detection success.
- 5. The detection accuracy of user-defined phrases is affected by emotions of the speaker with neutral and calm way of speaking having higher detection success.

## 2.5 Research Subjects and Sample

## Research subjects - Models

- Whisper Tiny
- Whisper Base
- Whisper Small
- Whisper Medium
- Whisper Large

#### Sampling Method – Audio recordings (40, 10 for each category):

- 1. Clean speech (no background noise)
- 2. Noisy environment (some background noise mixed in)
- 3. Accent variation (non-native English)
- 4. Emotion variation (anxious, assertive, encouraging, happy, sad)

## Qualification Criteria:

Phrase Presence: contains exactly one occurence of user-defined phrase with manual annotation.

Length: Between 5 to 15 seconds.

Clarity: The phrase is easy to understand, doesn't have cut-outs, or loud pops.

# 2.6 Operationalization – Variables

Variable Type	Variable	Definition				
Independent	ASR Model	Studied Model: Whisper:				
		Tiny, Base, Small, Medium,				
		Large				
	Audio Condition	Condition of audio, four				
		types: Clean Speech,				
		Noisy Environment,				
		Accent Variation, Emotion				
		variation				
Dependent	Phrase Detection	Two measures:				
	Accuracy	True Positive Rate (TPR) =				
		(number of correctly				
		detected phrase				
		instances) / (total phrase				
		instances)				
		False Positive Rate (FPR) =				
		(number of detections				
		outside the ground-truth				
		timestamps) / (total non-				
		phrase audio duration)				
Confounding	Speaker Gender	Male vs female, could				
		change model				
		performance				
	Phrase Position in Clip	Where the phrase				
		appears, later or earlier,				
		could affect model				
		performance				
Hidden	Microphone quality	Different microphone				
		settings can make audio				
		quieter or change how				
		clear certain sounds are,				
		but could affect model				
		performance				

## 2.7 Research Methods

Real-Life Experimental Evaluation: One user-defined phrase will be defined and manually marked in 200 real recordings, then run each Whisper's model to try and detect the phrase.

Controlled Simulation: Starting from a clean audio clip I will create new files by adding noise, speeding up, slowind down, and adding echo to the audio, then run each Whisper's model on these to try and detect the phrase.

Prototype Field Test: I will record myself saying phrases once in silence, once with some background noise and feed it into Whisper's model and try to detect the phrase.

## 2.8 Research Tools

1. Whisper Python Script

Python Script(s) running the OpenAl Whisper Library and gathering the results

2. Audio Annotation Tool

Online website audiomass.co for finding phrases and vscode to edit .csv file with annotations.

3. Audio Processing

Python Script(s) with librosa and soundfile libraries.

4. Audio Recording

Microphone and audio recorder (OBS) with some voice in the background from a computer.

# 2.9 Expected Results

### **Quantitative Expected Results:**

- Higher TPR is expected from larger Whisper models, compared to smaller ones.
- Lower FPR are expected in clean audio compared to noisy, accented or speedaltered audio.

## Qualitative Expected Results:

- Larger Whisper Models are expected to produce fewer misrecognitions.
- Smaller models may struggle with noisy, accented or speed-altered audio.

# 2.10 Validity Threats

**Internal Validity:** 

The position of the phrase in the audio might affect detection

Microphone quality could influence model performance

#### Mitigations:

Have phrase in many different positions within the audio

Keep audio recording setup consistent

## **External Validity:**

Testing only one phrase might not generalize

Recordings from only one speaker or accent might limit generalization

### Mitigations:

Include many different phrases to test with

Include variations of speakers and accents in audio

## Conclusion Validity:

Misintepreting false positives or undetected phrases

### Mitigations:

Use objective metrics (TPR, FPR)

#### 2.11 Research Plan

#### 1. Preparation and setup

- Install and configure whisper models with python
- Set up Python Scripts for phrase detection and result extraction
- Set up OBS for recording and VSCode for .csv editing

#### 2. Data Collection and Annotation

- Search for datasets with different audio files that have different audio conditions in them: Clean, noisy, accented, different emotions (200 of them, 50 each)
- Use tools to add noise, speed up, slow down and echo
- Record myself with and without background noise (20 recordings without and 20 with)

- Choose phrases for detection for these files
- Use annotation tool capable of showing audio waves with timings (audiomass.co) to find the phrase and add the data to .csv in code editor (vscode).

#### 3. Model Evaluation

- Run each whisper model on all of these audio samples
- Collect and organize output from these runs, identify true positive and false positives.

### 4. Analysis and interpretation

- Calculate TPR and FPR for each model and audio conditions.
- Compare performance across models and conditions.

#### 2.12 Publication Goals

#### What to publish:

- Phrase detection accuracies (TPR and FPR) for different Whisper model sizes
- Comparison tables showing detection accuracy across the four audio conditions

#### Where to publish:

- IEEE journals
- ACM Digital Library
- ScienceDirect
- ArXiv

# 3. Pilot Study

## 3.1 Research subjects

For pilot study, limited subset of research subjects was selected

#### Models:

- Whisper Tiny
- Whisper Large

#### **Dataset Audio Files:**

• Clean speech: 3 audio files

• Noisy environment: 3 audio files

Accent variation: 3 audio files

• Emotions variation: 3 audio files

#### Self-recorded audio:

without background noise: 2 recordings

• with background noise: 2 recordings

#### Each file above will have one modification done to it:

Added noise: 16 files in total
Slowed down: 16 files in total
Speed up: 16 files in total
Added Echo: 16 files in total

#### Target phrases:

Cheese, reach, bags, worried, likes, try

#### Criteria:

- o Each recording contains one occurence of the phrase.
- Each recording is between 5 to 10 seconds in length.
- o The phrase is clearly spoken in all recordings.
- o Manual annotation was performed to mark the timestamp of phrases in audio.

# 3.2 Study Execution

### First i downloaded datasets for all 4 categories:

- Noisy and clean sets https://datashare.ed.ac.uk/handle/10283/2791
- Accent set <a href="https://www.kaggle.com/datasets/rtatman/speech-accent-archive?resource=download">https://www.kaggle.com/datasets/rtatman/speech-accent-archive?resource=download</a>
- Emotion set <a href="https://www.kaggle.com/datasets/tli725/jl-corpus">https://www.kaggle.com/datasets/tli725/jl-corpus</a>

After downloading these I selected some of the files for each category as described in 3.1.

I recorded myself using OBS with and without background noise, with 2 different texts, 2 with added background noise, 2 clean, resulting in 4 files in total.

Then for all of these audio files I made additional 4 files, each with one modification, using a simple python script that uses librosa and soundfile libraries to add effects.

```
import librosa
import librosa.effects as effects
import soundfile as sf
import numpy as np
def add_noise(audio_path, output_path, noise_level=0.05): 1usage
   y, sr = librosa.load(audio_path)
   noise = np.random.normal( loc: 0, noise_level, y.shape)
   y_noisy = y + noise
   sf.write(output_path, y_noisy, sr)
def add_echo(audio_path, output_path, delay=0.1, decay=0.3): 1usage
   y, sr = librosa.load(audio_path)
   delay_samples = int(delay * sr)
   echo = np.zeros_like(y)
   echo[delay_samples:] = y[:-delay_samples] * decay
   y_echo = y + echo
    sf.write(output_path, y_echo, sr)
def speed_up(audio_path, output_path, rate=1.5): 1usage
   y, sr = librosa.load(audio_path)
   y_fast = effects.time_stretch(y, rate=rate)
   sf.write(output_path, y_fast, sr)
def slow_down(audio_path, output_path, rate=0.75): 1 usage
   y, sr = librosa.load(audio_path)
   y_slow = effects.time_stretch(y, rate=rate)
   sf.write(output_path, y_slow, sr)
path = "D:/SoundFilter/pilot_study/noisy_testset_wav/p257_023.wav"
filename = os.path.basename(path)
name_without_ext = os.path.splitext(filename)[0]
```

Then i annotated selected phrase for each file in visual studio code by editing .csv file. For checking the timings for phrases I used website called: audiomass.co.

It resulted in the following .csv file:

	column 1	column 2	column 3	column 4
1	audio_file	phrase	start_time	end_time
2	afrikaans1.mp3	cheese	7.9	8.3
3	french1.mp3	cheese	11.3	11.8
4	german1.mp3	cheese	10.8	11.2
5	afrikaans1_slow.wav	cheese	10.7	11.2
6	french1_slow.wav	cheese	15.1	15.8
7	german1_slow.wav	cheese	14.5	15.2
8	afrikaans1_echo.wav	cheese	8.02	8.4
9	french1_echo	cheese	11.3	11.7
10	german1_echo.wav	cheese	10.8	11.3
11	afrikaans1_noise.wav	cheese	7.9	8.3
12	french1_noise.wav	cheese	11.3	11.8
13	german1_noise.wav	cheese	10.8	11.3
14	afrikaans1_fast.wav	cheese	5.3	5.7
15	french1_fast.wav	cheese	7.7	8.1
16	german1_fast.wav	cheese	7.4	7.8
17	p232_003_c.wav	cheese	4.348	4.8
18	p232_005_c.wav	bags	2.5	3.0
19	p232_011_c.wav	reach	2.075	2.5
20	p232_003_echo_c.wav	cheese	4.348	4.8
21	p232_005_echo_c.wav	bags	2.5	3.0
22	p232_011_echo_c.wav	reach	2.075	2.5
23	p232_003_noise_c.wav	cheese	4.348	4.8
24	p232_005_noise_c.wav	bags	2.5	3.0
25	p232_011_noise_c.wav	reach	2.075	2.5
26	p232_003_slow_c.wav	cheese	5.745	6.5
27	p232_005_slow_c.wav	bags	3.4	4.08
28	p232 011 slow c.wav	reach	2.75	3.3

Of manually labeled timings of chosen phrases.

After labeling the files i made a script that uses Whisper's tiny and large models to try and find target phrases and save the results in separate .csv file.

#### Part of console output of running the script:

```
Processing 1-noise.mp4 with large model...

Transcript: I bought some fresh cheese at the market yesterday, it tastes amazing with crackers and a glass of wine.

Phrase 'cheese' in 1-noise.mp4: 1 detections

Processing 1-noise_echo.wav with large model...

Transcript: I bought some fresh cheese at the market yesterday, it tastes amazing with crackers and a glass of wine.

Phrase 'cheese' in 1-noise_echo.wav: 1 detections

Processing 1-noise_fast.wav with large model...

Transcript: Thank you so much, Jason. We hope you have a nice and easy progress in the next one.

Phrase 'cheese' in 1-noise_fast.wav: 0 detections

Processing 1-noise_noise.wav with large model...

Transcript: I note some bad things that were parted yesterday in place of the nation's progress in the last moment.

Phrase 'cheese' in 1-noise_noise.wav: 0 detections

Processing 1-noise_slow.wav with large model...

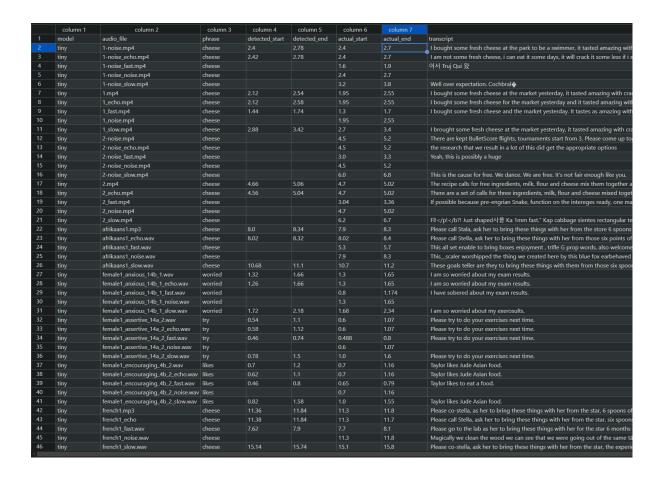
Transcript: I love some french cheese that I hought yesterday. It tastes just amazing. My crackers and my glass of wine.
```

## 3.3 Results

The resulting .csv files from trying to simply detect the phrase with Whisper models Tiny and Large:

	column 1	column 2	column 3	column 4	column 5	column 6	column 7
	model	audio_file	phrase	detected_start	detected_end	actual_start	actual_end
2	tiny	1-noise.mp4	cheese	2.4	2.78	2.4	2.7
3	tiny	1-noise_echo.mp4	cheese	2.4	2.8	2.4	2.7
4	tiny	1-noise_fast.mp4	cheese			1.6	1.9
5	tiny	1-noise_noise.mp4	cheese			2.4	2.7
6	tiny	1-noise_slow.mp4	cheese			3.2	3.8
7	tiny	1.mp4	cheese	2.12	2.54	1.95	2.55
8	tiny	1_echo.mp4	cheese	2.12	2.58	1.95	2.55
9	tiny	1_fast.mp4	cheese	1.44	1.74	1.3	1.7
10	tiny	1_noise.mp4	cheese			1.95	2.55
11	tiny	1_slow.mp4	cheese	2.88	3.42	2.7	3.4
12	tiny	2-noise.mp4	cheese			4.5	5.2
13	tiny	2-noise_echo.mp4	cheese			4.5	5.2
14	tiny	2-noise_fast.mp4	cheese			3.0	3.3
15	tiny	2-noise_noise.mp4	cheese			4.5	5.2
16	tiny	2-noise_slow.mp4	cheese			6.0	6.8
17	tiny	2.mp4	cheese	4.66	5.06	4.7	5.02
18	tiny	2_echo.mp4	cheese	4.56	5.04	4.7	5.02
19	tiny	2_fast.mp4	cheese			3.04	3.36
20	tiny	2_noise.mp4	cheese			4.7	5.02
21	tiny	2_slow.mp4	cheese			6.2	6.7
22	tiny	afrikaans1.mp3	cheese	8.0	8.34	7.9	8.3
23	tiny	afrikaans1_echo.wav	cheese	8.02	8.32	8.02	8.4
24	tiny	afrikaans1_fast.wav	cheese			5.3	5.7
25	tiny	afrikaans1_noise.wav	cheese			7.9	8.3
26	tiny	afrikaans1_slow.wav	cheese	10.68	11.1	10.7	11.2
27	tiny	female1_anxious_14b_1.wav	worried	1.32	1.66	1.3	1.65
28	tiny	female1 anxious 14b 1 echo.wav	worried	1.26	1.66	1.3	1.65

And additionally a separate .csv file that includes transcripts:



With these .csv files I calculated the metrics and I will display them with .csv files and diagrams.

## **Metrics Calculation**

#### False Positive:

We will calculate how much of overlap there is between the detected phrase and the actual phrase. For example if the phrase "cheese" appears at 2.92 - 3.48 seconds and the actual phrase occurs at 3.2 - 3.8 seconds, then the overlap is 0.28 (3.48 - 3.2), if we assume the threshold of **50%**, 0.28 is less than 50% of 0.6, so it's a **False Positive**.

The threshold was set at: 15%

True Positive:

The overlap has to be greater or equal to 15% of actual phrase duration.

Note: True Positive and False Positive are both **False** if phrase was not detected by model at all.

FPR & TPR (False Positive Rate & True Positive Rate):

TPR = True positives / Total Phrases

FPR = False positives / Total Phrases

Phrase position classification:

If phrase is before 3 seconds it's marked as **beginning**, less than 7 seconds is **middle** and else is **end**.

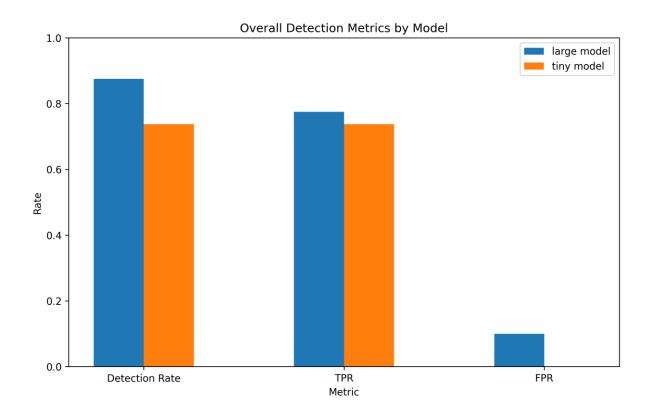
#### **Overall Results:**

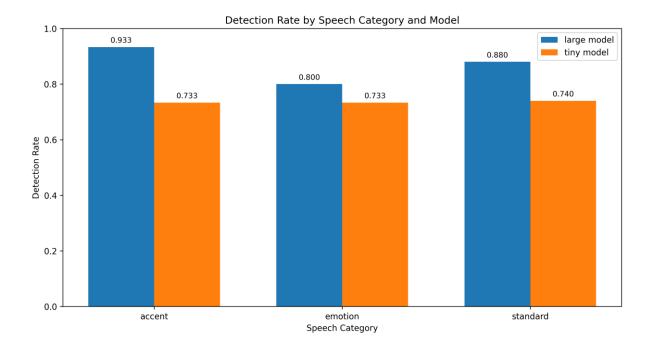


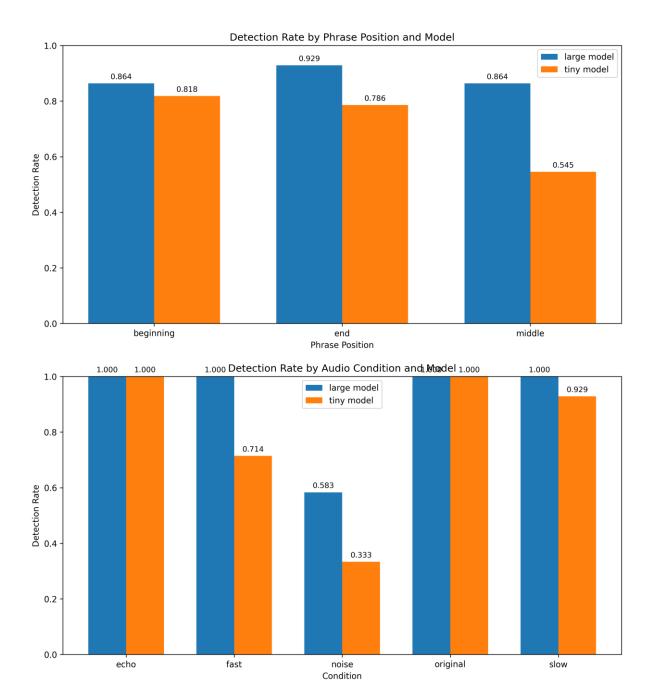
#### **Detailed Results:**

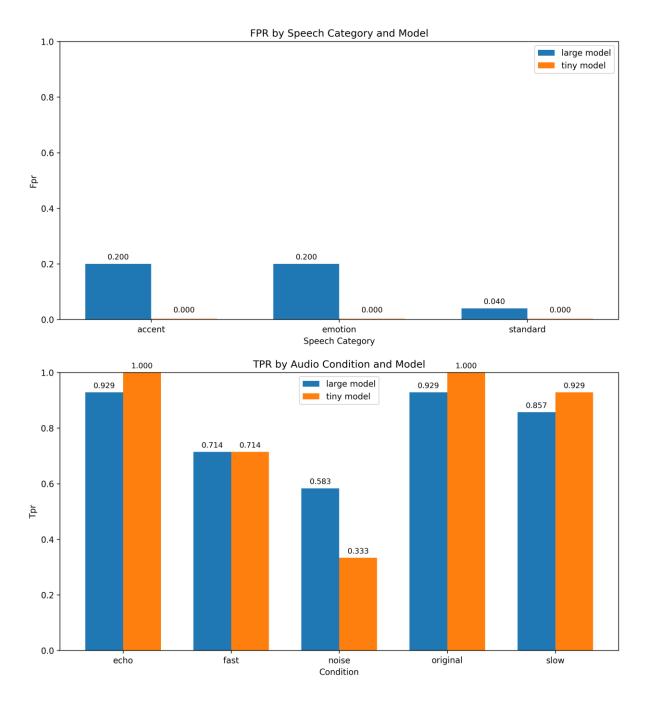
	column 1	column 2	column 3	column 4	column 5	column 6		column 8	column 9	column 10	column 11	column 12	column 13
1	model	audio_file	phrase	detected_start	detected_end	actual_start	actual_end	condition	speech_category	phrase_position	true_positive	false_positive	detection_made
2		1-noise.mp4	cheese						standard	beginning		False	True
3	tiny	1-noise_echo.mp4	cheese					noise	standard	beginning	True	False	True
4		1-noise_fast.mp4	cheese						standard	beginning	False	False	False
5	tiny	1-noise_noise.mp4	cheese					noise	standard	beginning	False	False	False
6			cheese						standard	middle	False	False	False
7	tiny	1.mp4	cheese					original	standard	beginning		False	True
8		1_echo.mp4	cheese						standard	beginning		False	True
9		1_fast.mp4	cheese					fast	standard	beginning		False	True
10		1_noise.mp4	cheese						standard	beginning	False	False	False
11	tiny	1_slow.mp4	cheese					slow	standard	beginning		False	True
12		2-noise.mp4	cheese						standard	middle			False
13	tiny	2-noise_echo.mp4	cheese					noise	standard	middle	False	False	False
14		2-noise_fast.mp4	cheese						standard	middle	False	False	False
15	tiny	2-noise_noise.mp4	cheese						standard	middle	False	False	False
16		2-noise_slow.mp4	cheese						standard	middle		False	False
17		2.mp4	cheese					original	standard	middle	True	False	True
18		2_echo.mp4	cheese						standard	middle			True
19	tiny	2_fast.mp4	cheese					fast	standard	middle	False	False	False
20	tiny	2_noise.mp4	cheese						standard	middle	False	False	False
21	tiny	2_slow.mp4	cheese					slow	standard	middle	False	False	False
22	tiny	afrikaans1.mp3	cheese					original	accent	end	True	False	True
23	tiny	afrikaans1_echo.wav	cheese	8.02	8.32	8.02	8.4	echo	accent	end	True	False	True
24	tiny	afrikaans1_fast.wav	cheese					fast	accent	middle	False	False	False
25	tiny	afrikaans1_noise.wav	cheese					noise	accent	end	False	False	False
26	tiny	afrikaans1_slow.wav	cheese	10.68				slow	accent	end	True	False	True
27	tiny	female1_anxious_14b_1.wav	worried		1.66		1.65	original	emotion	beginning	True	False	True
28	tiny	female1_anxious_14b_1_echo.wav	worried					echo	emotion	beginning	True	False	True
29	tiny	female1_anxious_14b_1_fast.wav	worried			0.8		fast	emotion	beginning	False	False	False
30	tiny	female1_anxious_14b_1_noise.wav	worried					noise	emotion	beginning	False	False	False
31	tiny	female1_anxious_14b_1_slow.wav	worried		2.18	1.68	2.34	slow	emotion	beginning	True	False	True
32	tiny	female1_assertive_14a_2.wav						original	emotion	beginning	True	False	True
33	tiny	female1_assertive_14a_2_echo.wav	try	0.58		0.6		echo	emotion	beginning	True	False	True
34	tiny	female1_assertive_14a_2_fast.wav		0.46		0.488		fast		beginning			True
35	tiny	female1_assertive_14a_2_noise.wav	try			0.6		noise	emotion	beginning	False	False	False
36	tiny	female1_assertive_14a_2_slow.wav						slow	emotion	beginning		False	True
37	tiny	female1_encouraging_4b_2.wav	likes				1.16	original	emotion	beginning	True	False	True
38	tiny	female1_encouraging_4b_2_echo.wav	likes					echo		beginning		False	True
39	tiny	female1_encouraging_4b_2_fast.wav	likes	0.46	0.8	0.65		fast	emotion	beginning	True	False	True
40		female1_encouraging_4b_2_noise.wav	likes						emotion	beginning	False	False	False
41	tiny	female1 encouraging 4h 2 slow way	likes	0.82	1 58	10	155	slow	emotion	heainnina	True	False	True

# Overall Metrics Comparison & Other Diagrams:









## **Quality Evaluation**

FPR is misleading – Tiny model appears to have perfect precision, but since it doesn't make any detections at all then the FPR of it is zero, since it made no attempt at detecting anything.

A lot of the results are same for both models, being 1.0, which shows that low sample of data in some cases gives less accurate results.

## 3.4 Conclusions

Both whisper tiny and large models proved to be capable of detecting phrases well. Manual annotation using audiomass.co and CSV editing in VSCode was effective for creating data.

Python scripts effectively edited audio files, calculated metrics and displayed diagrams.

#### 4. Conclusions

## 4.1 Design

The design worked well when used on smaller scale with pilot study to test how speech models detect phrases in different audio types. We learned that comparing tiny and large models shows differences in their performance. We learned that more audio samples are need to get more accurate results.

## 4.2 Pilot Study

Pilot study showed that Large Whisper model can perform better when it comes to detecting phrases than tiny model. We learned that testing approach of different audio types (clean, noisy, with different accent, emotions) works well to test model performance. Manual annotation was time-consuming, but python scripts saved a lot of time.

## 5. Literature

- Noisy and clean sets <a href="https://datashare.ed.ac.uk/handle/10283/2791">https://datashare.ed.ac.uk/handle/10283/2791</a>
- Accent set <a href="https://www.kaggle.com/datasets/rtatman/speech-accent-archive?resource=download">https://www.kaggle.com/datasets/rtatman/speech-accent-archive?resource=download</a>
- Emotion set <a href="https://www.kaggle.com/datasets/tli725/jl-corpus">https://www.kaggle.com/datasets/tli725/jl-corpus</a>
- https://arxiv.org/abs/2305.13516 Scaling speech technology to 1000+ languages
- https://www.sciencedirect.com/science/article/pii/S1877050922014338 Evaluation of the fficiency of state-of-the-art Speech Recognition engines
- https://ieeexplore.ieee.org/document/10348186 A study of Audio-to-Text Conversion Software Using Whispers Model
- <a href="https://ieeexplore.ieee.org/document/10721926">https://ieeexplore.ieee.org/document/10721926</a> WhisperSum: Unified Audioto-Text Summarization

- https://ieeexplore.ieee.org/document/10544133 Speech Recognition
   Paradigms: A comparative Evaluation Of SpeechBrain, Whisper and Wav2Vec2
   Models
- https://ieeexplore.ieee.org/document/10800465 Evaluating Automatic
   Transcription Models Utilising Cloud Platforms
- https://ieeexplore-1ieee-1org-10000076o0031.han.bg.pg.edu.pl/stamp/stamp.jsp?tp=&arnumber=10493971 -Automatic and Multilingual Speech Recognition and Translation by using Google Cloud API
- https://arxiv.org/abs/2006.11477 Wav2vec 2.0: A framework for Self-Supervised Learning of Speech Representations
- <a href="https://ieeexplore.ieee.org/document/1318504">https://ieeexplore.ieee.org/document/1318504</a> Is word error rate a good indicator for spoken language understanding accuracy