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## NFI-FRITS: A forensic speaker recognition database and some first experiments

David van der Vloed<sup>1</sup>, Jos Bouten and David A. van Leeuwen<sup>1,2</sup>

<sup>1</sup> Netherlands Forensic Institute, <sup>2</sup> Radboud University Nijmegen  
{d.van.der.vloed,d.van.leeuwen}@nfi.minvenj.nl

### Abstract

In this paper we describe the collection of a speech database with forensically realistic data. It consists of speech material obtained from lawfully intercepted telephone conversations collected during police investigations. The speech material therefore is very similar to the kind we encounter in casework at the Netherlands Forensic Institute. The database is augmented with metadata describing language, accent, speaking style and acoustic conditions. A total of 604 speakers have been identified in 4188 conversation sides. After manual speaker attribution using various forms of available metadata, the speech content has been anonymised by zeroing out fragments that might disclose the real identity of speakers. Additional to the database description, this paper reports on some speaker recognition experiments using a commercially available forensic speaker recognition system. We can observe some effect of spoken language in terms of calibration, but overall the systems appears not too sensitive to accent or language.

### 1. Introduction

In this paper the Netherlands Forensic Institute's Forensically Realistic Intercepted Telephone Speech database (NFI-FRITS) is presented, a database originating from recordings of telephone speech intercepted by Dutch law officials. It is similar in set-up to the AHUMADA 3 speech database [1] recorded in Spain by the Guardia Civil. The purpose of collecting the database is to obtain experience with automatic speaker recognition systems in a forensic setting, with the ultimate goal of supporting evidence reporting in forensic speaker comparison cases in a Bayesian framework. The intention is to share the database with public institutions; for more information the reader can contact the first author.

One of the conditions set by the law officials for using this material is that the finalised database has been anonymised, meaning that the metadata and audio can not contain any information that can disclose the real identity of a particular speaker. Therefore human annotators listened to all the recorded material and zeroed the audio fragments that contained such information. The annotators further provided some basic metadata as well, the single most important of which was (an encoded) speaker identity. The database contains forensically realistic material, i.e., audio data from real intercepted telephone speech, originating from real police investigations. It is thus of the type that is frequently encountered in NFI casework, but not originating

from that source.

In forensic speaker comparison the general problem is to compare speech from a trace, typically a recording of speech containing incriminating evidence, to that of a suspect in terms of the identity of the speaker. In what is sometimes called the *paradigm shift* [2,3] the aim is to present the result of the comparison in a probabilistic way, and in analogy to the way DNA evidence is reported in court, the goal is to present the speaker comparison in terms of a *likelihood ratio*

$$r = \frac{P(\text{speech} | H_p, I)}{P(\text{speech} | H_d, I)} \quad (1)$$

where  $H_p$  is the *prosecutor's hypothesis*, stating that the perpetrator and suspect are the same person, and  $H_d$  is the *defence's hypothesis*, stating that the perpetrator is not the same person as the suspect.<sup>1</sup> Important to note is the specification of other information, or circumstances, in the case, specified as  $I$  that are conditioning the probabilities in the numerator and denominator in exactly the same way. Thus, the likelihood ratio only considers the likelihood of the hypotheses, and nothing else. The 'speech' in (1) represents all available speech material relevant to the case, i.e., both the suspect reference recording and the questioned recording (the trace). The log of the likelihood ratio

$$\ell = \log r \quad (2)$$

is known as the 'weight of evidence' [4] and has a long history dating back to Turing in 1941. The log likelihood ratio has the nice property that different, independent, pieces of evidence are additive, and in this sense can be metaphorically viewed as items that can be put on the weighing scales of the Goddess of Justice [4].

In recent years a growing understanding of a procedure for producing likelihood ratios in case work has emerged. Originally, attempts were made to estimate the probability density functions (PDFs) similar to numerator and denominator in (1) [5,6], but later it was realised that the comparison score of an automatic speaker recognition system can be used directly in a score-to-likelihood ratio function that is learned using empirical data [2, 7-14], a process known as *calibration*. Either way, the circumstances of the case  $I$  should be reflected in the data used to either directly estimate the PDFs or the empirical data that is used in the calibration process. If there are circumstances  $I$  for which it is linguistically or acoustically plausible that they influence the comparison value of a speaker recognition system, then the same linguistic or acoustic conditions should be applied to the data used for calibration. Well designed and popular databases used in automatic speaker recognition research

<sup>1</sup>The defence hypothesis can be more specific than that, as long as it is opposing  $H_p$ .

Table 1: Metadata provided with the raw recording data. ‘Date-time’ is a format specified as YYYY-MM-DD HH:MM:SS which can be sorted lexicographically.

Field	Format
Filename	salted hash
Case	ID in range 1–13
Calling telephone number	salted hash
Called telephone number	salted hash
IMEI intercept target	salted hash
IMSI intercept target	salted hash
Start date time	Datetime
End date time	Datetime
Total duration in seconds	number
Intercept target is calling in / out	in / out

such as Switchboard [15, 16] and Mixer [17] do not cover the same circumstances as found in forensic case material, in terms of language, speaking style, quality, etc. In forensic speech databases such as NFI-FRITS, AHUMADA [1], GFS1.0 corpus [18] and NFI-TNO [19] these circumstances are covered to some extent, and by providing metadata it is possible to make a selection representing  $I$  in (1).

After presenting a description of the NFI-FRITS database in Section 2, this paper reports some experiments on a smaller sub-set of the database containing Turkish speakers in Section 3. This allows us to study circumstances  $I$  in Section 4 where the trace contains speech in this language, and we can even study cases where the suspect reference recording is spoken in a different language: Dutch, but with a Turkish accent.

## 2. Description of Database

### 2.1. Speech material

The NFI-FRITS (recordings of telephone speech intercepted during real police investigations) consists of Dutch, Moroccan Arabic, Berber and Turkish speech material. This material was obtained to facilitate research on data typically encountered in forensic practice, much like the data used by [1, 18, 19]. To comply with legal regulations, the data was anonymised by zeroing speech that contained information traceable to an individual. Annotators native in the relevant languages were hired to perform this task. The NFI-FRITS consists of 4188 speech files with speech from 604 different speakers.

### 2.2. Data processing

The raw data were provided as 2-channel, 8 kHz 8 bit A-Law encoded waveform files and in the majority of recordings both ends of the conversation are recorded in separate channels. These files came with some metadata, such as the two telephone numbers involved in the telephone call, IMEI numbers and case names. These can be found in Table 1.

The audio files were first split in two single channel files (a and b), expanded to 8 kHz, 16 bit PCM and stored in a database, along with the provided metadata. The native human annotators listened to all recordings and identified speaker names (of the type ‘John,’ ‘John’s girlfriend,’ ‘guy from the pizza place’), determined a gender for the speaker (male/female) and then assigned single channel files from the database to that speaker. In this process, information like the telephone number, the case, and the spoken content of the speech file was taken into ac-

Table 2: Metadata provided by the annotators

Field	Format/value
Speaker ID	6 digit number
Language ID	ID in range 1–12, see Table 4
Language proficiency	native, good, poor
Loud sounds	yes/no
Background music	No / short / long
Audible noise	No / weak / intermediate / strong
Formal conversation	Yes / No
Age group	Adult / very young / very old
Conversation type	Dialogue / monologue
Background speakers	Yes / No
In motorised vehicle	Yes / No
Strong reverberation	Yes / No
Emotional speech	No / short / long
Whisper	No / short / long
Raised voice	No / short / long
Regional speech	No / Yes
Remarks	Plain text

count. Subsequently, the audio was listened to again and all the speech that contained information that could possibly identify an individual was removed by setting all the digital sample values to zero in these fragments within the audio file. Additionally, other metadata was added by the annotators, only based on their listening, like spoken language, language proficiency, a broad indication of the age of the speaker and so forth. A complete list can be found in Table 2. This process was carried out until about five recordings were assigned to an individual speaker, after which a new speaker was selected and the process was repeated.

In total, 604 different speakers were identified (117 female / 427 male) and 4188 conversation sides were assigned to these speakers (1068 female / 3120 male).

After this phase all the data were listened to again by a forensic speech scientist. The speaker assignment of the recordings was verified by listening. During this operation background sounds were marked for exclusion by marking all times of intervals to be excluded. Thus, call tones, periodic background sounds, background speakers, crosstalk, messages from the mobile phone operator, etc. can be automatically excluded. The time interval marking makes it possible to generate a version of the database that contains audio that resembles edits as typically used in case work. However, due to the large number of files and the limited available resources, it is possible that the end result still contains some audio that would have been edited out in case work.

The last step was to anonymise the metadata, by deleting or by replacing it with salted hashes or ID-numbers. The metadata replaced by hashes are: ‘calling telephone number,’ ‘called telephone number,’ ‘IMEI’ and ‘IMSI.’ The metadatum that was replaced by an ID is ‘case name.’ Speaker names were deleted. Some new metadata were generated, which can be found under in Table 3.

### 2.3. Speaker identities

Due to the origin of the data there is no absolute certainty about speaker identities assigned by the annotators. However, they based their choice on names and other information about the speakers they gathered by listening, layman speaker recognition and the fact that two speech samples coming from the same

Table 3: Semi-automatically generated metadata

Field	Format / value
File ID	Number
Annotator	Number (range 1–5)
Annotation date and time	Datetime
SNR in dB	number
Net duration (seconds)	number
Duration marked	
background sounds (seconds)	number

telephone number are very likely to be from the same speaker, especially if it is a mobile phone number. The annotators were instructed to ignore the audiofiles about which they had doubts regarding speaker identity. Furthermore, the speaker identity was verified by another annotator who listened to the material, and when there was doubt about speaker identity, the audio file was discarded. Speaker identity is thus no absolute truth, but rather a truth by proxy.

The speakers are all people that were recorded in lawfully intercepted telephone calls. It is likely that a speaker was not aware that he or she was recorded at the time of speaking, although since precise circumstances of the investigations are unknown, and given the reason for interception and the potentially criminal activities of the speakers, this is not certain.

## 2.4. Recordings

### 2.4.1. Number of recordings per speaker

The aim of the database was to collect five recordings per speaker. However, some speakers in the database have more recordings associated with them, because initially no aim was set. Because of concerns that the total number of speakers in the recordings would become too low within the conditions of the allotted resources for producing the database, the aim was first set at ten, and later that number was lowered to five. Conversely, some speakers have less than five recordings assigned to them. This is because, during the annotation procedure, there sometimes was no new case material available and hence the annotators were instructed to continue with the case, aiming for speakers with three or four recordings. Finally, the annotator verifying the material could also remove recordings assigned to a speaker.

### 2.4.2. Duration of recordings

The gross duration of the recordings was stored with the recordings, defined as the entire duration of the provided audio file. The minimum gross duration was set at 30 seconds and the maximum duration was set at 600 seconds. This maximum duration was determined in the course of the project so that the annotators would not spend too much time and effort on a single recording.

In total 165 hours of speech has been annotated in 4188 segments. The histogram of duration is shown in Figure 1, the mean segment duration after automatic energy-based speech activity detection is 142 seconds. On average, 3.85 s of speech (2.7 %) was nulled as a result of the anonymisation procedure.

### 2.4.3. Languages

The languages included in this database are Dutch (nld), Moroccan Arabic (ary), Tarifit (rif) and Turkish (tur). These languages

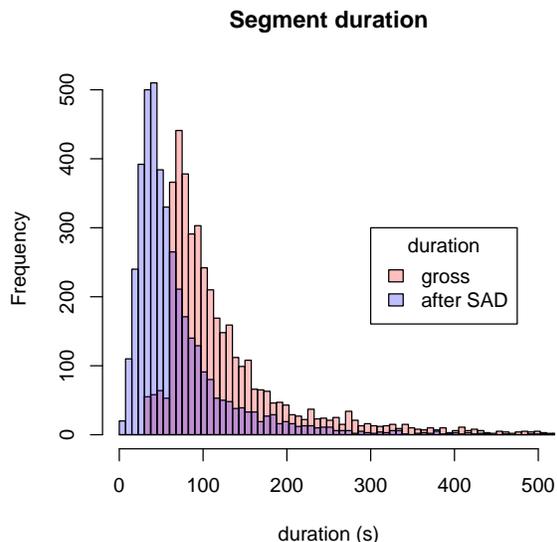


Figure 1: A histogram of a segment duration before and after automatic speech activity detection (SAD).

were chosen as the target languages because these are the languages that occur most frequently in case work at the NFI. Not unrelated, these are the languages that are most frequent in the provided material. Attempts to gather recordings in other languages would have become very difficult, since this would involve a lot of effort spent on just trying to find the recordings in those languages.

The actual values in the language field in the database are a bit more complex, so that the language field would be a bit more informative about the speaker and his or her background. The possible values are given in Table 4, together with the statistics of language proficiency as judged by the native annotators.

The ‘Dutch by an ⟨area⟩ speaker’ language fields were used for recordings containing ethnolectic Dutch. The Mix-languages were used when multiple languages were used by the same speaker in a single recording.

## 3. Experimental design

In order to give an impression of how this speech database can be used for forensic speaker recognition research we have conducted an experiment with an automatic speaker recognition system. The full database is quite heterogeneous in terms of speaker population and spoken language, so we decided to carry out a first characterisation experiment using a preliminary subset of the data.<sup>2</sup> From the entire database, we selected segments (i.e., conversation sides in NIST SRE 2006 parlance) in which either Turkish (“tur”), Dutch with a Turkish accent (“nld-t”) or a mix of Dutch and Turkish (“nld/tur”) was spoken. This selection resulted in 60 distinct speakers in 534 segments, approximately 10 % of NFI-FRITS. For each speaker, the available segments were divided in equal numbers over ‘train’ and ‘test’ segment classes, where in the case of an odd number of segments the ‘train’ segment class received one more than the ‘test’ class. This procedure resulted in 211 training segments (58 speakers)

<sup>2</sup>The final release of the database is expected later this year.

Table 4: Distribution of spoken languages and proficiency

Language — <i>abbreviation in this paper</i>	Language proficiency			Total
	native	good	poor	
Dutch	2416	62	5	2483
Turkish — <i>tur</i>	472	27	-	499
Moroccan Arabic	142	49	-	191
Dutch by Moroccan speaker	20	184	1	205
Berber (Tarifit)	116	-	-	116
Dutch by Turkish speaker — <i>nld-t</i>	60	296	10	366
Dutch by Caribbean speaker (Surinam/Dutch Antilles)	37	-	-	37
Mix of Arabic and Dutch	71	13	-	84
Mix of Arabic and Berber	3	1	-	4
Mix of Berber and Dutch	17	2	-	19
Mix of Dutch and Turkish — <i>nld/tur</i>	16	120	2	138
Other	27	13	6	46

and 323 test segments (59 speakers). The imbalance is because the speaker recognition system employed has default minimum requirements for training duration (30 sec) and signal-to-noise ratio (10 dB).

The automatic speaker recognition system computes scores (or likelihood ratios) for all train segments vs. all test segments, resulting in 68 153 trials. These can further be classified according to train/test spoken language according to Table 5.

In one operational condition the speaker recognition system can specify ‘reference population’ speech material for score normalisation and calibration purposes. For this, we used one segment for each of 44 speakers chosen outside the test database, roughly equally distributed over the three language conditions.

### 3.1. Speaker recognizer conditions

We used a commercially available automatic speaker recognition system in this research<sup>3</sup>, we therefore know little details about features and background training data. Its engine is a modern i-vector system with PLDA scoring, and can operate in two modes: it can just produce an uncalibrated PLDA score, or it can produce a calibrated likelihood ratio that can be used in forensic evidence reporting. For the latter, the system needs a collection of utterances from a reference population, which can be seen as representative speakers of the alternative hypothesis  $H_d$ .

### 3.2. Spoken language conditions

Using different cuts of the test set according to Table 5, we can investigate the influence of spoken language on a speaker recognition trial. We can analyse results per language/accent, or look across-language/accent effects.

### 3.3. Normalisation for distribution of trials over speakers

Even though the collection of segments per speaker was steered towards a fixed number of segments per speaker, the availability of the data in the various police investigation cases still skewed this distribution, see Figure 2. To compensate for the different amounts of trials available per speaker, we apply trial weighting in the performance analysis steps. The details of trial weighting are described in an earlier paper [20], where the influence of different amounts of trials for different conditions in NIST SRE-

2008 was equalised. If the number of trials involving speaker  $s_1$  and  $s_2$  in hypothesis  $H$  is denoted by  $N(s_1, s_2, H)$ , then the steps in the DET plot [21] become dependent on the speakers involved:

$$\Delta P_{\text{FA}}(s_1, s_2) = \frac{1}{N_s N(s_1, s_2, H_d)}, \quad (3)$$

$$\Delta P_{\text{miss}}(s) = \frac{1}{N_s N(s, s, H_p)}, \quad (4)$$

where  $N_s$  is the number of target speakers whose influence is equalized. Similar expressions can be deduced [20] for performance metrics such as  $C_{\text{llr}}$  and  $C_{\text{llr}}^{\text{min}}$  [7]. We used version 0.8 of the R library `sretools` for the trial weighting analysis [22]. We believe [23] applied the same idea to speakers, by ‘‘averaging DET plots ... for every target / non-target speaker pair,’’ which is mathematically equivalent. In our formulation, each trial is weighted inversely proportional to the contingency table of trials for the factors train speaker ID and test speaker ID.

## 4. Results

### 4.1. Equalization

In a first experiment we analyse the effect of the trial weighting. In Figure 3 we show the effect on the DET-plot. We can observe that for this data set the curve becomes more ragged, which is probably caused by speaker combinations with relatively few mutual trials being weighted quite heavily in the DET (cf. (3)–(4)). The contingency table, being a product of two similarly skewed speaker distributions as in Figure 2, is even more skewed, with frequencies ranging from 1 to 726. A second observation is that for this data set the equal error rate  $E_{=}$  is lower when speaker trial weighting is in effect. In the following, we will use speaker weighting in the analysis.

### 4.2. Reference population and calibration

In the next experiment, we compare the effect of the reference population to the recognition performance. In Figure 4 we compare the DET plot for the condition without reference population to that with reference population, where in the latter case we show both the ‘raw scores’ and the ‘likelihood ratio’ output. The two curves with a reference population are not identical, meaning that the calibration is a non-monotonic function of the scores. However, there is no qualitative difference in discrimination performance for the three curves.

<sup>3</sup>Agnitio Batvox Eval version 4

Table 5: Target / non-target trial counts for the various combinations of spoken language/accent in the experimental data set.

train \ test	tur	nld-t	nld/tur	total
tur	762 / 23304	332 / 12520	61 / 3719	1155 / 39543
nld-t	241 / 9500	480 / 4722	46 / 1484	767 / 15706
nld/tur	116 / 6378	82 / 3386	90 / 930	288 / 10694
total	1119 / 39182	894 / 20628	197 / 613	2210 / 65943

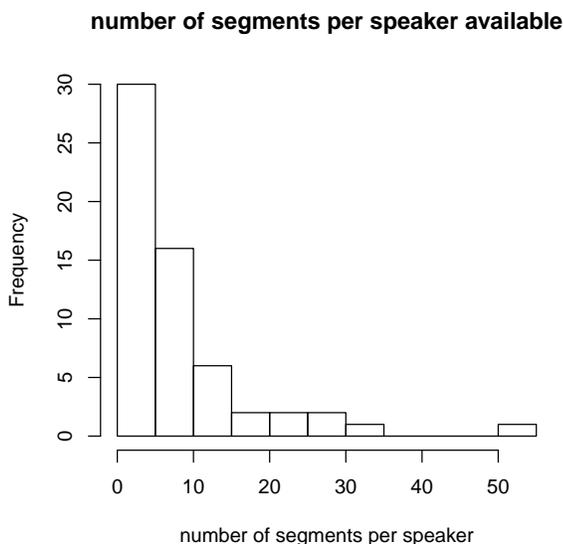


Figure 2: The distribution of the number of available segments per speaker, for the experimental selection of the database.

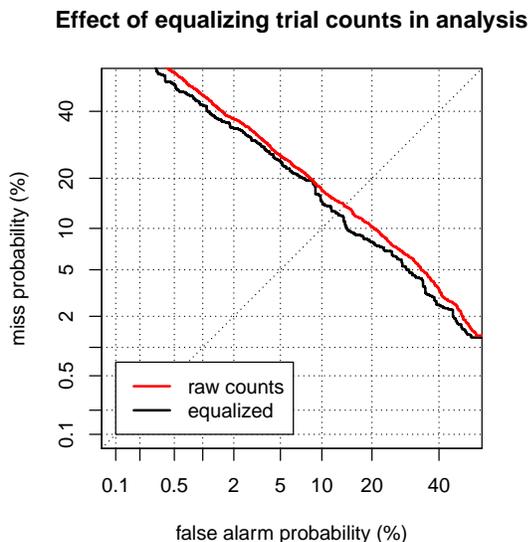


Figure 3: The effect of equalising the weight of different speakers, by compensating for their relative frequency. The recognizer is employed in the condition without reference population.

Comparison of recognizer condition

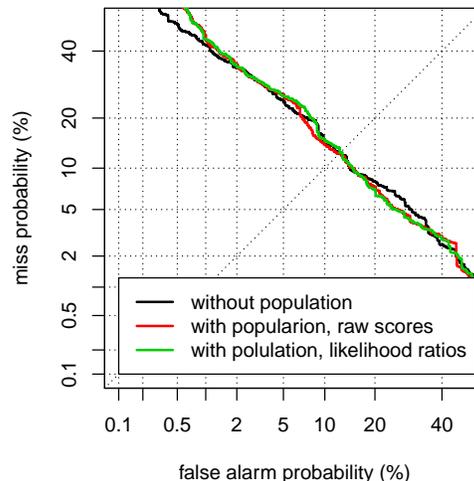


Figure 4: The effect of the condition of the speaker recognizer, with/without reference population, raw scores / log likelihood ratios.

If we consider not only discrimination but also calibration performance, there is quite a difference between the conditions. In Table 6 we illustrate the performance in terms of  $C_{llr}$ , a calibration-sensitive general performance metric for likelihood ratios [7]. In the first line in the table, it is not really fair to compare  $C_{llr}$  to  $C_{llr}^{\min}$ , because under this condition we take raw scores, which are not calibrated.  $C_{llr}$  measures calibration, essentially telling us to what extent the scores behave like calibrated log-likelihood-ratios, and we know that raw recognition scores—even when obtained by PLDA log likelihood ratio scoring—still need a calibration step. Specifying a reference population has a positive effect on  $C_{llr}$ , but the value is still quite far from the minimum attainable value  $C_{llr}^{\min}$  given the discrimination performance. Finally, the LLR values  $\ell$  (2) as produced by the recognizer result in a  $C_{llr}$  quite close to  $C_{llr}^{\min}$ , so we can conclude that the calibration is quite good. From now on we will work with the LLR scores from the recognizer condition where we used a reference population (cf, Section 3.1).

In Figure 5 the densities of  $\ell$  for target and non-target trials are shown. Essential to good calibration performance is that the log of the ratio of the red and blue curves is equal to the value on the  $x$ -axis, for all LLRs [24]. One can observe that where the lines cross, this is the case: this happens at  $\ell = 0$ . Cumulative densities are known as “Tippet plots” [5, 25] and are frequently shown in forensic science literature. These cumulative densities are essentially  $1 - P_{FA}$  and  $P_{miss}$ , and therefore contain the

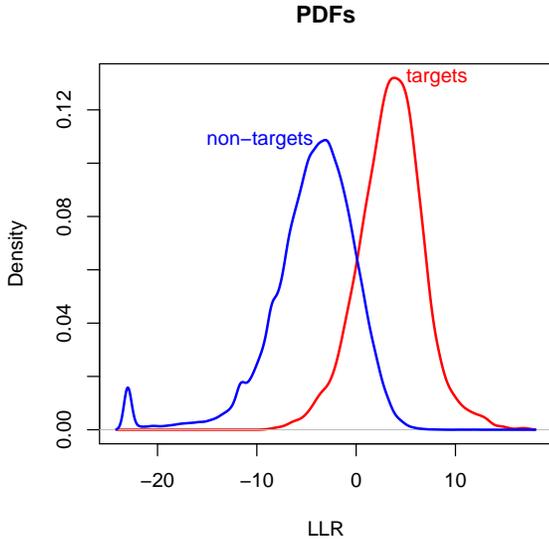


Figure 5: The probability density functions for target and non-target LLRs from the recognition system.

Table 6: Performance metrics for the three conditions shown in Figure 4. We used speaker equalisation for the analysis. RP: reference population, LLR: log likelihood ratios

Condition	score	$C_{llr}$	$C_{llr}^{\min}$	$E_{=}$
without RP	raw	1.202	0.407	12.2 %
with RP	raw	0.724	0.402	12.1 %
with RP	LLR	0.447	0.405	12.3 %

same information as a ROC or DET plot, except that in these plots the LLR values are implicit. Sometimes the proportion of non-target trials with  $\ell > 0$  is referred to as the ‘misleading evidence’ [26], but of course for calibrated  $\ell$  this is just the equal error rate  $E_{=}$ .

### 4.3. Effect of language

We can now investigate the effect of spoken language on the recognition performance. From Table 5 we can see that our experimental sub-set from the NFI-FRITS database allows for many different train/test combinations of language. It is important to choose the conditions the same, under which  $H_p$  and  $H_d$  trials are selected, when computing likelihood ratios. For instance, when comparing a suspect speaking Turkish (‘tur’) with a perpetrator speaking Dutch with a Turkish accent (‘nld-t’) this language combination should be considered as part of the circumstances  $I$  in the likelihood ratio (1). Hence for a meaningful experiment we should take both target and non-target trials using the same language conditioning.

#### 4.3.1. Effect of language/accents mix

First we condition on the language/accents mix, as shown in the top three lines in Table 7, corresponding to diagonal entries in Table 5. We might conclude that calibration ( $C_{llr} - C_{llr}^{\min}$ ) is more problematic for the Turkish-accented Dutch than for native Turkish speakers. The discrimination performance ( $C_{llr}^{\min}$ ,

$E_{=}$ ) seems to vary quite a bit, however, the number of speakers  $N_s$  and available target and non-target trials ( $N_{tar}$  and  $N_{non}$ ) aren’t very high and consequently DET curves are quite ragged.

#### 4.3.2. Cross language effects

Another condition worth studying is the effect of different languages spoken in train and test. In the fourth line of Table 7 we have tested segments spoken in Turkish vs. segments spoken in Dutch with a Turkish accent. Hereby we have pooled trials with Turkish as either train or test language. The discrimination performance ( $E_{=}$ ) is not really different from conditions with the same language for train and test (cf. lines 1 and 2 from the table), but the calibration ( $C_{llr} - C_{llr}^{\min}$ ) suffers. A fair comparison of this condition is perhaps to the performance obtained when the trials of lines 1 and 2 are pooled. We have shown this in the 5th line of the table, which indicates remarkably good calibration performance. It may seem counterintuitive that the pooling of two trial sets gives rise to better performance than that of the sets individually, but it can be seen as an effect of the trial weighting procedure that we employ.

## 5. Discussion and conclusions

On the one hand, the experiments in the previous section are examples of investigations that can be performed with a multi language and accented forensic database such as NFI-FRITS. On the other hand, it shows the limitations in the way we compare conditions of the speaker comparison, because different conditions will select different sections of the data with different speakers and different numbers of speakers. The focus of this research is the methodology of using data from the database rather than characterisation of the absolute performance of the recognizer under different circumstances.

In the introduction we mentioned that the metadata in NFI-FRITS can be used to select circumstances  $I$  relevant to a forensic case. By selecting Turkish speakers in Section 3 we simulate such conditions. Then in Section 4 we used a further conditioning to investigate if there are major performance differences in speaker recognition for spoken language (native Turkish, Dutch or a language mix), and the results are that we observe a similar effect to what has been reported earlier [9], namely that in general discrimination performance does not vary a lot with spoken language, but that there is a larger effect in terms of calibration performance.

We have not conducted a statistical significance test in the analysis of language effects. Apart from statistical tests for detection costs [27, 28] we are not aware of such an analysis for the measures  $E_{=}$ ,  $C_{llr}$  and  $C_{llr}^{\min}$ . The speaker-equalised trial weighting we use (cf. (3) and (4)) further complicates such a rigorous analysis. We therefore limit ourselves to the more qualitative remark that when we take our test data to be more specific to the case at hand, the number of available speakers, segments and trials from the database diminishes rapidly, with a much more ragged DET curve and larger uncertainties in performance characteristics.

One might be concerned that the discrimination performance ( $E_{=} \approx 12\%$ ) is not high enough to be used in Court. However, as can be seen from Figure 5, the expected value for the LLR in the case of  $H_p$  is about 5, which corresponds to a LR of about 150. Depending on the circumstances of the court case, this can be quite an informative value.

With the recording and annotation of NFI-FRITS we hope to have provided an infrastructure for investigating the appli-

Table 7: The performance of the recognizer’s log-likelihood-ratio scores when operating with reference population, conditioned by language/accent mix.

exp	train	test	$C_{llr}$	$C_{llr}^{\min}$	$E_{=}$	$N_{tar}$	$N_{non}$	$N_s$
1	tur	tur	0.533	0.499	14.0	762	23k	49
2	nld-t	nld-t	0.634	0.493	17.2	480	5k2	19
3	nld/tur	nld/tur	0.383	0.331	8.9	90	930	18
4	tur	↔ nld-t	0.666	0.492	15.8	573	22k	58
5	lines 1 and 2 pooled		0.493	0.432	13.7	1k2	28k	58

cation of automatic speaker recognition systems to forensic speaker comparison. With the annotation of a number of linguistically and acoustically relevant parameters the database can be used to select calibration material relevant to a particular case. This allows validation of the use of automatic speaker recognition systems for forensic speaker comparison.

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