IDIAP RESEARCH REPORT

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Martigny - Valais - Suisse

Swiss French PolyPhone and PolyVar: telephone speech databases to model inter- and intra-speaker variability

G. Chollet [†‡] J.-L. Cochard [†] A. Constantinescu [†]
C. Jaboulet [†] Ph. Langlais [*]

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Dalle Molle Institute for Perceptive Artificial Intelligence • P.O.Box 592 • Martigny • Valais • Switzerland

phone +41 − 27 − 721 77 11
fax +41 − 27 − 721 77 12
e-mail secretariat@idiap.ch
internet http://www.idiap.ch

† IDIAP, CP 592, CH-1920 Martigny
‡ CNRS URA820 F-75634 Paris, France
* LIUAPV, Avignon, France

Swiss French PolyPhone and PolyVar: telephone speech databases to model inter- and intra-speaker variability

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Abstract. Following the demand of the speech technology market, a number of companies and research laboratories joined their forces in order to produce valuable and reusable resources, especially speech databases.
Serving their purpose, the collected databases are used for developing, testing, enhancing and evaluating speech technology products, like interactive voice servers, listening typewriter, speaker verification and identification systems, *etc.* Especially for capturing intra-speaker variability, the PolyVar database was designed and recorded at IDIAP, as a complement to the Swiss French PolyPhone database, which adresses inter-speaker variability issues.
We will detail in the following the specific problems of speech database collection (sampling the speaker population, selection of vocabulary items, ...), and will present actual development we carried out at IDIAP throught the PolyPhone and PolyVar databases.

Contents

1 Introduction

Following the demand of the speech technology market, a number of companies and research laboratories joined their forces in order to produce valuable and reusable resources, especially speech databases. Examples of such existing consortia are COCOSDA or SpeechDat, where the latter aims at developing multilingual PolyPhone speech databases recorded over the telephone. Other existing PolyPhone database are MacroPhone [BTG94], Dutch Polyphone, Voice across Hispanic America. MacroPhone consists of approximately 200 000 utterances by 5 000 speakers, who were recorded in the United States. For the distribution of these and many other speech corpora, different national and international database centers have emmerged. LDC (Linguisic Data Consortium), ELRA (European Language Resources Association), and BAS (Bavarian Archive for Speech Signals), shall be named here as examples.

Serving their purpose, the collected databases are used for developing, testing, enhancing and evaluating speech technology products, like interactive voice servers, listening typewriter, speaker verification and identification systems [SJT+92, Nie94, Cho94, MAC94, Str94], *etc.* Especially for capturing intra-speaker variability, the PolyVar database was designed and recorded at IDIAP, as a complement to the Swiss French PolyPhone database, which adresses inter-speaker variability issues.

2 General overview

We will detail in the following the specific problems of speech database collection (sampling the speaker population, selection of vocabulary items, ...), and will present actual development we carried out at IDIAP throught the PolyPhone and PolyVar databases.

Figure 1 shows the overall processing of database collection that we will describe in the following sections.

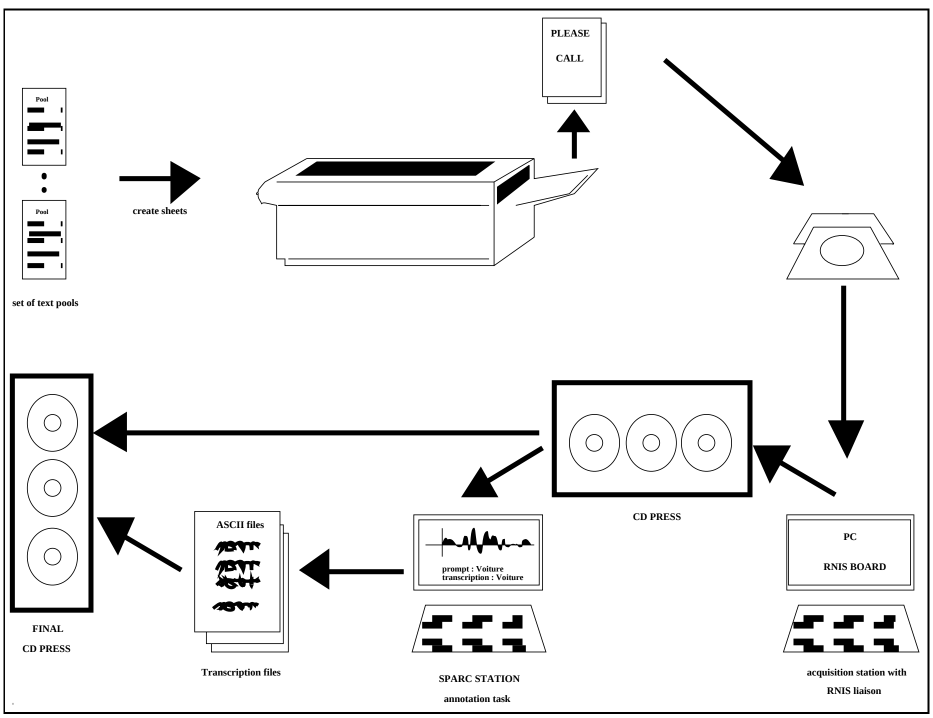

Figure 1: Overview of the collection database process.

3 Database specifications

3.1 Contents

The goal of the Swiss French PolyPhone database, was to record 5 000 speakers over the telephone network. Therefore, persons from all parts of French speaking Switzerland were approached and received calling sheets. Each sheet distributed to the speakers was made up of 38 items. The desired utterances were:

1 sequence of 6 single digits including the hash (#) and the star ($\star$) symbols

1 sheet id number (5 connected digits / 1 natural number)

1 telephone number (spontaneous)

1 16-digit credit card number

2 natural numbers (1 + sheet id)

2 currency amounts

1 quantity

3 spelled words

1 time phrase (prompted, word style)

1 date (spontaneous)

1 date (prompted)

1 yes/no question

1 city name (prompted)

1 city name (spontaneous)

5 application words from a list of 105

1 name (spelling table)

1 mother tongue (spontaneous)

1 education level

1 telephone type

10 sentences (read)

1 query to telephone directory (given the name and the city of subject)

1 free comment on the call

The application word list, with a total of 105 entries, showed control words for different possible voice servers, e.g. telebanking, information kiosk, and ticket reservation.

As a complement to the PolyPhone database, the collection of the PolyVar database was decided and started. The contents of the latter corpora, which was designed with regards to intra-speaker variablility, are similar to those of its sibling, PolyPhone.

3.2 Speakers

Speaker recruitment for the Swiss PolyPhone database was taken care of by LINK, a marketing firm in Lausanne. They addressed people from all over French speaking part of Switzerland. The coverage of regions is proportional to their size: the number of speakers from every city depends on the population size city. Other sampling criteria applied in order to obtain a broad demographic coverage were:

- Sex

- Age (−29, 30 − 49 and 50+ years)
- Employment status (full-time, part-time, unemployed)

The participation rate of the people approached, was rather high and resulted as follows:

- 66% agreed to call the server when asked to participate,
- 63% of those who agreed did really call,
- which gives a total of 42% effective participants.

Unlike for the PolyPhone database, the speakers participating in the collection of the PolyVar database were not selected by any demographic means. It was rather the availability criteria which imposed fundamental constraints, delimiting candidates mainly to members of IDIAP, their families, and befriended researchers. However, it was tried at least to balance the number of male and female speakers to some degree. The demographic unbalance, which at first might be rated negative, does not appear grave at a second thought: it is expectable, that intra-speaker variability will hardly depend on the person's age or social background, if recordings are collected during a period of a few months.

3.3 Generation of calling sheets

As already mentioned, each speaker was asked to read 10 sentences. In order to ensure good phonetic coverage for the resulting database, a *greedy* algorithm was used to select the sentences from some different corpora (newspapers, Internet texts, . . .). The sheet generation program computes the phonetic value of possible sentences and compares them to the actual phonetic coverage (see figure 2) resulting from the annotated calls. The phonetic value of a sentence is defined as the sum of phoneme, diphone, triphone, and polyphone enhancements obtained, when compared to the actual coverage. In this context, the expected enhancement brought to a class (phonemes, diphones, triphones, polyphones) is the square of the subtraction result of each element in the class to the mean average of all elements (see figure 3).

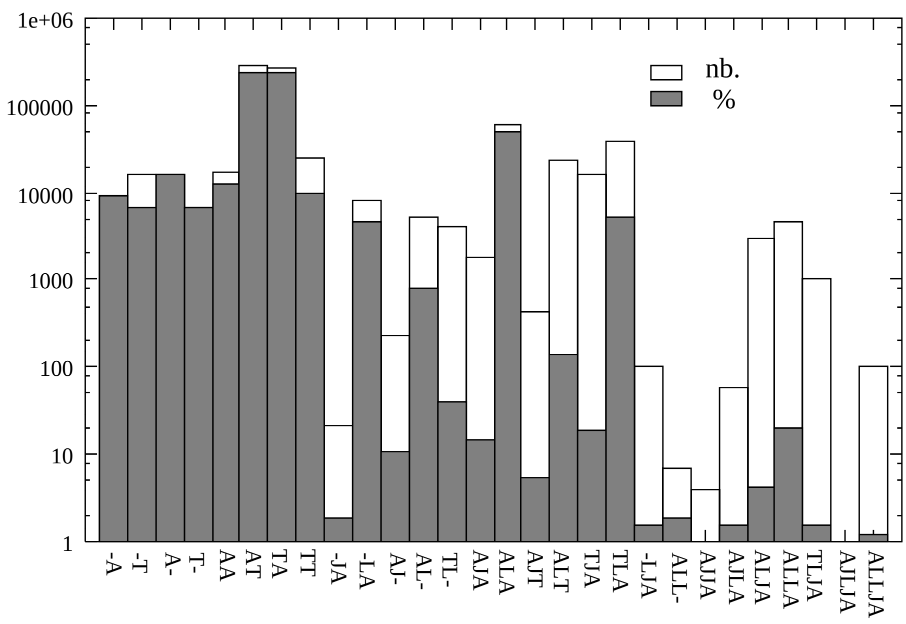

Figure 2: Example of polyphones coverage.

A similar greedy algorithm is applied to numbers, since their pronunciations vary considerably depending on the context surrounding them. Therefore, the left and right adjacent phonemes are taken into account for each number word. Thus, by selecting numbers with respect to the context in which the included number words are found, a good sample is obtained.

The spelled words are selected from a list of names and titles. Here again, a greedy algorithm ensures a quite homogene distribution of letters and special characters (e.g. accents) in several contexts used for spelling.

Finally, the function words to appear in the sheets, are determined randomly from the function word list.

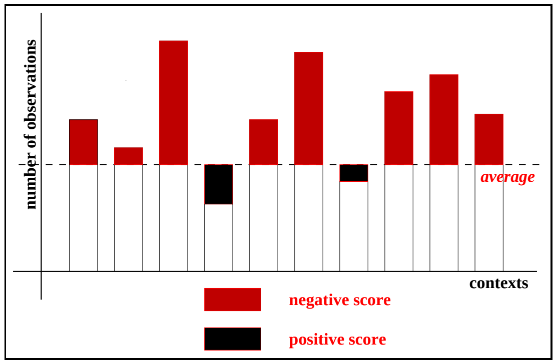

Figure 3: Principle of the greedy algorithm for coverage purpose.

As can be seen in figure 2, even with large amount of sentences (more than 20 000) there is a lack of several units (here polyphones). This points out a crucial problem from speech recognition point of view, since even rare units need to be modeled. In near futur, we will need to take into account sentence generation technology in order to fill the gaps of phonetic units and also to be able to learn linguistic structures (syntactic, semantic, ...) that will be integrated to comprehension task. In order to be aware of the problem of phonetic coverage, we show in figure 4 the evolution of the polyphones coverage with the number of sentences.

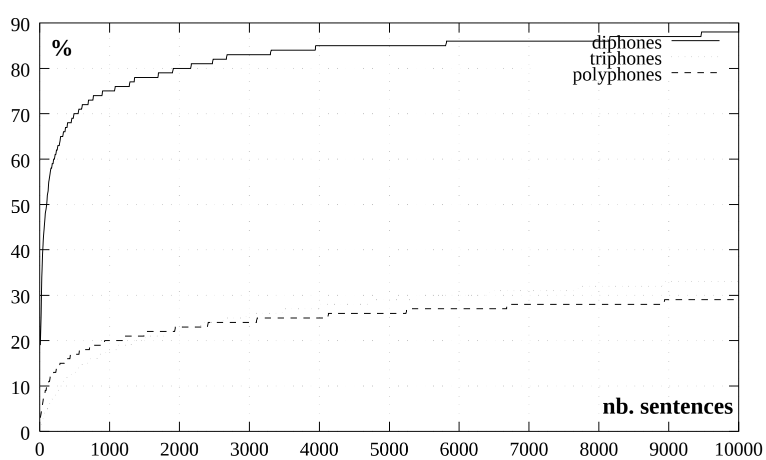

Figure 4: diphones, triphones and polyphones coverages compared to the number of sentences analysed.

4 Technical details

4.1 Recording

For the recording of the PolyPhone calls, a Rhetorex platform was used. The recording site was located at Bern and was operated by the Swiss Telecom PTT on SwissNet II (Swiss ISDN). Thanks to this platform, 15 ISDN lines could be monitored and recorded from in parallel. The recordings were stored on CD-ROM, using 8 bits A-law, and then sent to IDIAP for post-processing.

The PolyVar calls were recorded at IDIAP on a PC based platform, using a DialSys board commercialised by ACSYS. This board offers an anolog telephone interface.

We observed in several experiments differences in the two kind of recordings that show the importance of telephone line adaptation.

4.2 Final format of speech database

The processed and annotated calls are compressed with the SHORTEN algorithm (which proved to give a profit of 20% for PolyPhone recordings and 40% for PolyVar calls) and stored on CD-ROMs. Each item of the prompt sheet is stored in a separate file with a NIST header.

This is an example of a NIST header for an item of a call. All but the first 8 fields are specified by NIST standard:

```
database_id -s26 Swiss French Polyphone 0.0
recording_site -s17 Swiss Telecom PTT
sheet_id -i 13946
prompt -s26 Informations consommateurs
text_transcription -s26 Informations consommateurs
speaking_mode -s4 read
sample_begin -r 0.200000
sample_end -r 2.725125
sample_count -i 23401
sample_n_bytes -i 2
channel_count -i 1
sample_coding -s26 pcm,embedded-shorten-v1.09
sample_rate -i 8000
sample_byte_format -s2 10
sample_checksum -i 12379
```

Field one names the database, field two stores the recording site, the third field value is the calling sheet id. In the two following fields the prompted text and the orthographic transcription of the real utterance are included. The sixth field explains whether the utterance was "read" or "spontaneous". Field seven and eight, label beginning and end of the usable speech (in seconds). When possible, the recorded signal was cut 200 ms before and 200 ms after the usable speech segment.

5 Transcription

In the context of the SpeechDat project, international annotation guidelines for speech databases were developed as a common basis, allowing for supplementary individual extensions to the standards agreed upon. For the Swiss French PolyPhone and PolyVar databases, verification of orthographic transcription of the recorded utterances was performed at IDIAP after receiving collected calls on CD-ROMs. For further processing, the calls were converted to 16 bits linear.

5.1 General guidelines for orthographic transcription

The orthographic transcription is intended to be done at a lexical level. Only few acoustic events are mentioned in the transcription : unintelligible words, unusual prononciation of words or letters, hesitations on parts of sentences. The transcriber should write what was really pronounced, and not what the speaker was supposed to say by.

The transcription is intended to be a quick procedure. Transcribers should not have to agonize over decisions, but rather realize that their transcription is intended to be a rough guide that others may examine further for details.

All items (even numbers, date, time, ...) are transcribed in letters; abbreviations are not used, unless they are spoken in their abbreviated form.

One objective of the transcription is to keep as much speech files as possible in the corpus. Thus we try to avoid deleting items due to extra noises or disfluencies.

Moreover, general rules for written French should be respected during the transcription. For example, the transcription is case sensitive, and all proper names are to be capitalized.

5.2 Transcription interface

We developed a tool called Annotator for the purpose of verifying and correcting the transcription of utterances. We are about to include the phonetic transcription, which at present is performed in a separate step, directly into the annotation tool.

The Annotator interface (see figure 5) works under SunOS or Solaris on Sun workstations. Those must have audio equipment. Moreover, the annotation tool requires an installed version of Xwaves from Entropics and the public domain package TCL-TK.

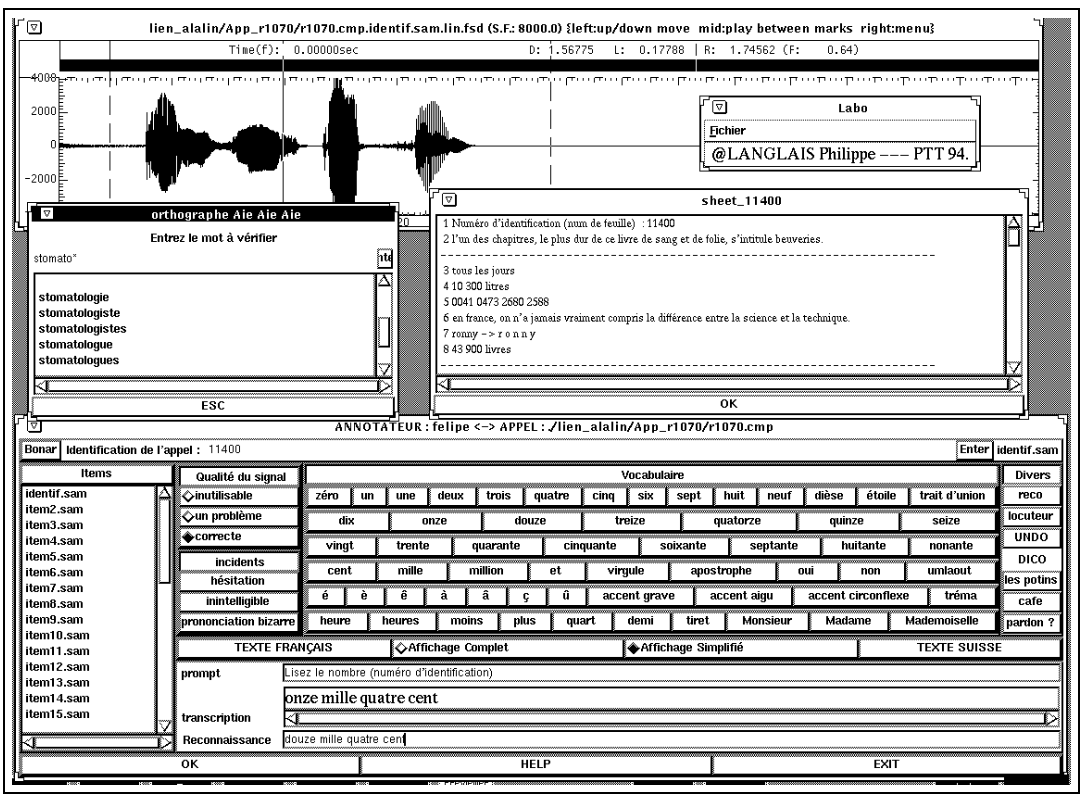

Figure 5: ANNOTATOR interface for transcription purpose.

6 Automatic processing

The system $\mathsf{ETC}_{\mathsf{vérif}}$ [CF95], for "Environment for Cooperative Treatment applied to the verification of speech samples" ("*E*nvironnement de *T*raitement *C*oopératif appliqué à la *vérif*ication d'échantillons de parole"), under development in our laboratory, is intended to provide valuable help for the difficult task of phonetic labelling of speech corpora.

The task dedicated to $\mathsf{ETC}_{\mathsf{vérif}}$ is to compare a given speech sample to a given written text, and to decide whether the utterance is or is not coorect. This decision is based on a processing of two streams of input data: the speech sample, on one hand, and the prompted text, on the other hand, using different knowledge sources (KSs). $\mathsf{ETC}_{\mathsf{vérif}}$ is defined as a multi-agent system, decomposed into two layers (see figure 6): a kernel, a general purpose platform called ETC that is application-independent, and a periphery that bears all the knowledge of the application domains. The ETC platform provides different services to the host application:

- a model of Ω-*agents* that defines a standard interface to distinct entities representing specific areas of expertise;
- a model of μ-*agents* that gives rise to the notion of active data, allowing a decentralized structuring of local hypotheses;
- a framework for the definition of the system evolution dynamics.

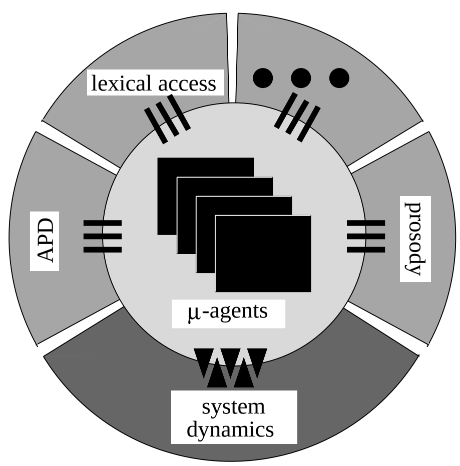

Figure 6: General overview of $\mathsf{ETC}_{\mathsf{vérif}}$ architecture.

The original intention behind the selection of a multi-agent approach is two-fold. Firstly, we consider this approach as the most promising one to be successful in building a system that offers the capability to exchange or add KSs. Secondly, this approch gives us the means to prevent the combinatory explosion problem that occurs with an exhaustive search in a huge solution space.

Now, we clearly see that it is possible to favorably combine $\mathsf{ETC}_{\mathsf{vérif}}$ approach and human interaction. One can find here an illustration of our global approach and its capability to integrate new agents actively contributing to the resolution process. In this context, the user is in position, through a graphical interface, to intervene in the resolution process, as any other agent can do, by making local propositions that point the system to a distinct area of the solution space.

Thus the phonetic transcription of large speech corpora will become a reasonnable task as the system can provide online assistance with a phonetic alignment that can integrate some local user-made modifications. In a fully automatic or in a user-assisted mode, the system keeps the control of the final solution. This particular feature of $\mathsf{ETC}_{\mathsf{vérif}}$ if we are able to set up robust evaluation strategies, will provide the guarantee of the reliability of annotated databases.

6.1 Ω-agent, provider of solutions

An Ω-*agent* is a conventional problem solving module surrounded by a uniform interface. Each Ω-agent represents a specific area of expertise; it is thus able to provide local solutions to problems falling in the scope of its competence. As introduced by Lander [Lan94, p. 12], each Ω-agent can be queried in three different ways:

initial solution – the agent competence is requested to generate a proposal that solves a local sub-problem and that can be used as part of a composite solution.

local alternative – the agent competence is requested to provide an alternative solution with a focus of attention on what was conflicting in the previous proposal(s).

criticism – the agent competence is requested to evaluate a proposal in order to see if it can fit its own requirements.

These three primitives are sufficient to possibly ensure an entire covering of each agent solution space. But, due to the focus mechanism that plays a role of a heuristic function, there is no guarantee that, for a particular problem, every position in a particular solution space will be reached.

In $\mathsf{ETC}_{\mathsf{vérif}}$, the set of Ω-agents currently available are:

- HMMs for phonemes and words (built from the Polyphone),
- a bottom-up, rule-based, acoustic-phonetic decoder [MG91],
- an hybrid LVQ-HMM system for phonemes recognition [Tor94],
- a grapheme to phoneme transcription based on BDLEX [PdFP92],
- a macro-prosodic module that labels syntactic constituents boundaries [LC95].

6.2 μ-agent, element of a composite solution

Each atomic result provided by an Ω-agent, e.g. a phonetic label, a word-form, a prosodic label, etc., is encapsulated in a framework that defines its interacting capabilities, called a μ-*agent*. Each type of atomic result is assigned a distinct set of behaviors. In the Figure 7, an aspect of each μ-agent is addressed, namely its capability to weave relations with other μ-agents. In this artificial situation, three kind of relations are presented:

solid line – express neighborhood of phonetic labels based on temporal information or of word-forms based on writing convention.

solid line with white arrows – shows a case of phonetic clustering. Two distinct answers have been provided within the same class of phonemes on compatible temporal intervals.

dashed line – denotes the fact that a local solution has been used by an Ω-agent to produce other solutions.

The purpose of all these relations is two-fold. Firstly, they are used to define a simple path traversing algorithm, in order to combine atomic solutions in a coherent composite solution. Secondly, the relations play a predominant role in the dynamics of the system as they are central in the cycles of evaluation-improvement of the current solution with the final objective to have a solution with a high enough confidence rate.

6.3 The dynamics of the system

For each global configuration of μ-agents, the system has to evaluate the current set of composite solutions. If a particular composite solution (a path from left to right in Figure 7) get a high confidence score, this solution is provided as *the* answer to the global problem. Otherwise, a local

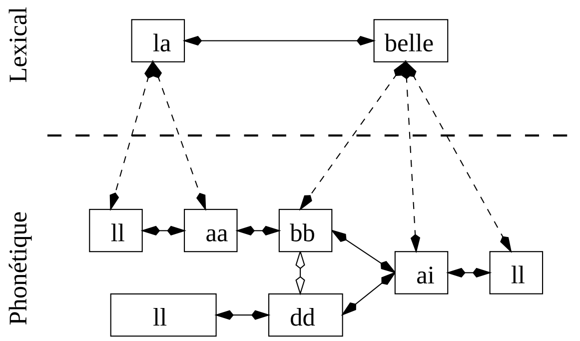

Figure 7: A partial state of a composite solution during a simulated processing of the sentence "*La belle ferme le voile*".

improvement process is started on relatively low confidence μ-agents areas of the best solution. This cycle is performed as long as no sufficiently good solution is available and as far as improvements are still possible.

The determination of a good solution is directly related to the fact that an atomic result is assigned a reliability estimate. The computation of a reliability measure is based on the following formulas 1–3:

$$C(A) = \lim_{i \to \infty} C_i(A) \tag{1}$$

$$C_i(A) = 1 - \sum_{j=1}^{i} \frac{1}{(2 + Ce_{j-1}(A))^j}, \text{ for } i \geq 1 \tag{2}$$

$$C_0(A) = 1$$

$$Ce_i(A) = \sum_{B=1}^{N} V(A,B)\ C_i(B) \tag{3}$$

The reliability of a μ-agent A, $C(A)$, is defined as the value on which $C_i(A)$ is converging. The definition of $C_i(A)$ insures that successive values converge in the interval $]0,1[$ since $Ce_i(A) > 0$. $Ce_i(A)$ denotes the contribution of A neighbourhood in the computation of A reliability. The definition of $Ce_i(A)$ relies on the assumption that all μ-agents are denoted by natural numbers from 1 to N and $V(A,B) = 1$ if A is directly connected to B, $V(A,B) = 0$ otherwise.

Figure 8 gives a graphical illustration of the evolution of $C_i(A)$ for all μ-agents mentionned in the figure 7.

Once the reliability measure is computed for all μ-agents currently present in the system, an average relaiability value $\bar{C}$ is computed for each path that covers the entire speech sample. The path with the highest $\bar{C}$, called $\bar{C}_M$ is selected as a focus of interest. Two situations may occur at this point, either $\bar{C}_M$ is higher than a predefined threshold, or it is lower. In the first case, nothing has to be done; the system is considered to be successful in solving the problem. In the second case, an attempt to improve the current best composite solution is performed by requesting local alternatives to Ω-agents responsible of low reliable elements of this solution.

It is precisely at this point that a end-user can intervene. As the system is strongly asynchronous, a human intervention, by means of adding hypotheses (phonetic, prosodic, or any other that is related to a class of μ-agent) will be automatically considered as a new element to be integrated in a new iteration of reliability measure, selection, decision. The figure 9 gives an overview of the first version of a GUI that allows the user to build graphical realizations of local hypotheses with temporal boundaries. All the relations are automatically computed as requested by the various classes of μ-agents.

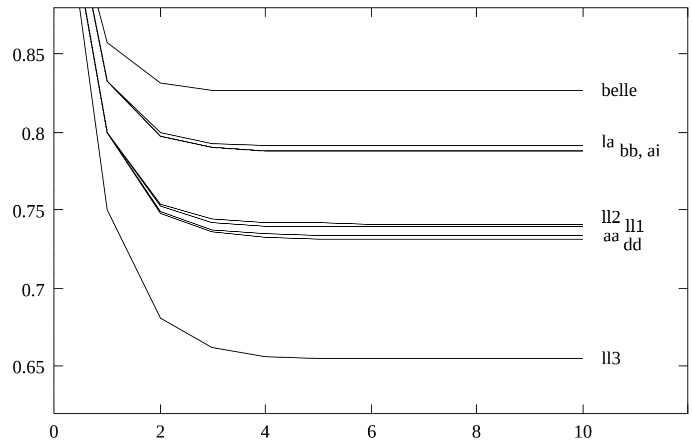

Figure 8: Graphical illustration of the convergence property of the currently implemented reliability measure.

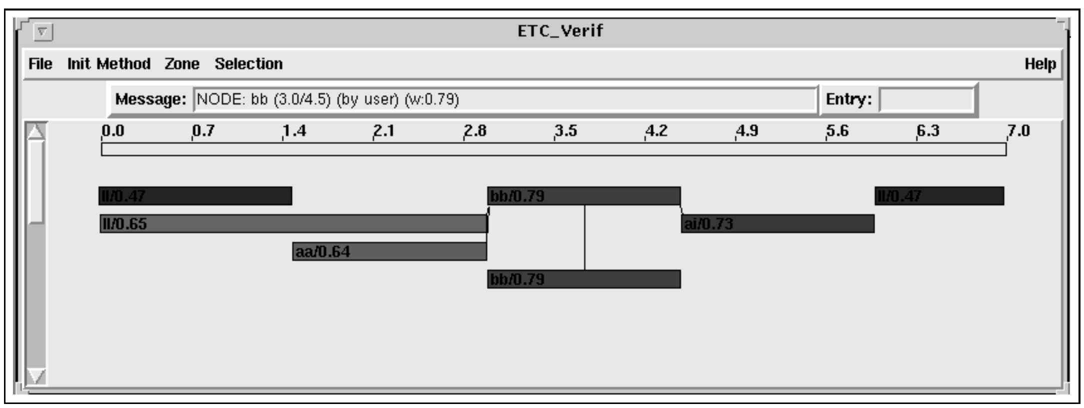

Figure 9: First prototype of a GUI that lets the user interact with a process of problem solving based on the multi-agent philosophy implemented in .

7 Applications

first application, parts of the subcorpora of pronounced digits were extracted and used as training and testing material for our HMM based digits recognizer. Though only a fraction of the available utterances was used, the usefulness and necessity of such speech databases become obvious by looking at figure 10, which illustrates the enhancement of performance in relation to the amount of training material.

Since IDIAP is presently involved in several projects concerned with speech recognition at the level of digits (ATTACKS), spelling, keyword spotting (CERS), as well as with speaker verification tasks (CAVE, M2VTS), we will continue to exploit the developed PolyPhone and PolyVar databases in order to improve and test existing speech technology systems and to develop new ones. Moreover, we intended to implement our research results into real world applications, like tele banking and other voice accessed security sensitive domains. It is also foreseen to develop new voice driven servers for services like information kiosk, or personal assistants in collaboration with partners, e.g. Swiss Télécom PTT.

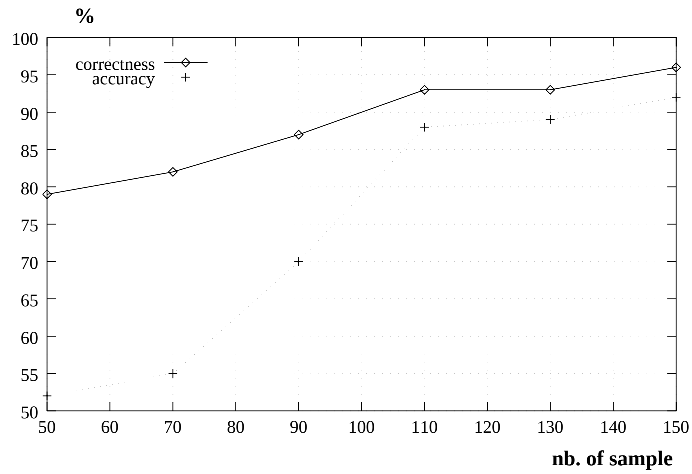

Figure 10: Average performance of our digits models compared to the number of learning examples used to build each model.

8 Conclusions

The need of standardized, reusable speech databases is obvious. The benefits they offer, are at hand: a large set of data allows to enhance existing speech technology and to develop new products. Thanks to such speech databases, it is easier to assess speaker verification and speech recognition systems, and to evaluate new different approaches, as they arise, e.g. flexible vocabulary. The efforts of different database consortia with the goal of collecting these costly reusable resources are worthy to be undertaken. Upon completion, these speech corpora are made available through organisations like LDC or ELRA. The former, for instance, is already distributing several speech and text databases, where the range of items on the ordering form varies from corpora of spoken digits (TIDIGITS), over continuous speech databases (ARPA), to speaker identification data (YOHO).

This working structure of producers, distributors and users, will continue to grow, offering more and more new resources. Resources, which at turn will help enhancing performance of speech technology systems — thus bringing us step by step closer to our goal.

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A Hardcopy of a prompting sheet

feuille d'appel n°13946

Votre appel téléphonique au N$^{\underline{o}}$ gratuit 155 28 14

Tout d'abord, au nom de la recherche scientifique suisse, de l'économie en général et plus particulièrement des Télécom PTT et de l'institut LINK, nous tenons à vous remercier de votre précieuse collaboration.

Directives sur la façon de procéder

Lisez attentivement les quelques directives suivantes :

- **lors de l'appel :**
  - **la voix enregistrée du répondeur annonce les numéros en gras (colonne de gauche) de chaque rubrique**
  - **le répondeur enchaîne avec l'énoncé explicatif de la demande;** à titre purement informatif cet énoncé est mentionné dans la colonne du milieu (et aux rubriques 35-36 un choix de réponses est proposé entre parenthèses)
  - **un "bip sonore" vous indiquera que c'est-à-vous de parler: lisez alors le texte de la rubrique concernée**
- **avant de faire le N$^{\underline{o}}$ gratuit** 155 28 14, parcourez une première fois les 38 rubriques de ce document
- si vous souhaitez une information complémentaire, vous pouvez téléphoner au N$^{\underline{o}}$ 031 / 338 64 57 (MM. Mury ou van Kommer pendant les heures de bureau)

Ci-dessous commence votre tâche effective

N$^{\underline{o}}$	Explication / texte annoncé par l'automate	Ce que vous devez lire (juste après le "Bip sonore")
1	Lisez le nombre :	*13946*
2	Lisez la phrase :	*La fermière élève des oies.*
3	Lisez le(s) mot(s) :	*oui*
4	Lisez la quantité :	*4 032 mètres*
5	Lisez le numéro de carte de crédit :	*6493 3578 0602 1217*
6	Lisez la phrase :	*Déchausse-toi avant d'entrer.*
7	Lisez le nom puis épelez-le :	*Bexen* *B e x e n*
8	Lisez la somme :	*28 011 Dollars U S*

page suivante s.v.p

feuille d'appel n°13946

9	Lisez la phrase :	*L'épervier a saisi un moineau dans ses serres.*
10	Lisez le(s) mot(s) :	*informations consommateurs*
11	Lisez la phrase 11 :	*Le coupable a fait des aveux spontanés.*
12	Lisez la phrase 12 :	*Il nous a noyés dans des explications invraisemblables.*
13	Lisez l'heure :	*22:23*
14	Lisez le mot :	*Nicolas*
15	Lisez la phrase :	*Ce musicien a du génie.*
16	Lisez le(s) mot(s) :	*réservation*
17	Lisez la somme :	*14 811 Lires*
18	Lisez la date :	*Mercredi 4 octobre 2022*
19	Lisez la phrase :	*Les hameçons, les filets sont des engins de pêche.*
20	Lisez le nombre :	*12 028,711*
21	Lisez le(s) mot(s) :	*Le temps*
22	Lisez le nom puis épelez-le :	*Valaisan* *V a l a i s a n*
23	Lisez la phrase :	*Je sens une douleur à l'épine dorsale.*
24	Lisez le(s) mot(s) :	*l'heure*
25	Lisez un à un les chiffres et signes :	** 0 7 9 5 4*
26	Lisez le nom puis épelez-le :	*Scacco* *S c a c c o*
27	Lisez la phrase 27 :	*Cette maison a été le théâtre d'un crime.*
28	Lisez la phrase 28 :	*Il communie tous les dimanches.*

page suivante s.v.p

feuille d'appel n° 13946

29	Lisez le nom de ville :	*schwytz*
30	Veuillez poursuivre en donnant un numéro de téléphone que vous connaissez par cœur :	....
31	Indiquez votre langue maternelle :	....
32	Répondez par Oui ou par Non à cette question : êtes-vous de sexe féminin ?	....
33	Indiquez votre date de naissance :	....
34	Indiquez la ville dans laquelle vous avez commencé votre formation scolaire, à savoir la 1ère année d'école primaire :	....
35	Indiquez le niveau final de votre formation scolaire (école primaire, école professionnelle ou école supérieure) :	....
36	Indiquez le type de téléphone que vous utilisez en ce moment précis (standard Tritel, téléphone sans fil, Natel C, Natel D, un appareil importé, ou une cabine téléphonique publique) :	....
37	Veuillez maintenant faire comme si vous étiez en ligne avec le 111 ...pour demander le n° de téléphone de la personne imaginaire dont les coordonnées se trouvent en face :	*...PIPOZ-PILET NICOLE* *BOIS-NOIR 18* *CERNIER...*
38	Votre éventuel commentaire sur l'ensemble de cet entretien :	....

Nous vous remercions de votre collaboration et espérons que la taxcard vous sera utile ... et la chance favorable lors du tirage au sort du voyage !