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A Spoken Document Retrieval System for TV Broadcast News in Spanish and Basque[*]

Sistema de recuperación de noticias de televisión en castellano y euskera

A. Varona, S. Nieto, L.J. Rodriguez-Fuentes, M. Penagarikano, G. Bordel, M. Diez
University of the Basque Country, UPV/EHU
GTTS, Department of Electricity and Electronics
amparo.varona@ehu.es

Resumen: El sistema de indexado y búsqueda de contenidos multimedia que se presenta en este trabajo (*Hearch*) es un buscador de aspecto convencional pero con la capacidad de devolver segmentos de vídeo gracias a la transcripción automática de sus contenidos de voz. El sistema consta de un *back-end* que capta, procesa e indexa los recursos, y de un *front-end* que permite realizar búsquedas y configurar y monitorizar el funcionamiento de los distintos módulos, mediante una interfaz web. Actualmente se encuentra operativa una versión de la herramienta que trabaja frente a repositorios de noticias en castellano y euskera (*http://gtts.ehu.es/Hearch/*). Para evaluar el rendimiento del sistema se dispone de 6 programas de noticias en castellano y 7 en euskera. Puesto que el módulo de Reconocimiento Automático del Habla introduce bastantes errores, se ha propuesto y evaluado una aproximación basada en añadir términos afines a los de la pregunta para ampliar los resultados proporcionados por el sistema. Como resultado se obtiene una pequeña mejora del rendimiento.

Palabras clave: Recuperación de recursos multimedia, Reconocimiento Automático del Habla

Abstract: This paper presents a spoken document retrieval system (*Hearch*) looking like a conventional search tool, which retrieves audio/video segments based on the automatic transcription of speech contents. The system consists of a *back-end* that captures, processes and indexes audio/video resources, and a *front-end* that allows to search contents, configure various modules and display performance statistics through a web interface. An early version of this tool is available (*http://gtts.ehu.es/Hearch/*), which searches and retrieves segments on TV broadcast news repositories in Spanish and Basque. To evaluate the performance of the system, six manually transcribed TV broadcast news in Spanish and seven in Basque have been used. An approach based on extending the query with the so called *friendly terms* has been proposed and evaluated, attempting to minimize the effect of errors introduced by the Automatic Speech Recognition module. This approach led to slight performance improvements.

Keywords: Spoken Document Retrieval, Automatic Speech Recognition

1. Introduction

The fast growth of multimedia (audio/video) contents available in internet makes it necessary searching into such kind of resources. Spoken Document Retrieval (SDR) systems look like conventional search tools (such as Google, Bing, etc): users introduce *query terms* and the system outputs *audio segments* where those terms have been pronounced, sorted according to a given measure of relevance. To achieve that goal, speech must be converted to text or some text describing resource contents must be used. Some systems index and categorize videos using the titles or brief descriptions (metadata) that accompany these resources, such as the tags

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provided by users. These texts are just short descriptions, shallow categorizations or partial transcriptions, so the resulting index is very coarse and the search cannot focus on specific items. Products developed by companies, such as VideoSurf[1], Delve Networks[2] and Truveo[3] use this kind of information.

Clearly, search engines can take a key advantage from using Speech Technologies. The use of Automatic Speech Recognition (ASR) allows to transcribe spoken documents and then text-based Natural Language Processing (NLP) tools can be applied. But automatically generated transcripts still include a large number of errors, which must be taken into account in any information extraction process. In a first phase, several audio processing modules (audio segmentation, audio classification, language verification, speaker identification/diarization, speech recognition, etc.) must be sequentially applied to audio streams, and the information related to segmentation, audio type, language, speaker and the recognized transcription stored for further processing. Text processing tools can be applied to enrich the transcription with morphosyntactic information. Finally, the resulting text must be used to build an index to allow efficient information retrieval (Makhoul et al., 2000).

In the last years, some systems have been already developed applying Large-Vocabulary Continuous Speech Recognition (LVCSR). SpeechBot (Thong et al., 2002) was an experimental web-based tool from HP Labs that used speech recognition to create searchable keyword transcripts from thousands of hours of audio content. SpeechFind (Hansen, 2005) is a spoken document retrieval system developed by the Center for Robust Speech Systems at the University of Texas at Dallas which was used to transcribe the National Gallery of Spoken Words, which covers up to 60000 hours of USA historic recordings from the last 110 years. The Massachusetts Institute of Technology (MIT) system (Glass et al., 2007) dealt with improving the access to on-line audio/visual recordings of academic lectures. The system developed at NTT (Ohtsuki et al., 2006) indexes multimedia contents in Japanese. The ASEKS system (Ye et al., 2006) uses keyword spotting technology for indexing spoken documents in Chinese.

But LVCSR systems do not provide universal coverage: some query terms may not appear in the vocabulary. To solve the lack of coverage, some proposals such as those of Nexidia (Clements and Gavalda, 2007) and IBM (Mamou and Ramabhadran, 2008) are based on spoken term detection. First, the input speech is processed to produce a phonetic decoding, which is searched in a second stage to find the query terms. The main disadvantage of this approach is that search must be performed for each query, and each audio file, thus introducing considerable delays.

Other systems combine metadata information with ASR outputs. For example, the *Google Speech Research Group* (Alberti et al., 2009) developed an audio indexing system for the videos of YouTube corresponding to the 2008 presidential election race in the United States. Another example is a Korean spoken document retrieval system for Lecture Search (Lee and Lee, 2008).

Currently, the best positioned companies in the video search market are Blinkx[4], Ramp[5], Nexidia[6], TVEyes[7], Gaudi[8] (Google Audio Indexer appeared in the third quarter of 2008) and MAVIS[9] (Microsoft Research Audio Video Indexing System). It might seem, by the reported performances and the appearance of their interfaces, that these tools have solved the problem of indexing and searching multimedia contents. However, this is a misperception, since these tools focus on a very specific type of resources to optimize performance.

In this paper, a Spoken Document Retrieval System (*Hearch*) which deals with resources in both Spanish and Basque languages is presented and evaluated. The system retrieves audio/video segments based on the automatic transcription of speech contents. It consists of a *back-end* that captures, processes and indexes audio/video resources, and a *front-end* that allows to search contents, configure various modules and display performance statistics through a web interfa-

[1] http://www.videosurf.com/

[2] http://www.delvenetworks.com/

[3] http://www.truveo.com/

[4] http://www.blinkx.com/

[5] http://www.ramp.com/

[6] http://www.nexidia.com/

[7] http://www.tveyes.com/

[8] http://labs.google.com/gaudi

[9] http://research.microsoft.com/en-us/projects/mavis/

ce. An early version of this tool is available (*http://gtts.ehu.es/Hearch/*) working on 1TB of TV broadcast news in Spanish and Basque obtained from *Euskal Irrati Telebista* (EITB). A preliminary evaluation has been carried out on six manually transcribed Spanish and seven Basque TV broadcast news and two sets of query terms.

The rest of the paper is organized as follows. Section 2 describes the spoken document retrieval system, including modules related to audio processing (audio segmentation, language identification, speaker identification, automatic speech recognition), the NLP tools for text processing, the indexer, the search engine and the user interface. Section 3 describes the databases used in the evaluation. In Section 4 the Automatic Speech Recognition module (the *key* piece of the system) is evaluated independently. In Section 5 the whole spoken document retrieval system is evaluated. Finally, conclusions are given in Section 6.

2. The Spoken Document Retrieval System

The SDR system (*Hearch*) is based on a simple and efficient architecture, which allows to replace or integrate new modules in a easy and elegant way (see Figure 1). In the *back-end* the information obtained at each step is incrementally stored in an XML resource descriptor (Bordel et al., 2009). Audio signals are segmented and classified, the speech segments are transcribed and the resulting sequences of words are lemmatized yielding an enriched XML document. This document is taken as input by the *indexer* to update an index database, which is the core data structure of the system. In the *front-end*, each time a user makes a query, the query is lemmatized and then the search engine traverses the index database to find matching resources. A web interface allows users to access *Hearch* from remote locations, to enter queries and to receive search results.

2.1. The back-end

All the operations described in this section are performed once in *off-line* mode, before user smake their queries.

2.1.1. The crawler

Copies of the original resources are kept locally, and the audio streams are converted into PCM format for further processing. An XML file is generated including source information: size, format, etc.

2.1.2. Audio processing

Audio is processed in several steps. At each step, the XML document is enriched with information specific to a knowledge level.

– ***Audio Segmentation.*** This task consists of dividing a continuous audio stream into acoustically homogeneous regions called *segments*. There are robust and unsupervised techniques for doing it. (Kiranyaz, Qureshi, and Gabbouj, 2006). In particular, the identification of speech and non-speech segments is a key step. If non-speech segments are excluded from recognition, not only computation time is saved in ASR, but also better transcriptions are obtained. Small interruptions like coughs and other noises produced by the speaker are admitted inside of a speech segment (Rodriguez-Fuentes et al., 2010).

– ***Language Identification.*** In multilingual systems, identifying the language spoken in each segment is necessary for the ASR system to use adequate acoustic and syntactic models. In a previous work, language recognition results obtained for the four official Spanish languages (Spanish, Catalan, Basque and Galician) were presented. Best performance was achieved with the fusion of an acoustic system and 6 phonotactic subsystems. Acoustic systems take information from the spectral characteristics of the audio signal, whereas phonotactic systems use sequences of phones produced by several acoustic-phonetic decoders (Varona et al., 2010).

– ***Speaker Identification.*** This task consists in classifying the speech segments in terms of speakers, which has always a positive impact on the accuracy of the ASR system, by applying model adaptation techniques (unsupervised clustering of similar voices, Bayesian adaptation, etc.) (Diez et al., 2011). Moreover, if speaker profiles were available beforehand, then speaker turns could be identified, which is interesting from the point of view of indexing, since users might be interested in finding the segments corresponding to a given speaker.

– ***Automatic Speech Recognition.*** Speech recognition is the process of converting an acoustic signal to a sequence of words. ASR systems are invariably based on the well-known Bayes rule (Jelinek, 1999),

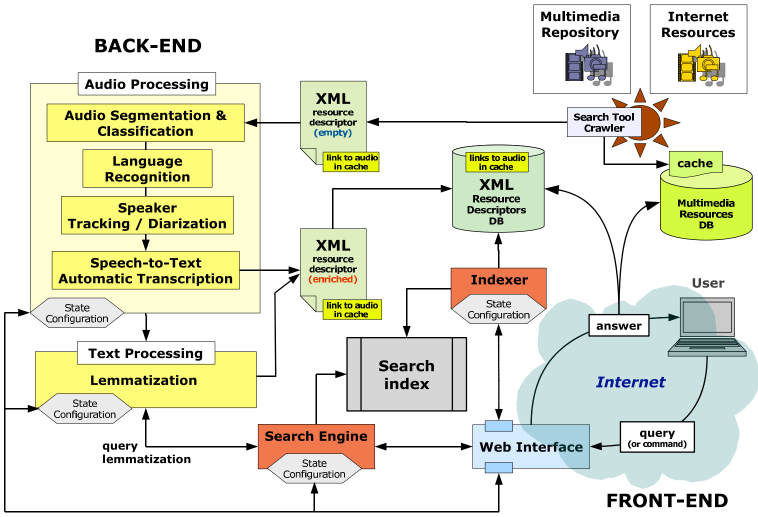


Figure 1: The Spoken Document Retrieval system (Hearch). In the *back-end*, (1) the crawler creates an XML resource descriptor for each multimedia file; (2) audio processing is carried out in five steps: audio segmentation, language identification, speaker identification (optional), automatic speech recognition (ASR) and text-based processing, yielding an enriched XML document; (3) this document is taken as input by the indexer to update an index database, which is the core data structure of the system. In the *front-end*, each time a user makes a query, the search engine traverses the index database to find the matching resources. A web interface allows users to access *Hearch* from remote locations, to enter queries and to receive search results.

i.e the recognizer looks for the most likely word sequence according to previously estimated acoustic models (typically, hidden Markov models) and language models (typically, n-grams). Acoustic models estimate the frequency distributions of sounds over time, and language models estimate the frequency of word sequences. Specific acoustic and language models are trained for each language. This module is currently implemented by means of Sautrela[10] (Penagarikano and Bordel, 2005), an open-software package for speech processing, entirely developed using Java. The Hearch architecture also supports the inclusion of other components, such as the recognition engine of HTK (Young, 2006).

– ***Text processing.*** The morphosyntactic analysis of the recognized sequences of words allows us to know the *lemma* of each word, which helps to index and search inflected forms. To analyse words in Spanish, the well-known FreeLing package (Atserias et al., 2006) is used. Basque is a deeply agglutinative language and specific tools are applied to carry out lemmatization (Aduriz et al., 1998).

2.1.3. The indexer

The collection of enriched XML documents is taken as input to the indexer tool, which creates a hierarchized structure of term references. The indexing process uses a finite set of terms: the *vocabulary* (predefined) of the ASR module $T = t_1, t_2, ..., t_{|T|}$. Each segment in the document is treated as a subset of terms (the order is not taken into account). To lessen computational costs, terms supposed to be useless to represent document content (the so called *stopwords*, such as articles, conjunctions, pronouns, etc.) have been eliminated. Each segment is represented as a vector, and a weight $w_{jk} > 0$ is associated with each term k of a segment $d_j = (w_{j1}, w_{j2}, ..., w_{jk}, ..., w_{j|V|})$. For a term that does not appear in segment d_j, $w_{jk} = 0$. Usually, each term weight is computed based on variations of the tf or $tf - idf$ schemes[11] (Mills et al., 2000).

In this work, the indexer is implemented by means of Apache Lucene (Hatcher, Gospodnetic, and M., 2010), a high-performance full-featured text search engine library written entirely in Java. The index structure, which contains location information for each term, is dynamically updated each time a new XML resource is added to the system.

[10] http://sautrela.org/

[11] tf is the Term's Frequency, defined as the number of times that term k appears in the document d_j. The Inverse Document Frequency idf gives more weight to terms occurring in few documents.

2.2. The front-end

The information retrieval module (the search engine) and the web interface work *on-line*, by processing user queries and presenting a ranked list of results.

2.2.1. The *search* engine.

The information retrieval process begins when the user formulates a query. Again text processing tools (lemmatization) are applied to represent the query q in the same way as segments d_j, i.e. as a feature vector. The system retrieves those segments matching the query terms in the index database, and ranks them according to some predefined *matching measure* (Frakes and Baeza-Yates, 1992). The standard *relevance ranking* of segments of *Apache Lucene* has been modified in order to penalize short segments, so we assign the score taking into account the square root of the number of terms ($NumTerm$) contained in the segment in the following way:

$$Score = C \sum tf \cdot idf^2 \sqrt{NumTerm} \qquad (1)$$

where C is a fraction of the number of query terms found in the segment.

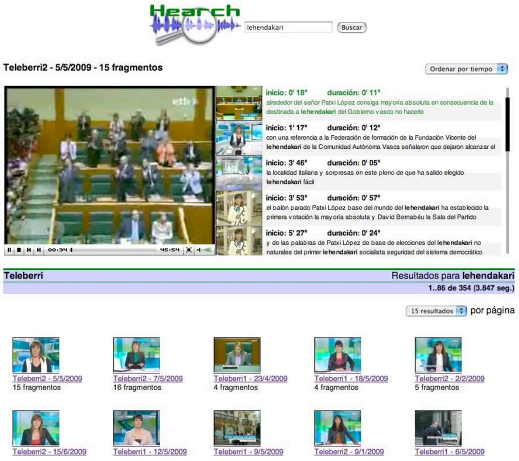
2.2.2. The user interface

The user interacts with the SDR system from a remote computer through internet, using a simple navigator (see Figure 2). A web server is accepting queries and sending them to the search engine, which returns a rank of matching segments. Finally, the system interface composes and presents successive HTML pages showing a list of matching items, with information about the resource name, location and size, segment boundaries (time stamps), links (thumbnails) to cached copies of the original multimedia resources, transcription excerpts which link to the full recognized transcriptions, and relevant metadata (topics, speakers, etc.) stored in the XML resource descriptors.

3. *Evaluation databases*

An early version of the SDR system is available at *http://gtts.ehu.es/Hearch/*, working on 1TB of TV news in Spanish and Basque recorded from EITB. For evaluation purposes, two small databases have been created in both Spanish and Basque languages:

- For Spanish, a collection of 6 TV news (4h 47min stored in 1GB) is available, including 3918 segments, manually segmented and transcribed, with 7779 different words (terms) in the vocabulary.



lehendakari

Teleberri2 - 5/5/2009 - 15 fragmentos

Ordenar por tiempo

inicio: 0' 18" duración: 0' 11"
alrededor del señor Patxi López consiga mayoría absoluta en consecuencia de la destinada a **lehendakari** del Gobierno vasco no hacerlo

inicio: 1' 17" duración: 0' 12"
con una referencia a la Federación de formación de la Fundación Vicente del **lehendakari** de la Comunidad Autónoma Vasca señalaron que dejaron alcanzar el

inicio: 3' 46" duración: 0' 05"
la localidad italiana y sorpresas en este pleno de que ha salido elegido **lehendakari** fácil

inicio: 3' 53" duración: 0' 57"
el balón parado Patxi López base del mundo del **lehendakari** ha establecido la primera votación la mayoría absoluta y David Bernabéu la Sala del Partido

inicio: 5' 27" duración: 0' 24"
y de las palabras de Patxi López de base de elecciones del **lehendakari** no naturales del primer **lehendakari** socialista seguridad del sistema democrático

Teleberri Resultados para **lehendakari**
1..86 de 354 (3.847 seg.)

15 resultados por página

Teleberri2 - 5/5/2009
15 fragmentos

Teleberri2 - 7/5/2009
16 fragmentos

Teleberri1 - 23/4/2009
4 fragmentos

Teleberri1 - 18/5/2009
4 fragmentos

Teleberri2 - 2/2/2009
5 fragmentos

Teleberri2 - 15/6/2009

Teleberri1 - 12/5/2009

Teleberri1 - 9/5/2009

Teleberri2 - 9/1/2009

Teleberri1 - 6/5/2009

Figure 2: User interface of the SDR system.

- For Basque, a collection of 7 TV news (4h, stored in 500MB) is available, including 3391 segments, manually segmented and transcribed, with 8093 different words (terms) in the vocabulary.

4. *ASR Evaluation*

The automatic speech recognition module is the *key* piece of the system. Acoustic and Language Models have been trained on large acoustic and text databases and the evaluation has been carried out on a small but independent set of speech segments taken from TV broadcast news.

– ***Acoustic features***. Audio signals are stored in PCM format at 16KHz, 16 bit per sample. Acoustic features consisted of 12 Mel Frequency Cepstral Coefficients (MFCC) together with log-energy, calculated every 10ms in sliding Hamming windows of 25ms (first order preemphasis with 0.97 coefficient, and a 26 channel filterbank were applied). Delta and double delta coefficients were also computed, resulting in a 39-component feature vector.

– ***Acoustic modeling***. The acoustic modeling is based on left-to-right continuos Hidden Markov Models with three looped states and 64 Gaussian mixtures per state. The phone inventory consisted of 23 phone units for Spanish and 26 phone units for Basque. Both languages share the five vowels but Basque has more fricative and affricate sounds.

For Spanish, the well-known Albayzin database (Moreno et al., 1993) was used to estimate acoustic models. It consists of 6800 read sentences from 204 speakers: 4800 from 164 speakers for training and 2000 from 40 speakers for testing.

For Basque, the acoustic database AS3200 (Aditu) (Basque-Government, 2005) was used. Recordings were made in an office environment, following the specifications of SPEECON[12] (Siemund et al., 2000). In this work, only the subset of read sentences was used, consisting of 8298 sentences from 215 speakers: 5346 from 140 speakers for training and 2952 from 75 speakers for testing.

– ***Language modeling.*** Two databases were used to estimate the language models.

- Spanish_ML contains text news in Spanish taken from the internet from 2005 to 2008 including various topics such as economy, society, sports, opinions, politics, etc. It contains 1.790.654 sentences and 50.862.981 words. The size of the vocabulary for this database is 302.430.

- Basque_ML contains text news in Basque taken from the internet from 2003 to 2008. It contains 2.589.284 sentences, 34.510.77 words and a vocabulary of 661.651 words.

The SRI Language Modeling Toolkit *SRILM* (Stolcke, 2002) was used to estimate n-grams Language Models (LM). Classical *Witten-Bell* smoothing was applied to trigram LMs. Three different vocabulary sizes (dictionaries) were evaluated: (1) *5K words*, including the 5000 most likely words in the whole text database; (2) *20K words*, including the 20000 most likely words in the whole text database; and (3) *closed* dictionary, which contains all the words in the 6 TV news in Spanish (7759) and the 7 TV news in Basque (8093) (see Section 3).

– ***Test sets***. The experiments were carried out on a selection of segments manually extracted from the databases described in Section 3. Segments were classified into three groups according to their audio characteristics: (1) *presenter*: audio signals in this group are clean because they were recorded in a TV studio and the presenter was reading a script; (2) *reporter*: this group contains audio signals recorded by reporters on the streets, so the speech is less formal than in previous ones and probably contains background noise; and (3) *spontaneous*: this group contains spontaneous speech (commonly produced in interviews) so the signals will probably contain a lot of disfluencies. Table 1 shows the number of segments, the number of words and the vocabulary size for each collection of segments.

Table 1: Number of segments, number of words and vocabulary size for each collection of segments, for both Spanish and Basque.

		segments	words	vocab
Spanish	presenter	180	5045	1797
	reporter	180	4482	1655
	spontaneous	96	2594	914
	total	456	12121	4366
Basque	presenter	180	2971	1707
	reporter	180	2885	1626
	spontaneous	90	1569	807
	total	450	7425	4140

– ***Results***. Test segments (see Table 1) were processed by the ASR decoder which outputs sequences of words (without lemmatization). Table 2 shows the ASR performance for the three sets of sentences (presenter, reporter and spontaneous) and the three LMs (5K, 20K and closed). Classical Word Accuracies[13] were calculated. Clearly, Spanish decoders run better than Basque decoders when LM of 5K and 20K were considered. However, for closed-set LMs best performances were obtained for Basque. That is because Basque is a highly inflected language in terms of both nouns and verbs, with 17 cases, thus the 5K or 20K LMs were less general than their counterparts for Spanish.

Performance was also measured when lemmatization was applied both to the recognized sequences of words and user queries. Table 3 shows Lemma Accuracy[14]. Performance improved, specially when considering vocabularies of 5K and 20K words, both for Spanish and for Basque.

5. *Information Retrieval Results*

– ***Choice of query terms***. We have manually defined four groups of queries, three of the them according to frequency: 15 terms that appear frequently in the vocabulary of

[12] SPEECON, launched in February 2000, focused on collecting linguistic data in different European languages for speech recognition applications

[13] Word Accuracy takes into account the Levenshtein distance in a dynamic string alignment: $Acc = (C - I)/N$, where C is the number of correctly recognized words, I is the number of inserted words, and N is the number of words in the reference transcription.

[14] Lemma Accuracy is defined the same way as Word Accuracy but considering lemmas instead of words.

Table 2: Word Accuracies for different ML (5K, 20K and closed-set) considering different types of segments (presenter, reporter and spontaneous).

		5K	20K	closed
Spanish	presenter	64.74 %	73.05 %	86.7 %
	reporter	68.45 %	77.71 %	89.47 %
	spontaneous	43.56 %	46.11 %	58.87 %
	mean	58.91 %	65.62 %	78.34 %
Basque	presenter	49.85 %	59.81 %	91.89 %
	reporter	49.57 %	60.67 %	92.65 %
	spontaneous	27.68 %	28.51 %	60.97 %
	mean	42.36 %	49.66 %	81.83 %

Table 3: Lemma Accuracies for different ML (5K, 20K and closed-set) considering different types of segments (presenter, reporter and spontaneous).

		5K	20K	closed
Spanish	presenter	67.76 %	75.62 %	87.18 %
	reporter	71.46 %	79.67 %	90.00 %
	spontaneous	46.26 %	48.77 %	59.91 %
	mean	61.82 %	68.02 %	79.03 %
Basque	presenter	56.58 %	65.57 %	92.53 %
	reporter	56.26 %	66.37 %	93.21 %
	spontaneous	34.75 %	35.46 %	62.82 %
	mean	49.19 %	55.80 %	82.85 %

the TV news, 25 terms that appear less frequently and 25 unusual terms. The forth group contains 30 queries composed of pairs of terms, with some kind of relationship between them, from the three previous groups. Eventually, we decided to join the first three groups into one, since performance differences observed for the three groups of frequencies were not significant. Thus, we considered two groups of queries, one composed of 65 one-term queries and another composed of 30 two-term queries.

– ***Expanding the query with friendly terms***. Adding *friendly terms* is an attempt to improve the performance of the system by searching for terms with high probability of appearing in the same sentence than query terms. To create a list of *friendly terms*, it is necessary to calculate for every sentence in a big set of sentences (for example the *Spanish_ML* and *Basque_ML* databases) the conditional probability of each term given another term. Given two terms A and B, the joint probability $P(A, B)$ is calculated as:

$$P(A, B) = P(A|B)P(B) = P(B|A)P(A) \quad (2)$$

This results in the well-known Bayes formula:

$$P(A|B) = \frac{P(B|A)P(A)}{P(B)} \quad (3)$$

To avoid that less common terms appear as good candidates, two conditions are imposed: (1) $P(B|A) > P(B)$; and (2) $P(B) > threshold$. This threshold should be close to the inverse of the number of different terms in the text. In this work, $threshold = 0{,}001$.

– ***Evaluation measures.*** The Search Quality Benchmarking package of *Lucene* was used to obtain all the search results given a query, and TREC Eval[15] was used to estimate the similarity between the segment and the query, by means of the *Mean Average Precision (MAP)* measure:

$$MAP = \frac{\sum_{q=1}^{Q} AveP(q)}{Q} \quad (4)$$

MAP is the mean of the average precision scores for each query $q \in Q$. The *average precision* is defined as follows:

$$AveP = \frac{\sum_{r=1}^{N} (P(r) \times rel(r))}{NumberRelevantSegments} \quad (5)$$

where r is the rank, N the number of retrieved segments and $P(r)$ the precision at a given rank r. If the segment at rank r is nonrelevant, the binary function $rel(r) = 0$. Sometimes, this metric is also referred to geometrically as the area under the Precision-Recall curve. The Percentage of Relevant Retrieved (PRR) segments is also used as evaluation measure in this work.

– ***Results***. The SDR system was applied on the set of segments described in Section 3, that is, 3918 for Spanish and 3391 for Basque. Table 4 shows MAP and PRR performances for the two sets of one-term and two-term queries mentioned above. Performance for Basque was worse than for Spanish, maybe because ASR results (the input to the IR system) were also worse. In Spanish the use of two-term queries yielded better results.

Table 5 shows MAP and PRR performance for the same sets when adding *friendly terms*. Experiments with different number of

[15] Trec_eval is a standard tool to evaluate TREC (Text REtrieval Conference) results using the standard NIST evaluation procedures. It was written by Chris Buckley and it is available from the TREC website at trec.nist.gov/trec_eval/.

Table 4: Mean Average Precision (MAP) and Percentage of Relevant Retrieved (PRR) segments when one-term and two-term queries were considered for different LM (5K, 20K, closed-set).

		one-term queries		two-term queries	
		MAP	PRR	MAP	PRR
Spanish	5K	0.4581	60.62 %	0.5200	68.39 %
	20K	0.4907	59.79 %	0.5493	67.34 %
	closed	0.6417	72.40 %	0.6638	75.88 %
Basque	5K	0.2719	37.08 %	0.2815	39.81 %
	20K	0.3379	38.73 %	0.3304	40.55 %
	closed	0.6704	69.26 %	0.6165	67.26 %

friendly terms were carried out. Table 5 only shows best results: (1) in the case of one-term queries, when 3 friendly terms were added; and (2) in the case of two-term queries, when 2 friendly terms were added (one per original term). Note that performance improved in all cases but not very significantly.

Table 5: MAP and PRR performance when (1) 3 friendly terms were added to one-term queries and (2) 2 friendly terms were added (one per original term) to two-term queries, for different LM (5K, 20K, closed-set).

		one-term queries		two-term queries	
		MAP	PRR	MAP	PRR
Spanish	5K	0.4687	64.23 %	0.5316	70.43 %
	20K	0.5015	63.81 %	0.5579	68.07 %
	closed	0.6562	72.96 %	0.6672	77.50 %
Basque	5K	0.2727	41.00 %	0.2841	42.75 %
	20K	0.3459	42.36 %	0.3350	43.26 %
	closed	0.6780	71.11 %	0.6294	69.74 %

6. Conclusions

In this paper, a Spoken Document Retrieval system (Hearch) has been described and evaluated. In the *back-end* an XML resource descriptor is created for each multimedia file, and audio processing is carried out in four steps: audio segmentation, language identification, speaker identification (optional), automatic speech recognition (ASR), yielding an enriched XML document. ASR transcriptions are further processed by NLP tools to get sequences of lemmas (instead of words), which is suitable for highly inflected languages (such as Basque). The enriched XML document is taken as input by the indexer to update an index database, which is the core data structure of the system. In the *front-end*, each time a user makes a query the search engine traverses the index database to find the matching resources. A web interface allows users to access *Hearch* from remote locations, to enter queries and to receive search results.

To evaluate the performance of the system a small database was available, consisting of 6 Spanish and 7 Basque TV broadcast news, all of them manually segmented and transcribed. Better results were obtained for Spanish, maybe because Basque is a highly inflected language, thus needing much more data to estimate reliable language models. Since the sequence of recognized words includes many errors, we have tried to improve system performance by extending the query with the so called *friendly terms*, which makes the system to retrieve many additional segments. This approach led to slight performance improvements for both Spanish and Basque.

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