SLAM: SEGMENTATION AND LABELLING AUTOMATIC MODULE

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Abstract

An interactive Segmentation and Labelling Automatic Module (SLAM), especially developed for Windows-based Personal Computers, is described. The system is extremely user-friendly and it was designed with the aim of supporting speech scientists in assessing the very heavy and time-consuming task of segmenting a big amount of speech material such as that caused by the tremendous spread of new and always bigger speech data-bases. The system, which is based on the Multi-Level Segmentation theory, was built using Microsoft C++ and Windows 3.1 SDK software[1], and runs preferably on Intel 386/486-based personal computers running DOS 5.00 or higher and equipped with VGA and SuperVGA boards.

Keywords: Multi-Level segmentation, Windows.

Introduction

Phonetic or phonemic labelling of speech signals is normally performed manually by phoneticians or speech communication experts. Even if various attractive graphic and acoustic tools are simultaneously available, there will always be some disagreement among skilled human labelling experts in the results of labelling the same wave form[1]. In fact, due to human variability of visual and acoustic perceptual capabilities and to the difficulty in finding a clear common labelling strategy, the manual labelling procedure is implicitly incoherent. Another important drawback of manual intervention in labelling speech signals is that it is extremely time consuming. Considering these and other disadvantages, the development of methods for semi-automatic or automatic labelling of speech data is becoming increasingly important [2] especially considering the present tremendous spread of new and always bigger speech data-bases. Moreover, even if

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segmentation and labelling are avoided by most of the more successful Automatic Speech Recognition (ASR) systems, generally based on Hidden Markov Model techniques, a completely labelled true continuous speech database will always be of interest for other classes of ASR systems, such as those based on Neural Networks techniques, or for linguistic and phonetic research.

Complete automatic labelling systems minimise assessment time of input/output speech data-bases and are at least implicitly coherent. In fact, using the same strategy, if they make some errors they always make them in a coherent way. Unfortunately, at the present time highly reliable automatic segmentation systems are still not on the market. The semi-automatic system being described constitutes an attempt to cover the gap between reliable but time consuming manually created segmentation data and those produced by fast but still unreliable automatic systems.

Segmentation Strategy

The system is based on the *Multi-Level Segmentation* algorithm (MLS) [3] which was originally developed by J.R. Glass and V.W. Zue [3-4]. Speech is considered as a temporal sequence of quasi-stationary acoustic segments, and the points within such segments are more similar to each other than to the points in adjacent segments. Following this viewpoint, the segmentation problem can be simply reduced to a local clustering problem where the decision to be taken regards the similarity of any particular frame with the signal immediately preceding or following it. Using only relative measures of acoustic similarity, this technique should be quite independent of the speaker, vocabulary, and background noise. A joint *Synchrony/Mean-Rate* (S/M-R) model of *Auditory Speech Processing* (ASP), proposed by S. Seneff [5], providing an adequate and efficient basis for phonetic segmentation and labelling, is used as pre-processing module, and in particular, both Envelope and Synchrony Detector parameters are simultaneously considered. Other pre-processing schemes, such as FFT, Cepstrum, or LPC based spectrogram, are supported by SLAM, but Auditory Models (AM) techniques are strongly

suggested. Advantages of using Auditory Model (AM) techniques Vs classical "short-term" analysis approaches for automatic speech segmentation have been shown in literature, especially in adverse conditions [6].

By a recursive technique, involving Euclidean-based similarity measure computation for each target frame, SLAM creates initial adjacent 'seed regions' which constitute the basis for the 'hierarchical structuring' segmentation procedure based on MLS [4]. Using the same framework utilised in building initial seed regions, each region is associated with either its left or right neighbour still using Euclidean-based similarity measures. By keeping track of the distance at which two regions merge into one, a multi-level description usually called *dendrogram*[4] can be constructed. In Table 1 the whole segmentation procedure is illustrated.

Algorithm:

1) **Find boundaries** $\{b_i, \ 0 \leq i \leq N\}, \ t_i < t_j, \ \forall i < j$

2) **Create initial region set**

   $R_0 = \{r_0(i), \ 0 \leq i < N\}, \ r_0(i) \equiv r(i, i+1)$

3) **Create initial distance set**

   $D_0 = \{d_0(i), \ 0 \leq i < N\}, \ d_0(i) \equiv d(r_0(i), r_0(i+1))$

4) **Until** $R_N = \{r_N(0)\} \equiv r(0, N)$

   For any k **such that** $d_j(k-1) > d_j(k) < d_j(k+1)$

   (a) $r_{j+1}(i) = r_j(i), \ 0 \leq i < k$

   (b) $r_{j+1}(k) = merge(r_j(k), r_j(k+1))$

   (c) $r_{j+1}(i) = r_j(i+1), \ k < i < N\text{-}j\text{-}1$

   (d) $R_{j+1} = \{r_{j+1}(i), \ 0 \leq i < N\text{-}j\text{-}1\}$

   (e) $d_{j+1}(i) = d_j(i), \ 0 \leq i < k\text{-}1$

   (f) $d_{j+1}(k-1) = max(d_j(k-1), d(r_j(k-1), r_{j+1}(k)))$

   (g) $d_{j+1}(k) = max(d_j(k+1), d(r_{j+1}(k), r_j(k+1)))$

   (h) $d_{j+1}(i) = d_j(i+1), \ k < i < N\text{-}j\text{-}1$

   (i) $D_{j+1} = \{d_{j+1}(i), \ 0 \leq i < N\text{-}j\text{-}1\}$

   Definitions:
   - b_i is a boundary occuring at time t_i.
   - $r(i, j)$ is a region spanning times t_i to t_j.
   - $r_j(i)$ is the ith region of the jth iteration.
   - $d(i, j)$ is the distance between regions i and j.
   - $d_j(i)$ is the ith distance of the jth iteration.
   - $merge(r(i, j), r(j, k))$ combines two adjacent regions to produce a region $r(i, k)$ spanning times t_i to t_k.
   - The distances $d_j(-1)$ and $d_j(N-j)$ are infinite.

Table 1. Algorithmical structure of multi-level hierarchical segmentation strategy (by J.R. Glass [4], pp. 47).

This hierarchical technique, incorporating some kind of temporal constraint, is quite useful in order to appropriately rank the significance of acoustic events. The final target segmentation can be automatically extracted [8] by appropriate pattern recognition techniques whose aim is that of finding the optimal segmentation path given the dendrogram structure and the target phonemic transcription, but also with minimal human intervention, which is limited exclusively on fixing the vertical point determining the final target segmentation (corresponding to that found on the horizontal line built on this point), and eventually deleting over-segmentation landmarks forced by this choice. Even when using the above described manual intervention, segmentation marks are always automatically positioned by the system and never adjusted by hand. Nevertheless, the manual positioning of segmentation boundaries is always permitted by SLAM whenever this should be requested by the user. As for the computation complexity, MLS algorithm is able to instantaneously analyse speech signal. due to the fact that it does not make use of the entire utterance for emitting segmentation hypotheses but it shows instead a local behaviour.

SLAM Software Implementation

As for the software implementation, SLAM was built using Microsoft C++ and Windows 3.1 SDK software, and runs preferably on Intel 386/486-based personal computers, running DOS 5.00 or higher and equipped with VGA or SuperVGA boards and at least 4 Mbytes of RAM. Only for audio facilities the present implementation makes use of the OROS-AU22 DSP board, but other A/D-D/A hardware could be easily considered.

Signal wave form files can be easily displayed together with their corresponding FFT, LPC, or AM-based spectrogram, energy, pitch and zero crossing files. At the present time, in order to use SLAM, all files should have already been created by other appropriate off-line software, but in the future their on-line creation will be directly included in SLAM. A part from the signal wave form, the user is completely free to visualise any combination of the related files. Various editing operations can be executed on the signal such as LISTEN (only if adequate hardware is available), ZOOM, SCROLL, CUT, PASTE, CLEAR, and COPY, making the system, not only a segmentation and labelling tool, which represents however its most important feature, but also a general speech assessment system. One important feature of SLAM, considered as a simple speech visualising system, is represented by the possibility to move the mouse within the various windows and to instantaneously visualise the corresponding values of active representations, such as signal amplitude or time position, energy, pitch or frequency. In order to segment and label speech signals, their corresponding spectral representation (FFT, LPC, AM based) is visualised by SLAM. On the basis of the chosen spectral information, the MLS algorithm can be applied in order to create various signal alignment hypotheses and the user can easily choose the best by using the mouse and clicking in any position within the dendrogram structure (see Figure 1). An incomplete list of SLAM function is illustrated in Tab. 2.
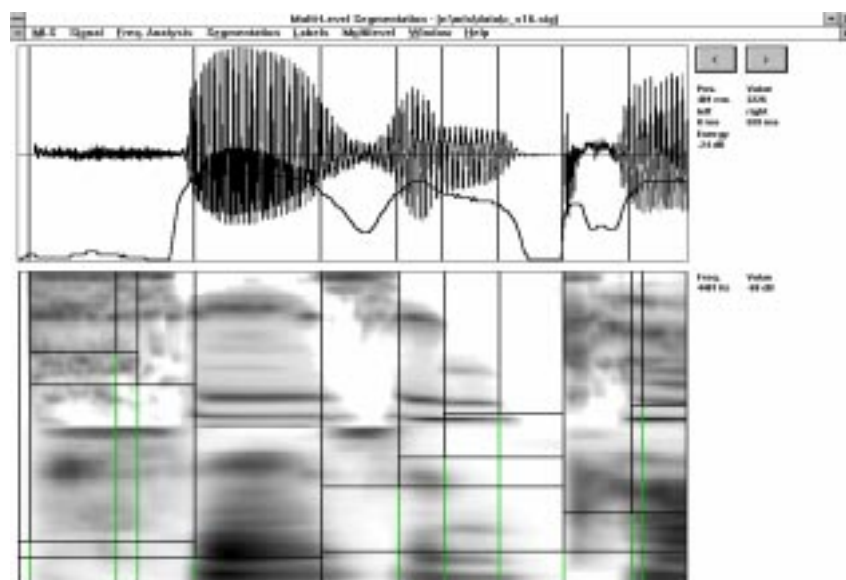
Figure 1. SLAM plot referring to the English sentence "Susan ca(n't)" uttered by a female speaker. Time wave form, Energy and final segmentation are plotted in the top, while AM spectrogram and its corresponding dendrogram are illustrated in the bottom.

SLAM Function list

- OPEN, CLOSE, and SAVE (portions) SIGNAL, ENERGY, PITCH, ZERO-CROSSING type files;
- OPEN, CLOSE, and SAVE (portions) FFT, LPC, AM based spectrogram type files;
- OPEN, CLOSE, and SAVE (portions) SPECTRAL-CHANGES and MULTI-LEVEL type files
  (see MLS algorithm, Tab 1);
- OPEN, CLOSE, and SAVE (portions) SEGMENTATION and LABELLING type files;
- LISTEN (only if hardware is available);
- ZOOM, SCROLL, CUT, PASTE, CLEAR, and COPY portions of signal;
- COMPUTE SPECTRAL-CHANGES ;
- COMPUTE MULTI-LEVEL structure;
- SET MLS Δ parameter (see MLS algorithm, Tab. 1);
- SET ANALYSIS PARAMETERS;
- ACCEPT SEGMENTATION from SPECTRAL-CHANGES;
- ACCEPT SEGMENTATION from MULTILEVEL;
- MANUAL SEGMENTATION (insert, delete, move);
- LABELLING (insert, delete, modify);
- ....(the present list is not complete)

Tab. 2 SLAM funcion list (this is only a limited set of functions).

The performance of the SLAM segmentation system when applied to a simple but significant segmentation task is reported in [6].

The user can also manually add new markers, besides those explicitly set by choosing a particular alignment hypothesis based on the dendrogram structure, in case of under-segmentation, or delete some markers in case of over-segmentation.

The use of AM versus FFT-based spectrogram greatly reduces this kind of manual intervention [6] thus emphasising the importance of using an adequate signal representation when dealing with speech segmentation, especially in noisy environment. A labelling capability is also included in SLAM where SAMPA [8] labels can be attached to each segmentation mark or modified by the user. Since Windows 3.1 MDI (Multiple Document Interface) standard was adopted in building SLAM, it is possible to open more than one window in order to visualise multiple signals and their related parameters, as well as to open more than one segmentation session, as illustrated in Fig. 2. The only limitation is given by the available amount of RAM.

Conclusions and Future Trends

SLAM's main feature, a part from performance [6], is its user-friendliness and given the great amount of speech data-bases this characteristic is very important for any useful segmentation system. In order to reduce manual intervention, SLAM will be transformed in a completely automatic segmentation and labelling system such as the one used in [7] leaving the best segmentation hypothesis to the system and permitting a human intervention in case of system errors. AIDA new Italian speech data-base [9] will represent a good test material for SLAM.
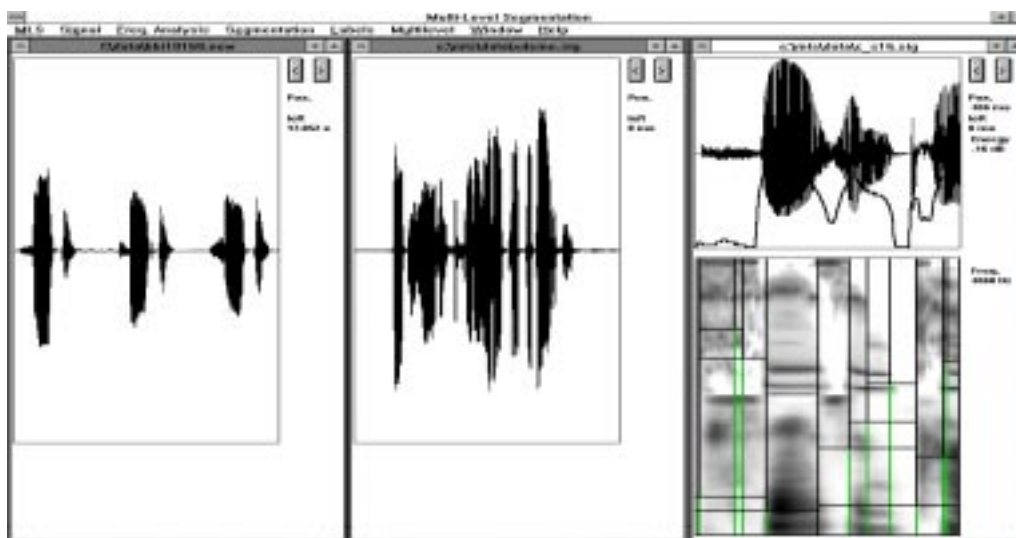
Acknowledgements

Figure 2. Use of SLAM with three simultaneous opened sessions

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