The 12th WSEAS International Conference on
MATHEMATICAL and COMPUTATIONAL METHODS in SCIENCE and ENGINEERING
(MACMESE '10)
University of Algarve, Faro, Portugal, November 3-5, 2010

Plenary Lecture:

Softcomputing Methodologies Applied to Audio-Based Information Retrieval

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Premises

- **Intuitive** and **efficient** access to multimedia information **is becoming a strategic option**, given the increasing availability of such information in large archives and on the web.

- **Audio information** is a **powerful medium** to communicate naturally with such systems, while accessing multimedia information semantically.

Premises (cont.)

- **Integrating multiple signal-processing algorithms and soft computing** is a **new approach** toward the development of an audio-based front end for multimedia retrieval.

- **Algorithm-based features extraction**, **artificial neural** networks used as pattern matchers, and **fuzzy-logic** used like classifiers, lead to the development of a content-based, audio-access system capable to retrieve information in a multimedia domain.

Premises (cont.)

- **Audio is an information medium** capable of embedding much more information than we tend to imagine.
- A generic **audio source** (e.g. phone ring, bell sound, people talking, etc.) **embeds information** that is typically overlooked but can easily be used as a key for media retrieval.
- For example, searching a film for a crowd segment is simpler and more effective if we **search for audio** rather than scene or title.

Premises (cont.)

- **Current search-engine** implementations are **very smart at retrieving text-based information** (e.g. web pages, documents, files in which text information is available)
- **Current search-engines are wanting in their ability to locate multimedia information**, especially audio and audio-related information.

Premises (cont.)

- **Two main requirements arise** in building a multimedia search engine:
  - **client-side ability to extract features from audio** and to transform text into audio features
  - **server-side ability to match and find those features** amidst vast, distributed multimedia information.

Premises (cont.)

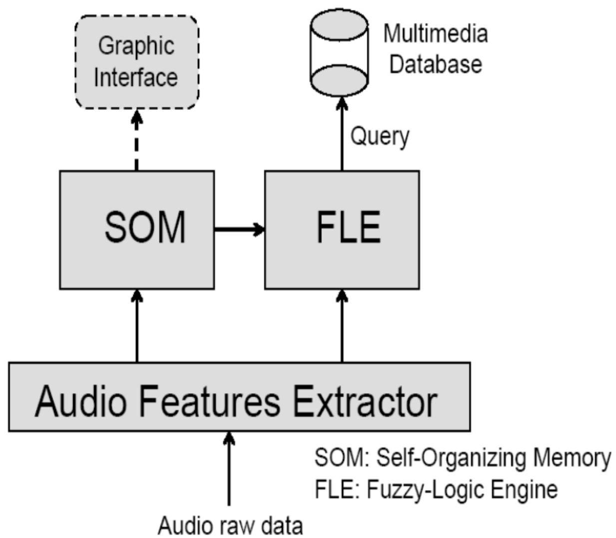
- **Human audio perception is fuzzy**, as are audio features: an exact match between query and target is often impossible.

- **Artificial Neural-network feature classifiers** have proven optimal in automatically indexing digital audio collections.

- **Fuzzy logic** has also been applied to audio classification tasks. Such classification is complementary to Artificial Neural-network based pattern matching.

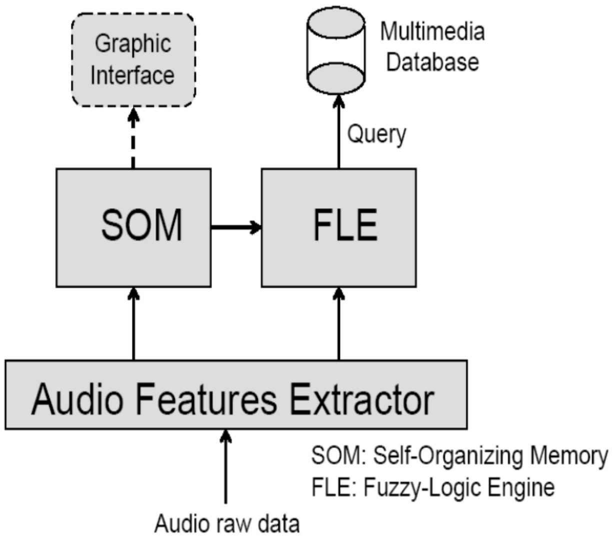
System framework

- The system consists of:
  - **an audio-feature extractor (AFE)**
  - **an artificial neural network-based classifier (ANN)**
  - **a fuzzy-logic inference engine (FLE)**



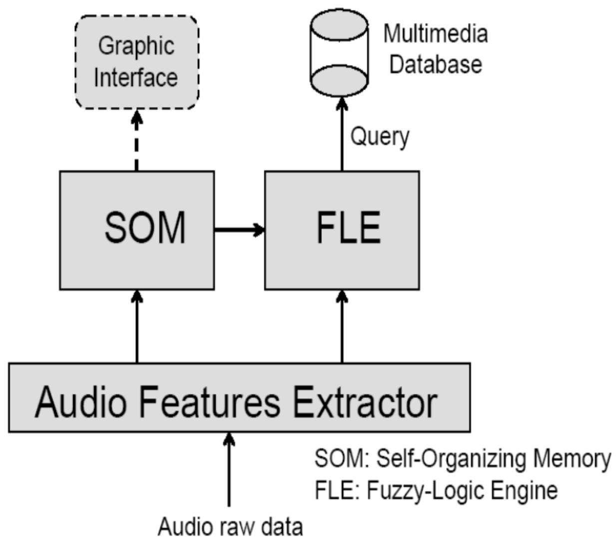
System framework (cont.)

- **Audio features are fed to the ANN-based classifier** that identifies the class the audio belongs to.



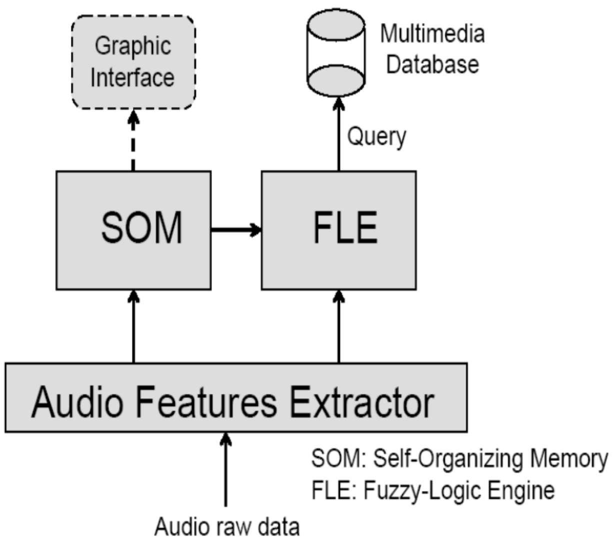
System framework (cont.)

- **The fuzzy-logic inference engine generates a smart query** to access an audio repository in search mode.



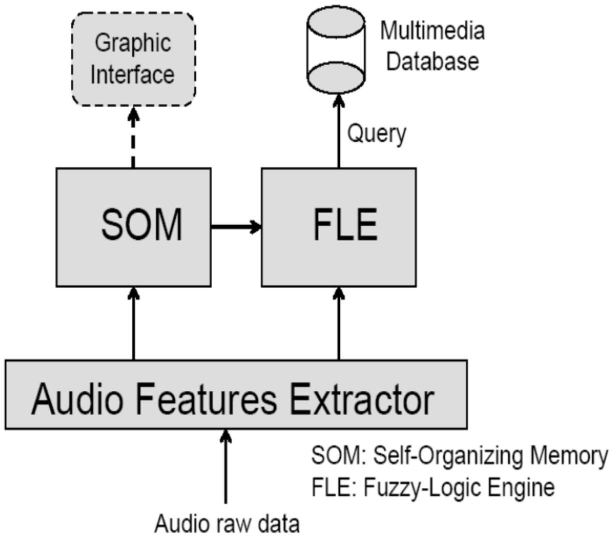
System framework (cont.)

- The audio-feature extractor consists of a set of **digital signal-processing algorithms applied to raw audio data** (low-level (physical), time-domain features and frequency domain features).



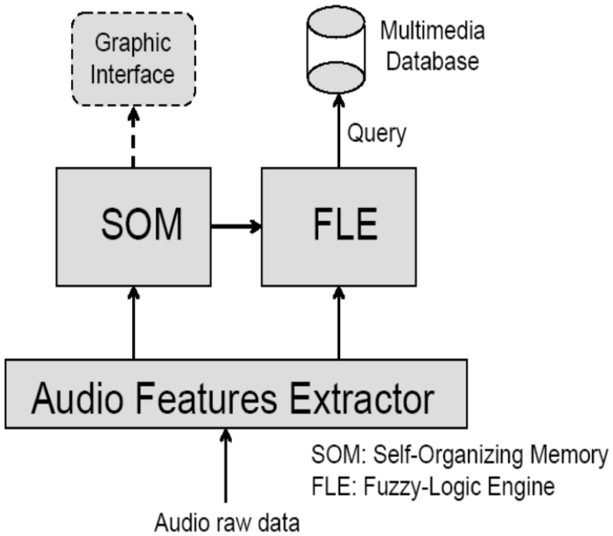
System framework (cont.)

- The **ANN-based classifier maps** the multidimensional space of audio features onto two-dimensional space to cluster information about features.
- **Clustered audio features** represent the data for the fuzzy-logic inference engine to classify.



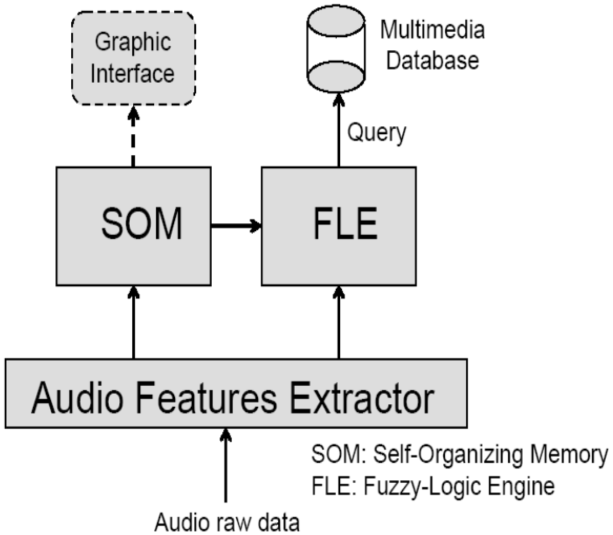
System framework (cont.)

- The **fuzzy-logic inference engine classifies** clustered data at the ANN output layer by applying a set of fuzzy rules and membership functions.



System framework (cont.)

- A **similarity query** is then generated fuzzily.



Audio-feature Extraction

- **Time-domain audio features** are calculated according the following general formula:

$$Q(n) = \sum_{m=0}^{N-1} T\left[s(m)\right] w(n-m)$$

- *s(n)* is the audio signal
- *Q(n)* is a short-time sampled calculation of a feature
- *T* is the transformation function applied to signal *s(n)*
- *w(n)* is the windowing function for short-time feature calculation
- window size is 20 ms (N samples for a given sampling rate).

Audio-feature Extraction (cont.)

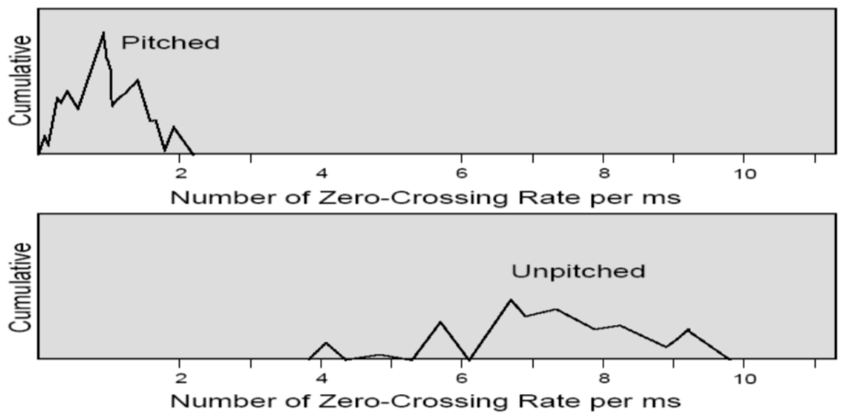
- **Root mean square (RMS)**

$$RMS(n) = \sqrt{\frac{1}{N}\sum_{m=0}^{N-1} s^2(m)}$$

Audio-feature Extraction (cont.)

- Zero-crossing rate (ZCR)

$$ZCR(n) = \sum_{m=0}^{N-1} 0.5 \left| sign(s(m)) - sign(s(m-1)) \right| w(n-m)$$

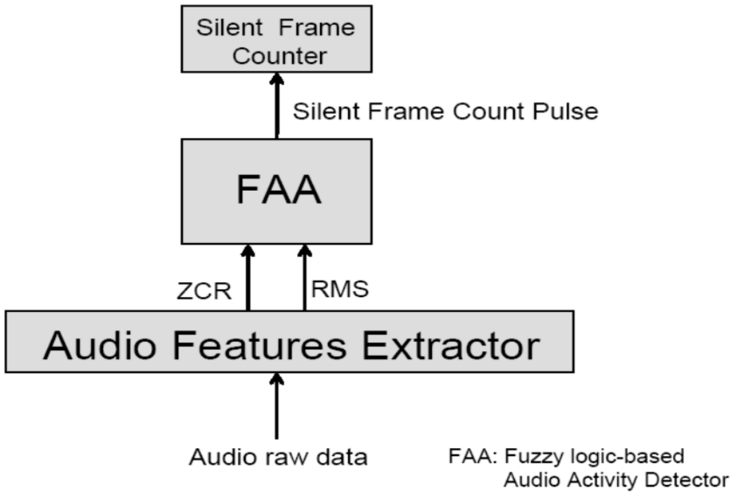


Audio-feature Extraction (cont.)

- An additional computed feature is the silent frame rate (SFR)

SFR = silent frames/total frames

- Fuzzy-logic calculation of the silent frame rate (SFR) audio feature.



Audio-feature Extraction (cont.)

- **Frequency-domain audio features** are calculated according to short-time Fourier analysis formula :

$$S_n(e^{j\omega}) = \sum_{m=0}^{N-1} s(m)e^{j\omega m}w(n-m)$$

$S_n(e^{j\omega})$ is a short-time computation of audio-signal energy *s(m)* in a limited bandwidth related to the chosen frequency.

Audio-feature Extraction (cont.)

- The calculated frequency-domain audio features are frequency centroid (FC) and cumulative band energy (CBE).

- FC is the balanced point of the spectrum, calculated as follows:

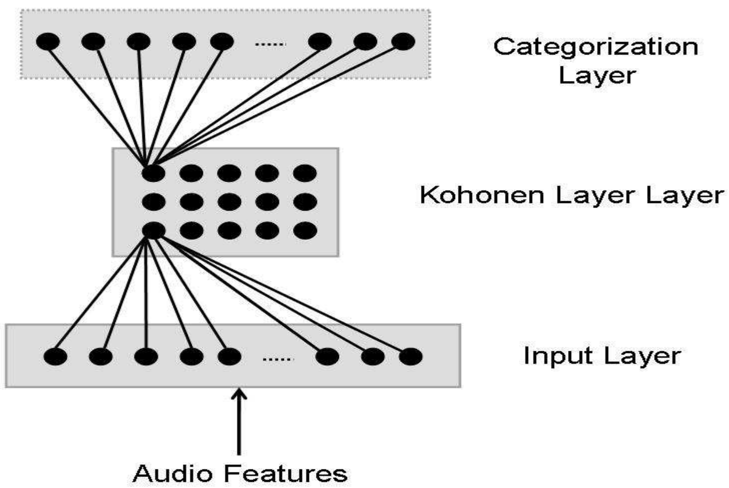
$$\omega_c = \frac{\int_0^{\omega_0} \omega \left| S(\omega) \right|^2 d\omega}{\int_0^{w_0} \left| S(\omega) \right|^2 d\omega}$$

- CBE is attained with the above formula repeating the calculation of energy at various frequency to cover four sub-bands:

$$B_1 = [0, \frac{\omega_0}{8}], B_2 = [\frac{\omega_0}{8}, \frac{\omega_0}{4}], B_3 = [\frac{\omega_0}{4}, \frac{\omega_0}{2}], B_4 = [\frac{\omega_0}{2}, \omega_0]$$

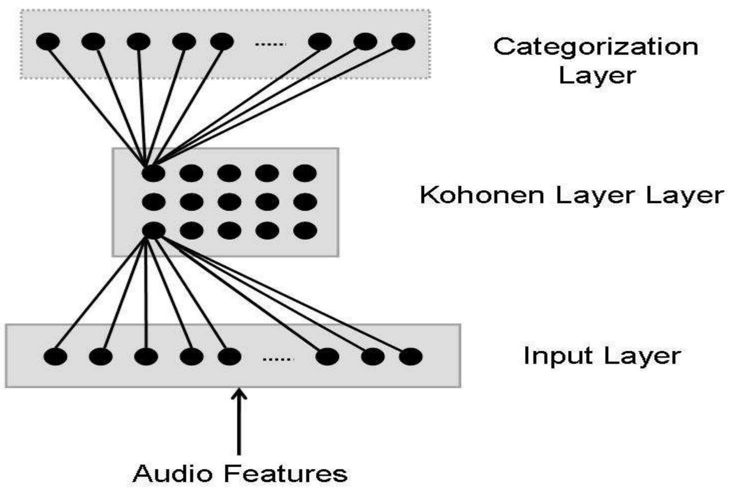
Sound-feature mapping

- The Kohonen feature-map (KFM) artificial neural network (ANN) was used to **map multi-dimensional space onto two-dimensional space**.



Sound-feature mapping

- A KFM can map n-dimensional input-vector space onto a neuron layer where neurons are organized according to similarities in input values.
- This ability has been successfully used to map speech sounds onto phonetic space for a high-performance implementation of speech recognition (phonetics-driven speech-to-text).



Sound-feature mapping (cont.)

- Euclidean distance was used to determine the winning node in the map:

$$D_i = \left| X - W_i \right| = \sqrt{(x_1 - w_{i1})^2 + (x_2 - w_{i2})^2 + \ldots + (x_M - w_{iM})^2}$$

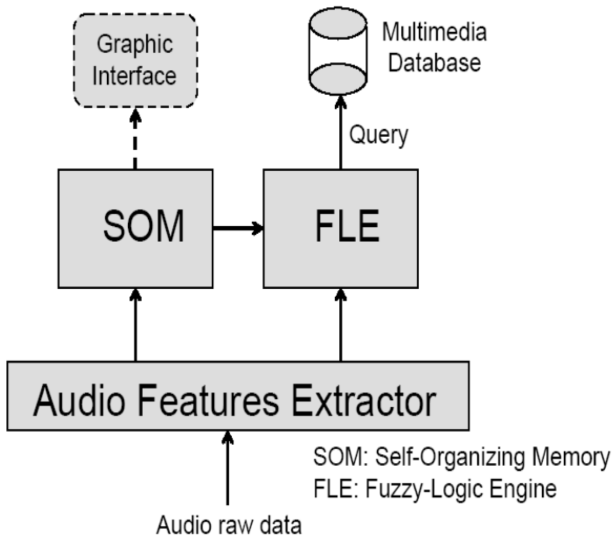
- When a node wins more than 1/N times (N is the number of Kohonen nodes), its distance is adjusted upward to attenuate its chance to win.

Sound-feature mapping (cont.)

- For nodes that win less than 1/N times, the distance is adjusted downward to make them more likely to win.
- The distance adjusting factor is: $B_i = g(1/N - F_i)$
- The adjusted distance D'_i is computed as: $D'_i = D_i - B_i$

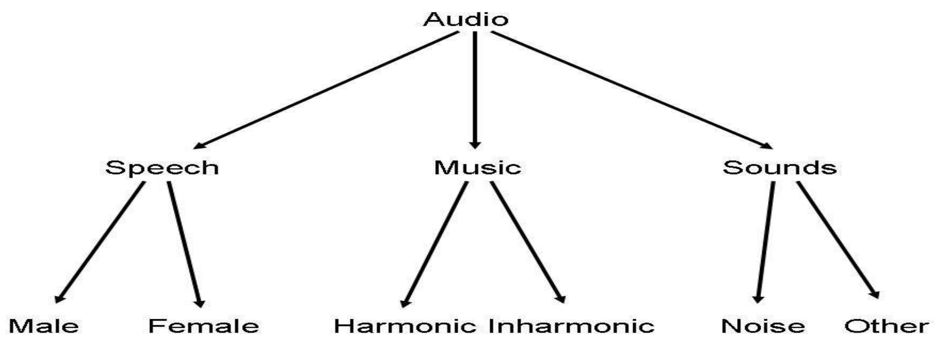
Fuzzy-logic KFM categorization

- To categorize the KFM's audio-feature mapping ability, an **upper layer is added to the Kohonen layer**. The upper layer consists of a fuzzy-logic engine (FLE) tuned to categorize sounds into types.



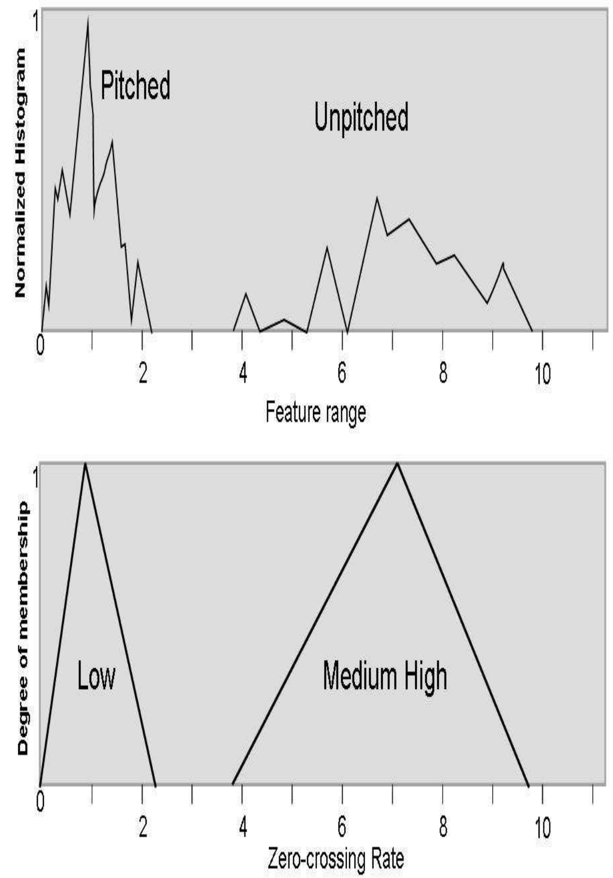
Fuzzy-logic KFM categorization

- Several important issues need to be resolved to set up the fuzzy rules and the membership function so that audio information can be classified in a hierarchical fashion and used for fast and effective search in the multimedia database:



Fuzzy-logic KFM categorization (cont.)

- Crisp information from the KFM layer and from certain measured audio features for the given sound class to be categorized is fuzzified.
- Each membership function is derived by looking at the statistics for each feature and how it is clustered by the KFM.
- A membership function is then derived from the shape of the feature's distribution, simply by superimposing membership shape on the distribution shape



Fuzzy-logic KFM categorization (cont.)

- The rule model is: **IF (Condition 1) AND (Condition 2) THEN (Category)**
- *Condition 1* and *Condition 2* are fuzzy evaluations of one feature in the audio-measurement domain and one in the KFM-mapping domain.
- *Condition* uses a fuzzy measurement derived from the membership function in terms of qualitative grade scale (e.g., very low, low, medium, high, very high) to represent a fuzzy measurement of the feature (e.g. *RMS is medium, ZCR is low*, etc.).
- For each audio category a set of AND rules are generated.
- A singleton function is used to defuzzify each audio object, thus determining its degree of belonging to an audio category.

Fuzzy-logic KFM categorization (cont.)

- The fuzzy-logic engine needs to be tuned for best performance. Two options are available for the purpose: manual tuning or automatic tuning.

- Manual tuning relies on an audio expert, who chooses among different membership functions. The audio expert may also create rules for best categorizing audio, based on her or his knowledge. A graphic user interface (GUI) is helpful for this task.

- Automatic tuning uses only a triangular membership function to fit the audio-feature distribution shape and fixed format rules. Automatic tuning can also be assisted by a genetic-like process, so that a large number of rules are generated at tuning-time, but only those used most often are kept at run-time.

Thank you for your attention (any question?)

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