Sphinx 3 for the Java™ platform
Architecture Notes

Overview

This is a living document that describes the architecture of Sphinx 3 for the Java™ platform (sphinx3j). The original form of the architecture was derived as a result of the face−to−face meeting of the sphinx3j team at Sun Microsystems Laboratories on February 21st and 22nd, 2002. The members of the sphinx3j team present at the meeting are as follows:

Evandro Gouvea,	Carnegie Mellon University
Philip Kwok,	Sun Microsystems Laboratories
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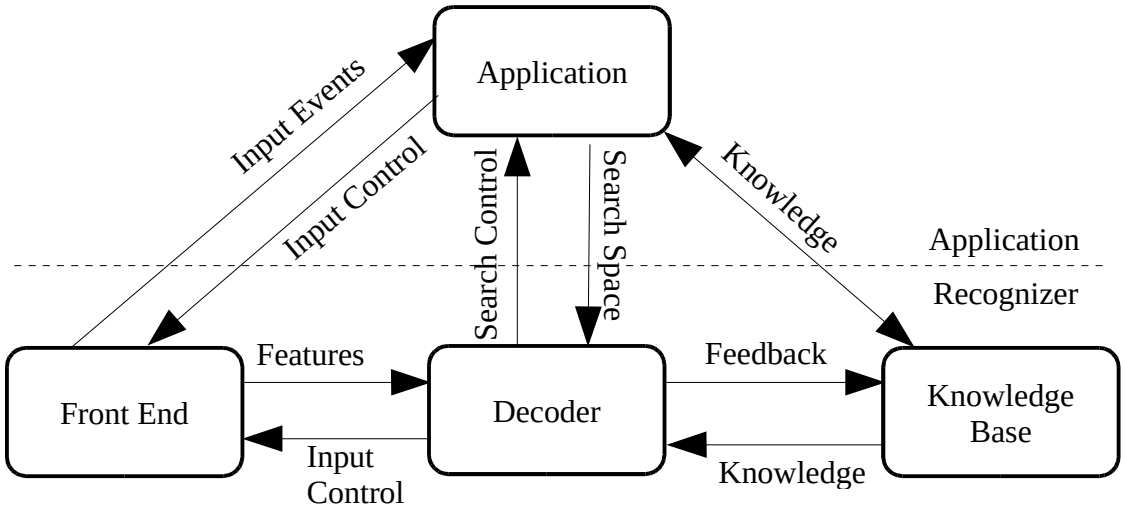
Since this is a living document, we will update it as work on sphinx3j progresses.

High Level Architecture

The high level architecture for sphinx3j is relatively straightforward. As shown in the following figure, the architecture consists of the front end, the decoder, a knowledge base, and the application.



The front end is responsible for gathering, annotating, and processing the input data. In addition, the front end extracts features from the input data to be read by the decoder. The annotations provided by the front end include the beginning and ending of a data segment. Operations performed by the front end include preemphasis, noise cancellation, automatic gain control, end pointing, etc.

The knowledge base provides the information the decoder needs to do its job. This information includes the acoustic model and the language model. The knowledge base can also receive feedback from the decoder, permitting the knowledge base to dynamically modify itself based upon successive search results. The modifications can include switching acoustic and/or language models as well as updating parameters such as mean and variance transformations for the acoustic models.

The decoder performs the bulk of the work. It reads features from the front end, couples this with data from the knowledge base and feedback from the application, and performs a search to determine the most likely sequences of words that could be represented by a series of features. The term "search space" is used to describe the most likely sequences of words, and is dynamically updated by the decoder during the decoding process.

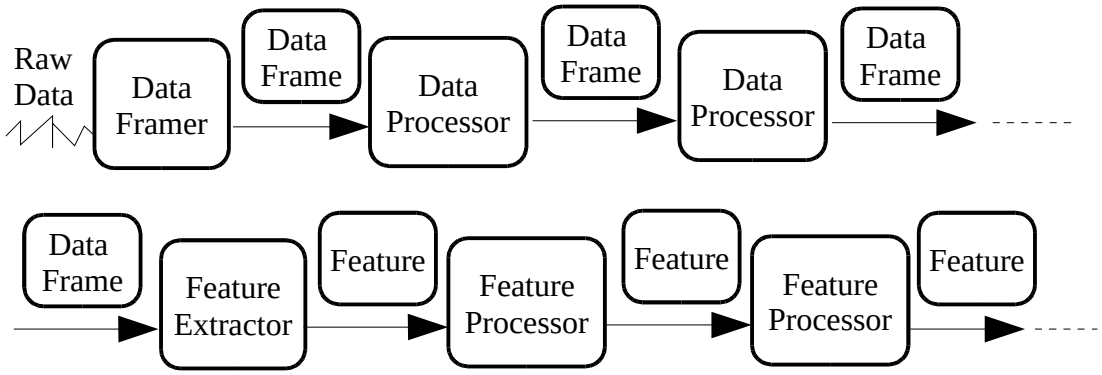
Unlike many speech architectures, the sphinx3j architecture allows the application to control various features of the speech engine, permitting more sophisticated speech application development. As depicted in the previous figure, the application can receive events from the front end and can also provide some level of control over the front end. The type of control can be as simple as turning the audio input on or off, but may also include more sophisticated operations.

During the decoding process, the application may also receive events from the decoder while the decoder is working on a search. These events allow the application to monitor the decoding progress, but also allow the application to affect the decoding process before the decoding completes. Furthermore, the application can also update the knowledge base at any time.

The following sections describe each piece of the high level architecture in more detail.

Front End

The front end can be broken in several simple pieces as depicted in the following illustration:



Examining the illustration left to right, top to bottom, the front end first reads in raw data via the Data Framer. This data can arrive by a stream, a file, or any other means. In addition, while the data is typically audio, there can be a number of simultaneous data sources, such as both video and audio. The Data Framer packetizes the data into annotated Data Frames. These Data Frames contain information about the data packets, including information such as if the data is the beginning or end of a segment.

The front end passes the Data Frames to a series of Data Processors. The Data Processors perform successive modifications to the Data Frames, such as automatic gain control, noise cancellation, down/up sampling, and preemphasis.

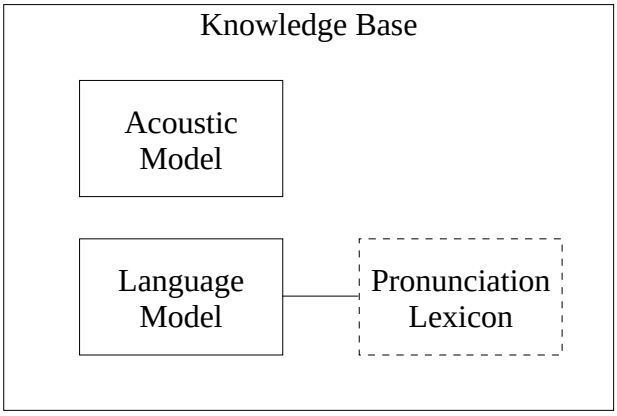
Once the preprocessing of a Data Frame is complete, the front end passes the Data Frame to a Feature Extractor. The Feature Extractor extracts the feature(s) necessary for the decoder to do its work. For audio, this typically involves obtaining cepstral and delta cepstral information, but can be anything the decoder accepts. Furthermore, the resulting Feature is not necessarily restricted to one data type. For example, the Feature may contain information for both audio and video.

The front end then passes Feature frames to to a series of Feature Processors. The Feature Processors may perform a number of operations including end pointing, noise cancellation, and cepstral mean calculation.

When the front end has completed processing a Feature, it passes it to the decoder, which is described later on this document.

Knowledge Base

The knowledge base provides the information the decoder needs to do its job. Typically, the knowledge base consists of the acoustic model and the language model.



The acoustic model provides the knowledge for converting Feature sequences into unit hypotheses, and the language model provides the knowledge for converting unit sequences into word and word sequence hypotheses.

Language Model

We're still creating the language model portion of sphinx3j, but at a minimum, sphinx3j will support three type of language models. The simplest type of language model is no language model at all, and is used in the case of isolated word recognition. The next common language model type is a context free grammar that can used in speech applications based on command and control. The final language model type is based on n−gram grammars, and is typically used for free form speech.

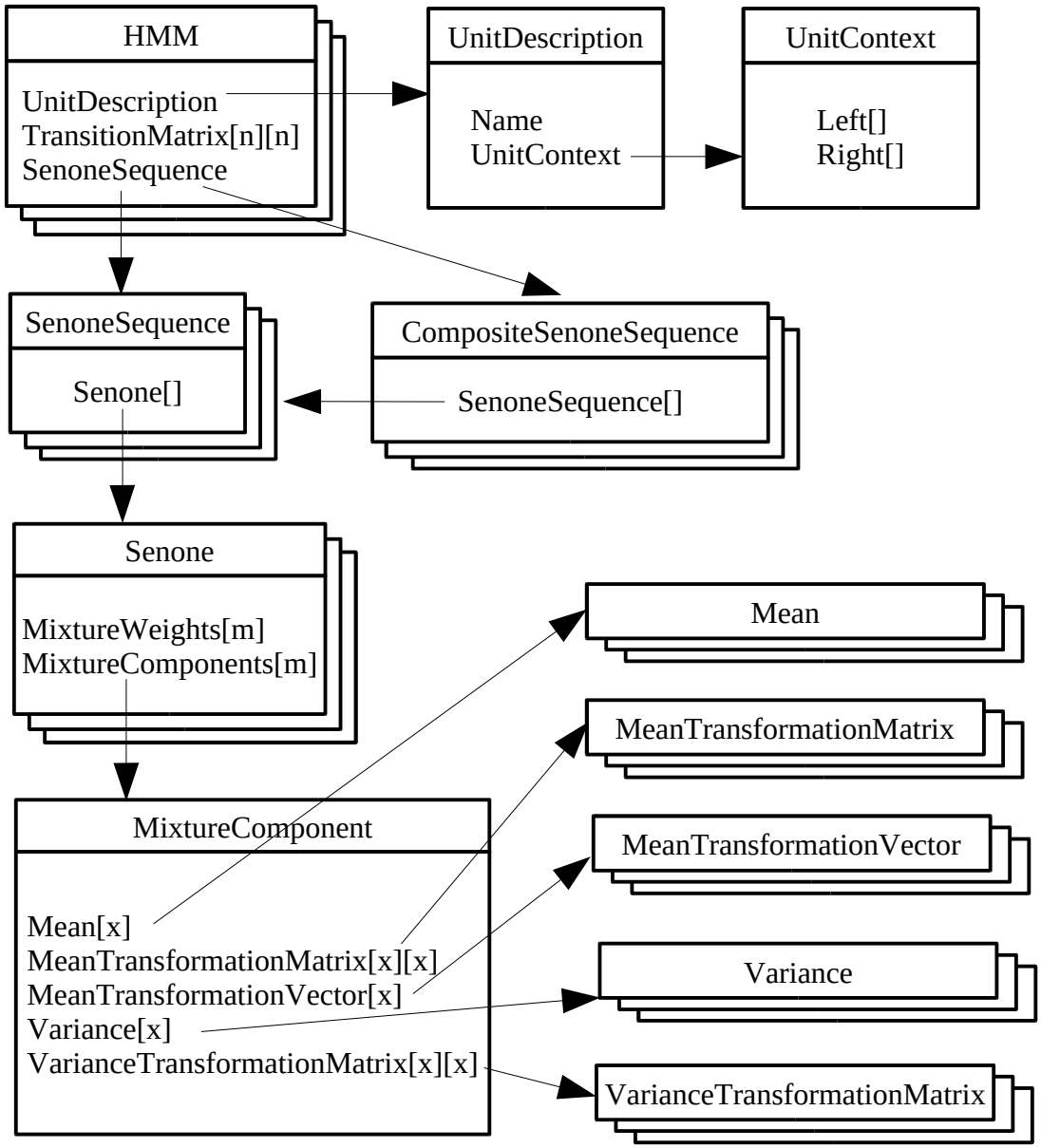
Associated with both the context free and n−gram grammar language models is a pronunciation lexicon that maps unit sequences to words in the lexicon. The meaning of a

unit varies depending upon the task. For example, for isolated word recognition, the unit could be a whole word. For large vocabulary continuous recognition, the unit could be a triphone.

As stated, we're still working on the format of the language model. As a result, this section is expected to change over time.

Acoustic Model

In sphinx3j, the acoustic model consists of a set of left to right Hidden Markov Models (HMMs), with one HMM per unit. The following diagram illustrates the definition of the HMMs of the acoustic models in sphinx3j:



In the drawing, any object shown as a "stack of cards" represents a shared pool of object instances. For example, there is a shared pool of Senones that are referred to by SenoneSequences. The set of shared pools allows the sphinx3j HMMs to support concepts known as "state tying" and "parameter tying." With state and parameter tying, the HMMs can share a large variety of features. There are at least two reasons for doing tying: the primary reason is to get sufficiently trained models, and the second reason is to help reduce the number of calculations during the search.

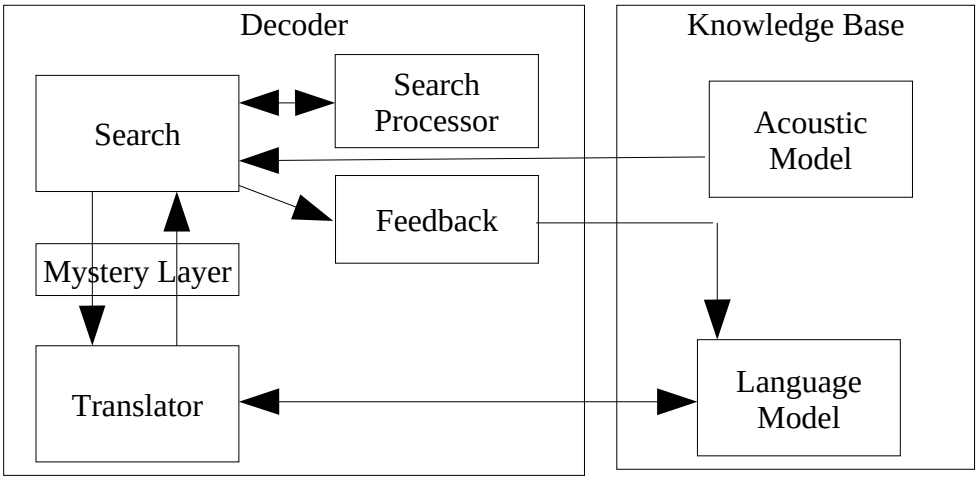
Each state of the HMMs in sphinx3j are called "Senones." Because the HMMs are left to right HMMs, the states can be represented as a sequence of senones. Because several HMMs can share the same senone sequence [[[WDW − give example]]], the HMMs point to a SenoneSequence that comes from a shared pool of SenoneSequences. In some HMMs [[[WDW − give example]]], the SenoneSequence of an HMM is a special CompositeSenoneSequence as illustrated.

Each SenoneSequence contains an ordered list of Senones, where each Senone comes from a shared pool of Senones (i.e., the states of the HMMs in sphinx3j are tied). As stated previously, each state in the sphinx3j acoustic model HMMs is Senone. The Senones are based on probability density functions (pdfs). As shown in the illustration, the pdfs are continuous Gaussian Mixtures. The exact type of pdf, however, does not have to be a Gaussian Mixture. Instead, the pdf merely needs to be able to take a Feature from the front end and return a score.

As illustrated by the MixtureComponent, each GaussianMixture obtains its parameters from several shared pools (e.g., the GaussianMixture parameters are tied).

Decoder

We are still working on the architecture for the decoder. The current decoder in Sphinx 3 combines a LexTree with an n−Gram language model. We feel this is too specific to include in sphinx3j, and are working on a flexible model that permits the Sphinx 3 model as well as other types of searches. At a minimum the other types of searches we are looking at include support for isolated word recognition, context free grammars, and n−gram language models. The current model thoughts are as follows:

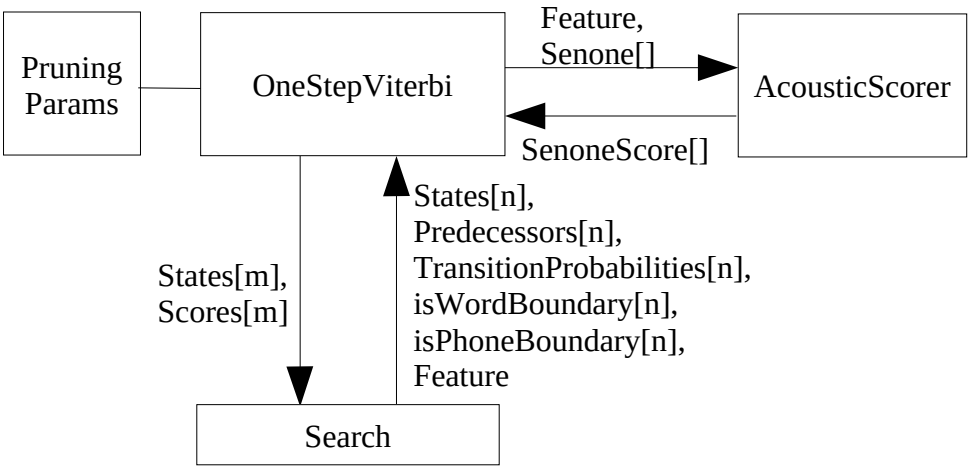


The basic idea behind this model is to allow for more flexibility when it comes to the search strategy. In this model, the decoder provides a translator between the language model of the knowledge base and the search algorithm of the decoder. The search reads information from the acoustic model, combines this with the translated language model, and passes the information off to a search processor. As the search gets results, it can pass feedback back to the language model via the feedback block. This allows the language model to be dynamically updated over time.

We're currently considering a very simple Viterbi−based search processor that we've affectionately named the "One Step Viterbi."

OneStepViterbi

Our current thoughts on the one step viterbi are that it accepts a set of states and transition probabilities, and returns a set of scores and unpruned states.



The OneStepViterbi accepts a set of n States along with a Feature. Each State represents a state from an HMM and has associated with it a set of predecessor states, transition probabilities, and whether the state represents a phone or word boundary. Note that in acoustic models with tied states, the states may share Senones. The OneStepViterbi determines the set of unique Senones and passes them off to the AcousticScorer along with the Feature for scoring. The AcousticScorer returns a score for each Senone.

After the OneStepViterbi receives the scores from the AcousticScorer, the OneStepViterbi may apply various pruning methods. Once it has pruned the various states, the OneStepViterbi returns a set of unpruned states with their associated scores.

The main ideas behind the OneStepViterbi are that it need not know about the language model and it also doesn't need to maintain a large search space.