

# Create conversational agents for Android

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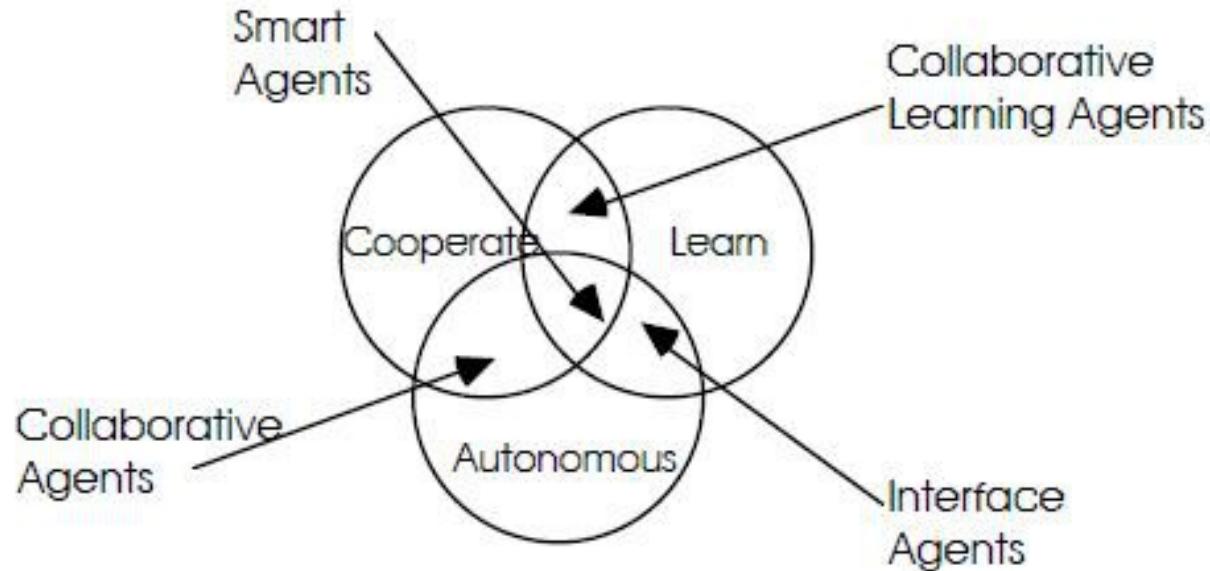
# Outline

- Definition of Conversational Agent
- Examples of agents
- How to realize it: a possible architecture
- The AT&T Speech Mashup Service
- What's AT&T Speech Mashup
- AT&T Architecture
- AT&T Speech Mashup Web Portal
- Web Portal functionalities
- Into details: Grammars and SSML Markup
- API and Clients developing
- What is a dialog flow
- “Hello Lab” tutorial for Android

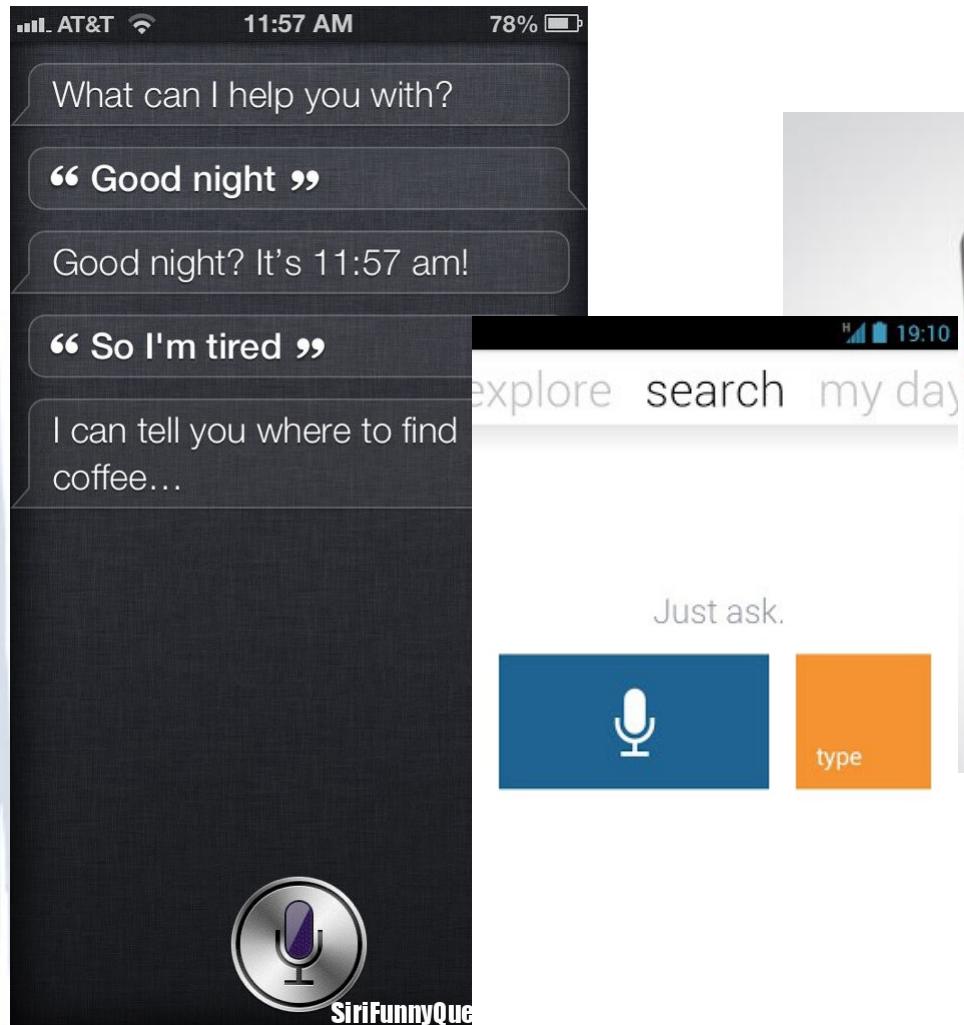
# Definition of Conversational Agent

An agent is a system to which the user can delegate the execution of his tasks. It has at least 4 main properties:

1. Autonomy
2. Reactivity
3. Pro-activeness
4. Social ability



# Examples of agents

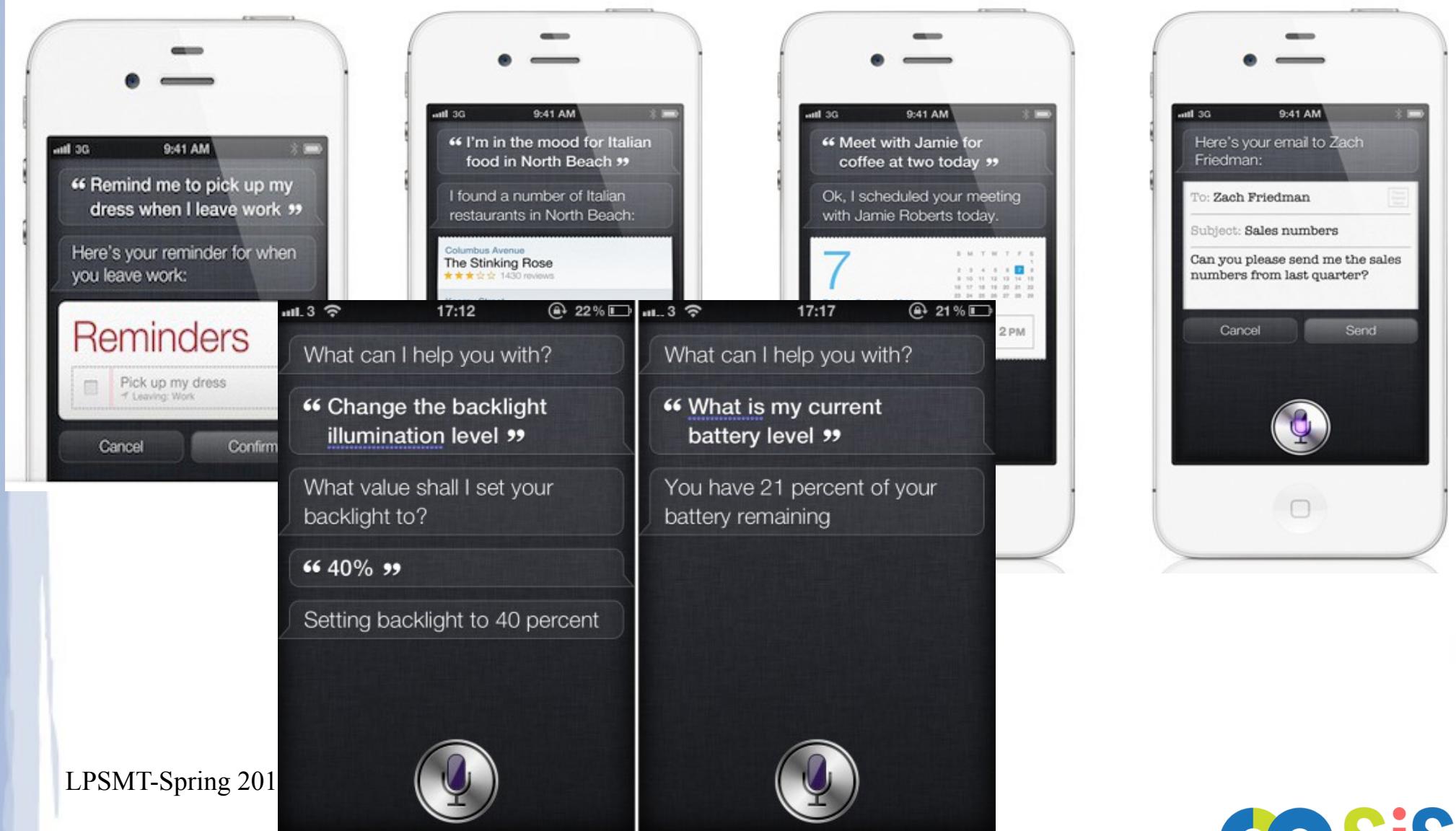


[Video Examples](#)

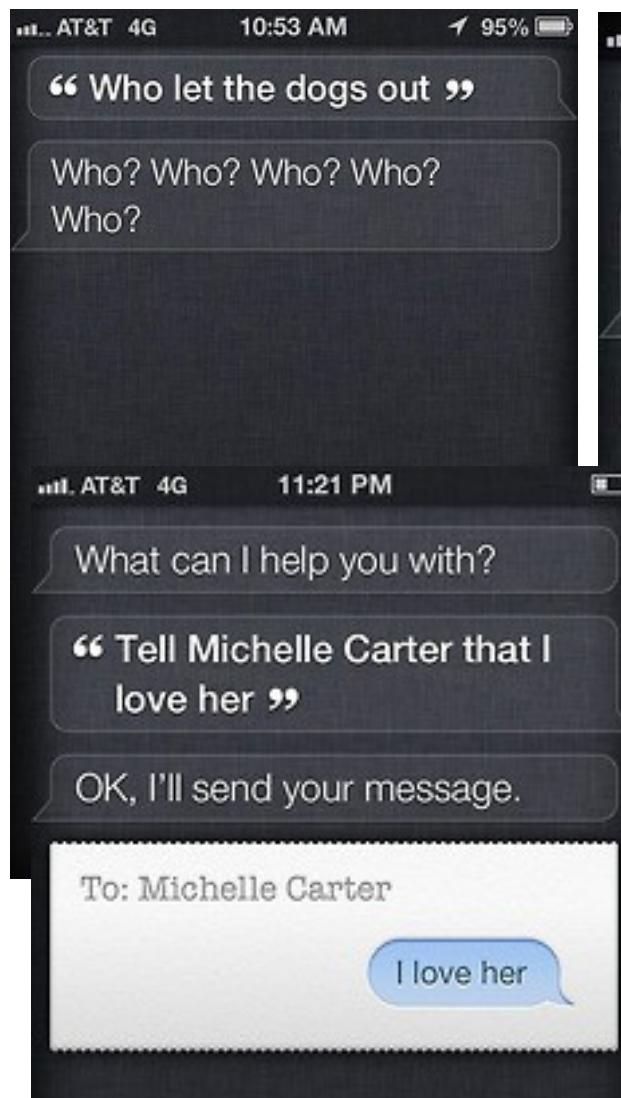
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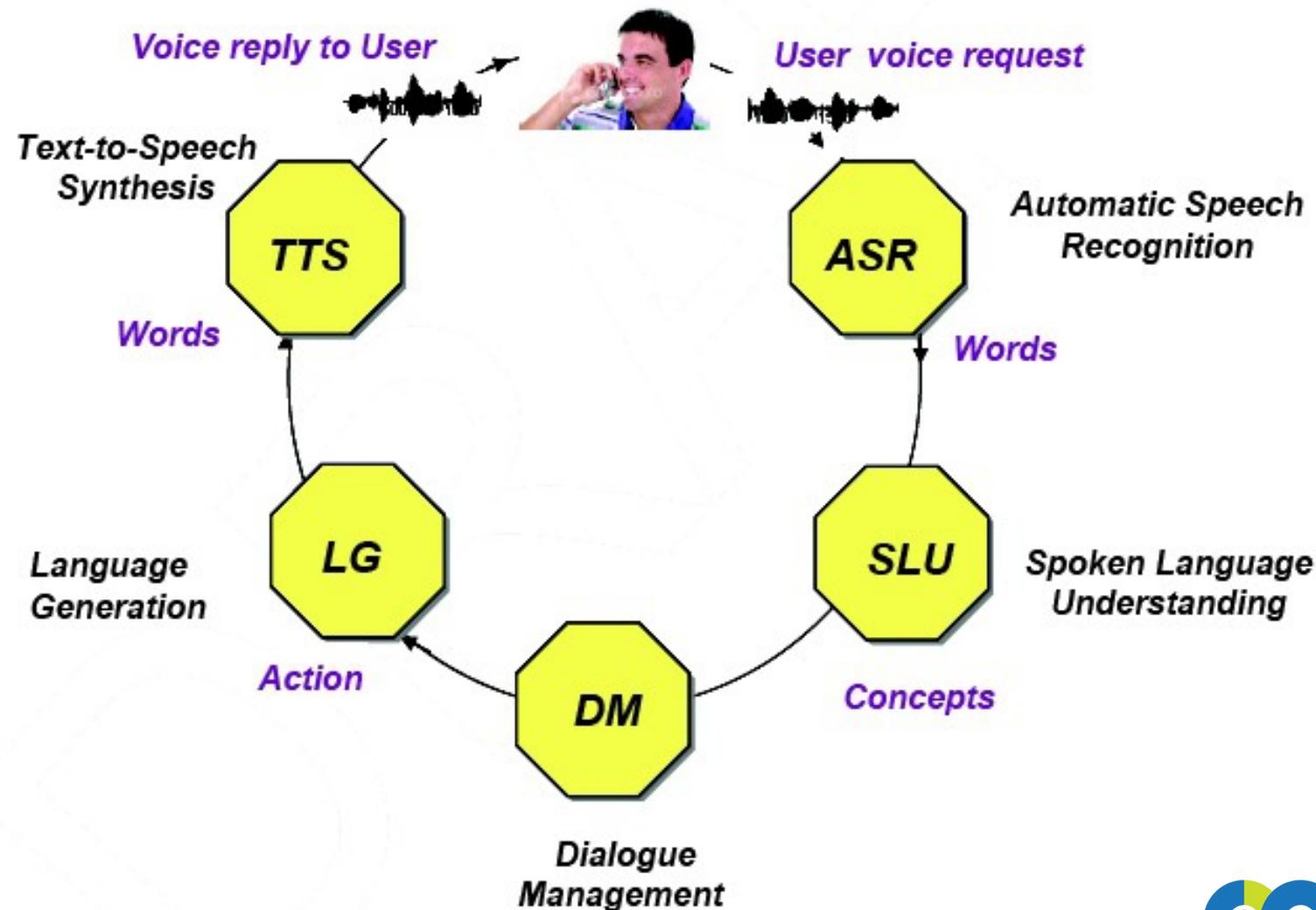
# Examples of agents



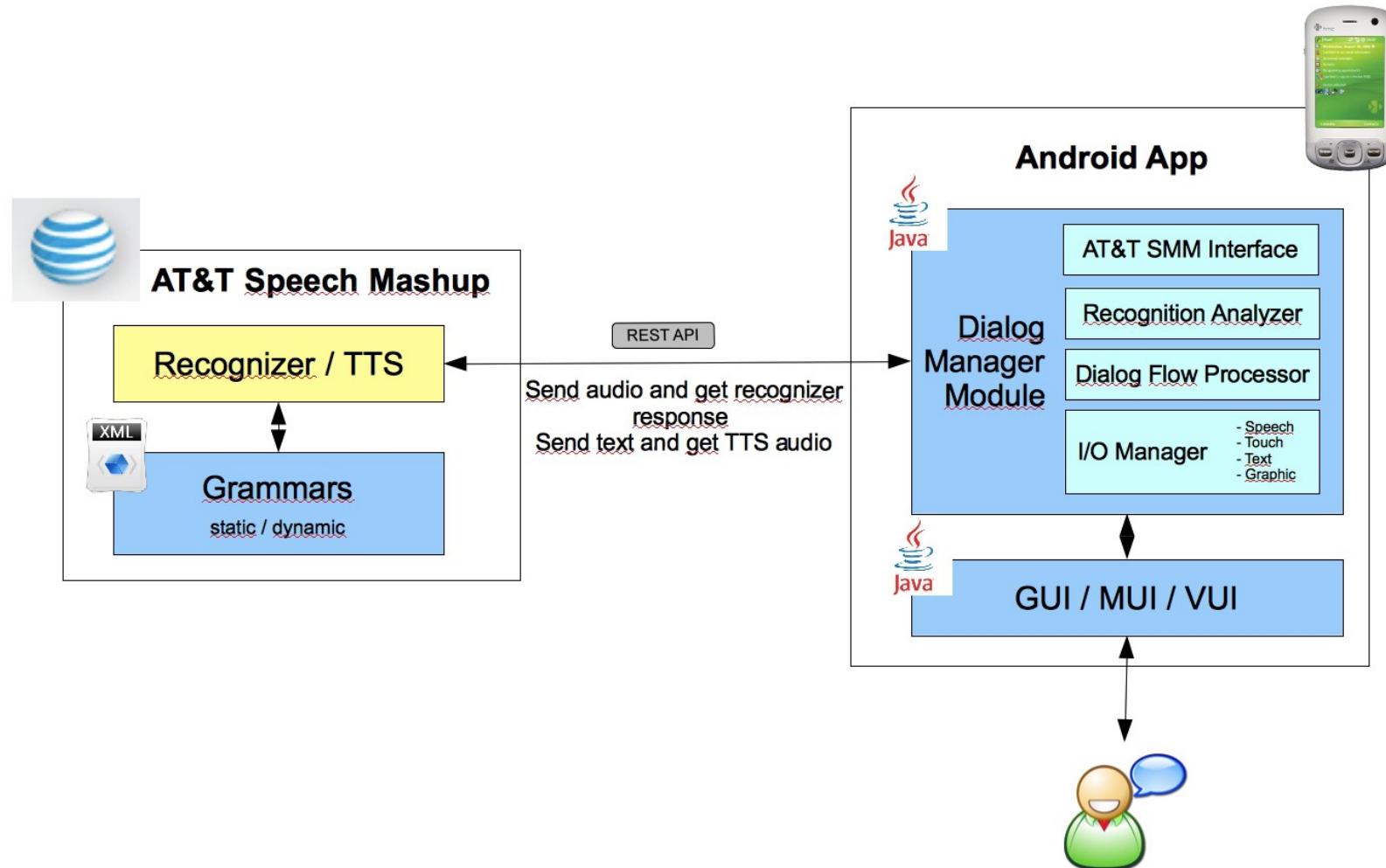
# Funny examples of agents



# Basic architecture of a generic Spoken Dialogue System

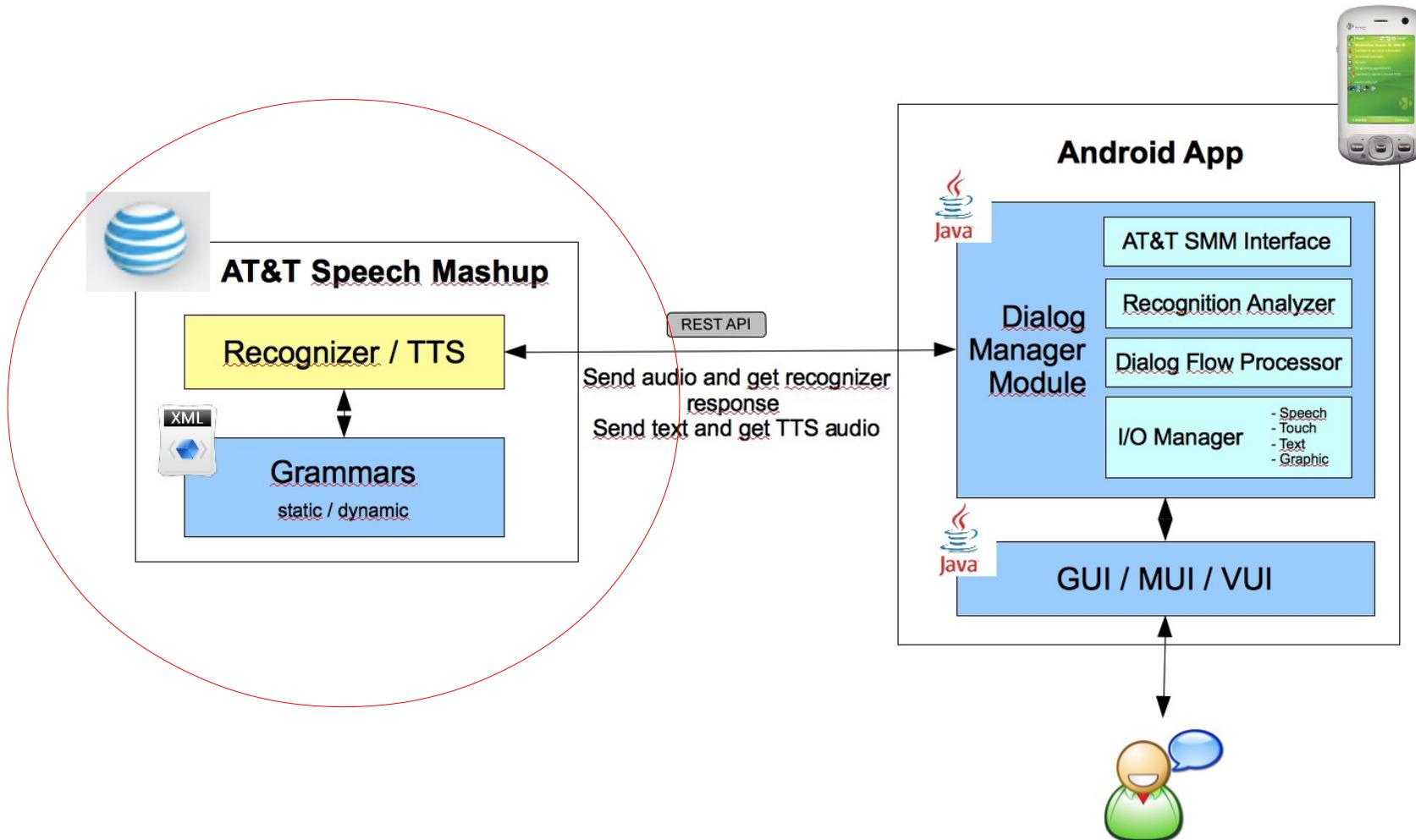


# A possible architecture



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# A possible architecture



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# AT&T Speech Mashup

## What's AT&T Speech Mashup

An AT&T speech mashup portal is a web service that implements speech technologies, including both automatic speech recognition (ASR) and text to speech (TTS) for web application

Speech mashup can be created for almost any mobile device, including the iPhone, as well as web browsers running on a PC or Mac, or any other network-enabled device with audio input

Using it, then, we can create complex speech applications using all the AT&T developing instruments.

# AT&T Speech Mashup

## What's AT&T Speech Mashup – Watson ASR

One of the fundamental component of the Mashup is the Watson ASR.

The Watson ASR is the automatic speech recognition component of the WATSON system responsible for converting spoken language to text.

Recognition main steps are:

- Identify the speech features
- Map features to basic language sounds contained in the acoustic model
- Match sounds to phrases and sentences in the grammar

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# AT&T Speech Mashup

## What's AT&T Speech Mashup – Grammars

ASR refers to user defined grammars to match sounds.

Actually the admitted grammar formats are the XML standard (W3C) usually called GRXML and the deprecated proprietary Watson BNF (WBNF)

As we are going to see it's possible to upload grammars or use the shared and builtin versions provided by the portal

# AT&T Speech Mashup

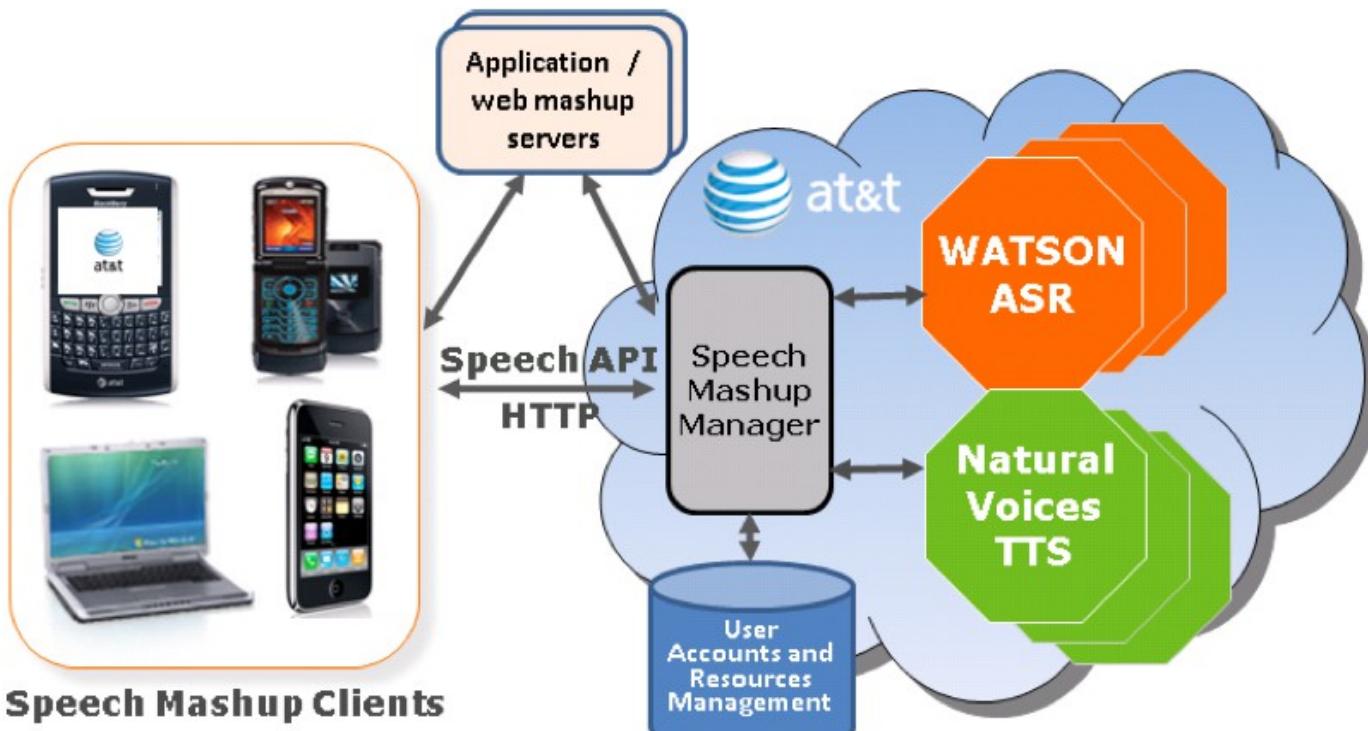
## What's AT&T Speech Mashup – TTS

The TTS, called Natural Voices, has built-in rules for normalizing text (such as converting common abbreviations to words) and assigning prosody to make the generated speech sounds as natural as possible.

In addition, Natural Voices (the TTS System) properly interpret Synthesized Speech Markup Language (SSML) tags embedded in the text to more closely control normalization, pronunciation and prosody

# AT&T Speech Mashup

## AT&T Speech Mashup Architecture



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# AT&T Speech Mashup

## AT&T Speech Mashup Web Portal

AT&T Speech Mashup provide a web portal to test and manage applications you create using the API

To use it and the API just register at the link:

<https://service.research.att.com/smm/>

You'll get the access to the platform and a unique UUID to send as a parameter when using the webservice

# AT&T Speech Mashup

## AT&T Speech Mashup Web Portal

Home | You are Logged in as karma | Logout

at&t Speech Mashup Portal

### Dashboard

Currently Selected Application: HelloWorld

 Manage Applications	 Manage Grammar Files	 ASR Test	 TTS Test	 View Logs	 Manage Transcriptions
 User Guide (html)	 Sample Code	 Message Board	 Bug Tracker	 Edit Home Page	 Edit Account Info

[Downloads](#)  
[View personal home page](#)

# AT&T Speech Mashup

## AT&T Speech Mashup Web Portal

Sections:

- **Manage Application:** in this page you can create different applications containing different grammars and dictionaries
- **Manage Grammar Files:** here you can upload, compile and view grammars
- **ASR Test:** In this section is possible to test the grammars by instantly recording an audio file
- **TTS Test:** in this page is possible to test the TTS by writing some text to be read
- **View Logs:** page containing all the logs of the applications
- **Manage Transcription:** this link open the interface for transcribing the recorded and uploaded audio files, so that it's possible to evaluate the recognition results
- **User Guide:** link to download the official guide

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# AT&T Speech Mashup

## AT&T Speech Mashup Web Portal

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- **Sample Code:** link to download the zipped file containing the clients examples
- **Message Board:** Link to google groups to ask about the AT&T Speech Mashup
- **Bug tracker:** Link to Bugzilla to report application bugs
- **Edit Home Page:** in this form you can write the HTML for your personal home page. The link to your personal homepage is below the two images rows
- **Edit Account Info:** in this page it's possible to change password, email and other fields associated to your profile

# AT&T Speech Mashup

## Web Portal functionalities

The portal, then, provide the following useful functionalities:

- Create and edit applications
- Upload, delete, rename, edit and view grammars
- Compile uploaded grammars even using special options, like SpeedVsAccuracy, vadSensitivity and nbest or changing the acoustic model and the associated dictionary
- Share grammars with all the other users. In future versions will be possible also to select users you want to share the grammars with
- Upload, delete, rename and edit dictionaries
- Instantly test the ASR selecting which grammar to use for the recognition process
- Get the ASR results in different formats: JSON (flat or nested slots), Watson JSON (indented or not), XML and EMMA
- Test TTS voices, even selecting the voice, using SSML Markup, getting notification on bookmarks, phonemes, viseme or word and getting the results in two possible formats: simple or ogg

# AT&T Speech Mashup

## Web Portal functionalities

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- Creating your own voice by uploading audio or using their interface for registering it. This part of the portal is not in documentation yet
- Check logs of all the applications
- Create transcriptions, selecting audio files to transcript by filtering per date
- Evaluate results with external tools after downloading transcription files

In addition the portal permits to set two URLs to be invoked before the ASR and after it. Through these options it's possible to modify the input parameters (like the audio got from the user speech) using an external webservice and send the elaborated data as input for the ASR and to elaborate the results before sending it back to the client, so that you can send different types of data, or use other statistics to decide which of the nbest it's better to use.

This method permits to upgrade the performances of the system, without modifying the client software.

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# AT&T Speech Mashup

## Into details: XML Grammars

This grammar matches only the words "internet", "call" and "map".

```
<grammar version="1.0" tag-format="semantics/1.0" xml:lang="en-US" root="word">  
  <rule id="word">  
    <item repeat="1">  
      <one-of>  
        <item>internet</item>  
        <item>call</item>  
        <item>map</item>  
      </one-of>  
    </item>  
  </rule>  
</grammar>
```

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# AT&T Speech Mashup

## Into details: XML Grammars

<one-of> tag create a list in which one of the contained <item> is possible

Repeat attribute set how many times the item should be repeated. If there isn't this attribute with a "0-1" value, the item must be said from the user

The special rule GARBAGE (<ruleref uri="GARBAGE"/>) define everything.

The weight attribute in the item tags define the weight to be associated to the word in the generated finite state machine. It must be between 0.0 and 1.0

If using the tag-format semantic in the definition of the grammar (<grammar tag-format="semantics/1.0" root="object">) then, it's possible to add a <tag> element to override the returned value of a grammar component using a script. Example:

```
<rule id="object">
  <one-of>
    <item>home    <tag> out="newloan"  </tag> </item>
    <item>refinancing <tag> out="refi"    </tag> </item>
    <item>refinance  <tag> out="refi"    </tag> </item>
    <item>loan      <tag> out="newloan"  </tag> </item>
    <item>interest  <tag> out="rates"   </tag> </item>
    <item>rate      <tag> out="rates"   </tag> </item>
    <item>rates     <tag> out="rates"   </tag> </item>
  </one-of>
</rule>
```

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# An example of a grammar (1/2)

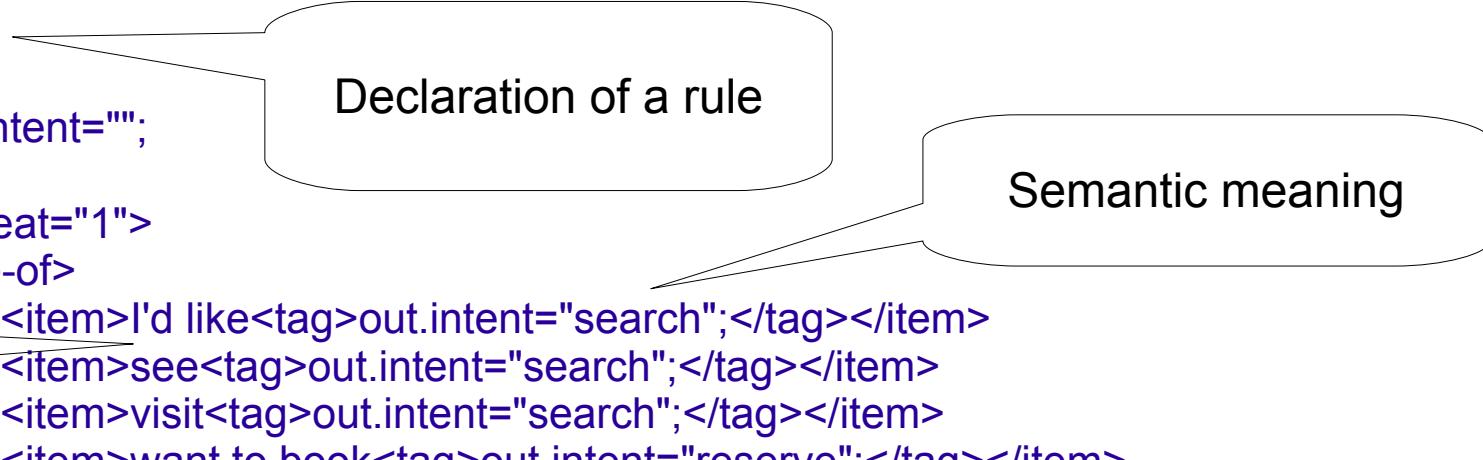
```
<grammar version="1.0" tag-format="semantics/1.0" xml:lang="en-US" root="main">
  <rule id="main">
    <item weight="0.1" repeat="0-1"><ruleref special="GARBAGE"/></item>
    <ruleref uri="#first"/>
    <ruleref uri="#preintent"/>
    <ruleref uri="#intent"/>
    <ruleref uri="#verb"/>
    <item weight="0.1" repeat="0-1"><ruleref special="GARBAGE"/></item>
    <ruleref uri="#article"/>
    <ruleref uri="#filters"/>
    <ruleref uri="#business"/>
    <ruleref uri="#place"/>
    <ruleref uri="#regards"/>
    <item weight="0.1" repeat="0-1"><ruleref special="GARBAGE"/></item>
    <tag>
      out.intent=rules.intent.intent;
      out.category=rules.business.category;
      out.place=rules.place.place;
      out.filterby=rules.filters.filterby;
    </tag>
  </rule>
```

Rules

Semantic returns  
of the grammars

# An example of a grammar (2/2)

```
<rule id="intent">
  <tag>
    out.intent="";
  </tag>
  <item repeat="1">
    <one-of>
      <item>I'd like<tag>out.intent="search";</tag></item>
      <item>see<tag>out.intent="search";</tag></item>
      <item>visit<tag>out.intent="search";</tag></item>
      <item>want to book<tag>out.intent="reserve";</tag></item>
      <item>book<tag>out.intent="reserve";</tag></item>
      <item>want to reserve<tag>out.intent="reserve";</tag></item>
      <item>reserve<tag>out.intent="reserve";</tag></item>
      <item>want to reserve<tag>out.intent="reserve";</tag></item>
      <item>want<tag>out.intent="search";</tag></item>
      <item>I'm looking for<tag>out.intent="search";</tag></item>
      <item>take me to<tag>out.intent="navigate";</tag></item>
      <item>take us to<tag>out.intent="navigate";</tag></item>
      <item>find<tag>out.intent="search";</tag></item>
      <item>where<tag>out.intent="navigate";</tag></item>
      <item>need<tag>out.intent="search";</tag></item>
      <item>go to<tag>out.intent="navigate";</tag></item>
      <item>get me to<tag>out.intent="navigate";</tag></item>
      <item>get us to<tag>out.intent="navigate";</tag></item>
    [</one-of>]
```



# AT&T Speech Mashup

## Into details: SSML Markup

SSML is a standardized XML markup language for modifying the way text is processed by TTS engines. Some of this markups are:

- Length of a pause: Begin `<Break time="3s"/>` now
- Say as: `<say-as interpret-as="value"> text </say-as>` (value can be acronym, address, currency, date, ignore-case, lines, literal, math, measurement, number, spell, telephone, time). Example: `<say-as interpret-as="date" format="dmy"> 1/2/2008 </say-as>`
- Speak tag, to define language to use: `<speak xml:lang="en_us">`
- Voice tag, to set the voice to use, or the language: `<voice name="mike"> text </voice>`
- Paragraph tag, to define a paragraph or change language: `<p xml:lang="en_us">`
- Sentence tag (or `<s>`, to set a sentence or change the language: `<s xml:lang="es_us">`
- Mark tag, to send back a bookmark to the client: `<mark name="bm1"/>`
- Break tag, to set a pause: `<break strength="level"/>` (level can be none, x-weak, weak, medium, strong, or x-strong)

# AT&T Speech Mashup

## Into details: SSML Markup

...

- Prosody volume level: <prosody volume="level | n | n%">text</prosody> (level can be silent, x-soft, soft, medium, loud, x-loud, default; number between 1 and 100)
- Prosody rate value: <prosody rate="value"> text</prosody> (value can be x-fast, fast, medium, slow, x-slow, or default or a percentage in this form: 2 = 200%, 0.5 = 50% of the default rate)
- Prosody Emphasis, to define the emphasis to use: <emphasis level="level">text</emphasis> (level can be strong, moderate, none, or reduced)
- Phoneme tag, to specify a particular pronunciation: <phoneme alphabet="ipa" ph="f aa 1 dh er 0"/>

More informations at [https://service.research.att.com/smm/download/SpeechMashupGuide.html.zip#\\_Toc295729831](https://service.research.att.com/smm/download/SpeechMashupGuide.html.zip#_Toc295729831)  
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# AT&T Speech Mashup

## Into details: Grammar Tools

Manage Grammars and Dictionaries

Applications ASR ASR Test TTS Test Voice Builder Logs Transcriptions

Currently Selected Application: HelloWorld

Grammars

Filename	Comp	Cmds	Shared
helloWorld	yes	no	no
helloWorld_tag	yes	no	no

Dictionary

Name

Upload Delete Rename Edit View: private

Compile Watson Cmds Share Grm Only Share Grm & Src Unshare

Grammar Details / [Compilation Options](#) / [Compilation Log](#)

Title: HelloWorld  
Description: Simple Grammar, to answer to question "What are you looking for?". Accept answer: pizza, pizzeria, res  
Size: source: 476 compiled: 20991  
Last modified: 2012-10-12 17:31:02  
Last compiled: 2012-10-12 17:31:13

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# AT&T Speech Mashup

## Into details: Grammar Tools

The grammars tools are separated in 4 main sections:

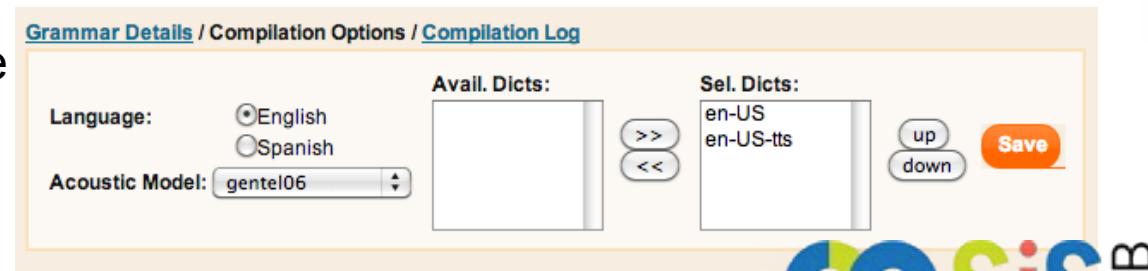
- The list of the grammars
- The list of the dictionaries
- The actions available for the selected grammar
- The box with details, compilation options and logs

To create a grammar you need to create it locally on your PC and the upload it on the portal.

After you upload the grammar you need to compile it. Then you can try to change the options for compiling. In the Compilation Options section you can select language (english or spanish), acoustic model and dictionaries to use.

There is no documentation about the available acoustic models.

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# AT&T Speech Mashup

## Into details: Grammar Tools

In the logs section, finally, you can check the logs when compiling the grammar.

When compiling a grammar it's possible to set even more parameters by pressing on the button "Watson Cmds". These are the parameters that WATSON admits that must be in the form "*set name=value*". Possible parameters are:

- config.speedVsAccuracy
  - Value range: 0.0 – 1.0 (default is .5)
- config.vadSensitivity
  - Value range: 1-100 (default is 50)
- config.nbest
  - Value range: n (default is 1)



Modifying these parameters you can set if the grammar should be more accurate or faster (speedVsAccuracy) or to be more or less sensitive when determining that audio is speech (vadSensitivity). With the nbest parameter, finally, you can set how many results you want the grammar to give you back.

# AT&T Speech Mashup

## Into details: Grammar Tools

You can even set the endpointing parameters, to make application decide automatically when the user start and stop speaking:

*activateEvh "timeouts"*

*activateEvh "speechstart-hmm"*

*timeouts.firstTimeout = 400*

*timeouts.secondTimeout = 500*

After the first timeout millisecond of silence WATSON will give you back a result only if it has a confidence score higher than the recognition threshold.

After the second timeout, instead, it will give you back the result in any case.

It's possible to set the Cmds even through the API by adding parameters in a string like this:

*...&control=activateEvh+%"22timeouts%22%3BactivateEvh+%"22speechstart-hmm*

*%"22%3Btimeouts.firstTimeout+=+400%3Btimeouts.secondTimeout+=+500*

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# AT&T Speech Mashup

## Into details: Grammar Tools

Unfortunately the documentation does not specify how to set the recognition threshold even if it say that you may want to set it in the application.

The GRXML grammar, in addition, in the W3C standard explicitly says:

*“Speech recognizer configuration: The grammar format does not incorporate features for setting recognizer features such as timeouts, recognition thresholds, search sizes or N-best result counts.”*

Then, at the moment, is not known where to set the recognition threshold for the application.

About dictionaries the documentation only says that there are 2 different dictionaries: a general large dictionary and another dictionary for TTS that generate spell for words not contained in other dictionaries. Nothing else is said, but it seems that you can add dictionaries and include them during the grammar compilation, to obtain the desired results.

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# AT&T Speech Mashup

## Into details: ASR Tools

The Portal gives you also the opportunity to test the grammar instantly. In the ASR Test page you can choose the grammar to use, the application and the result format.

Then, thanks to a Java applet, you can press a button and start speaking. The recorded voice, will be processed using the grammar and the application, and the interface will return the result of the

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ASR.

Watson Applet Test Page

Applications ASR ASR Test TTS Test Voice Builder Logs Transcriptions

URL: <http://service.research.att.com/smm/watson>

Proxy:

Result Format: EMMA

UUID: B9AD79580CD411E2B47A434161C4589C

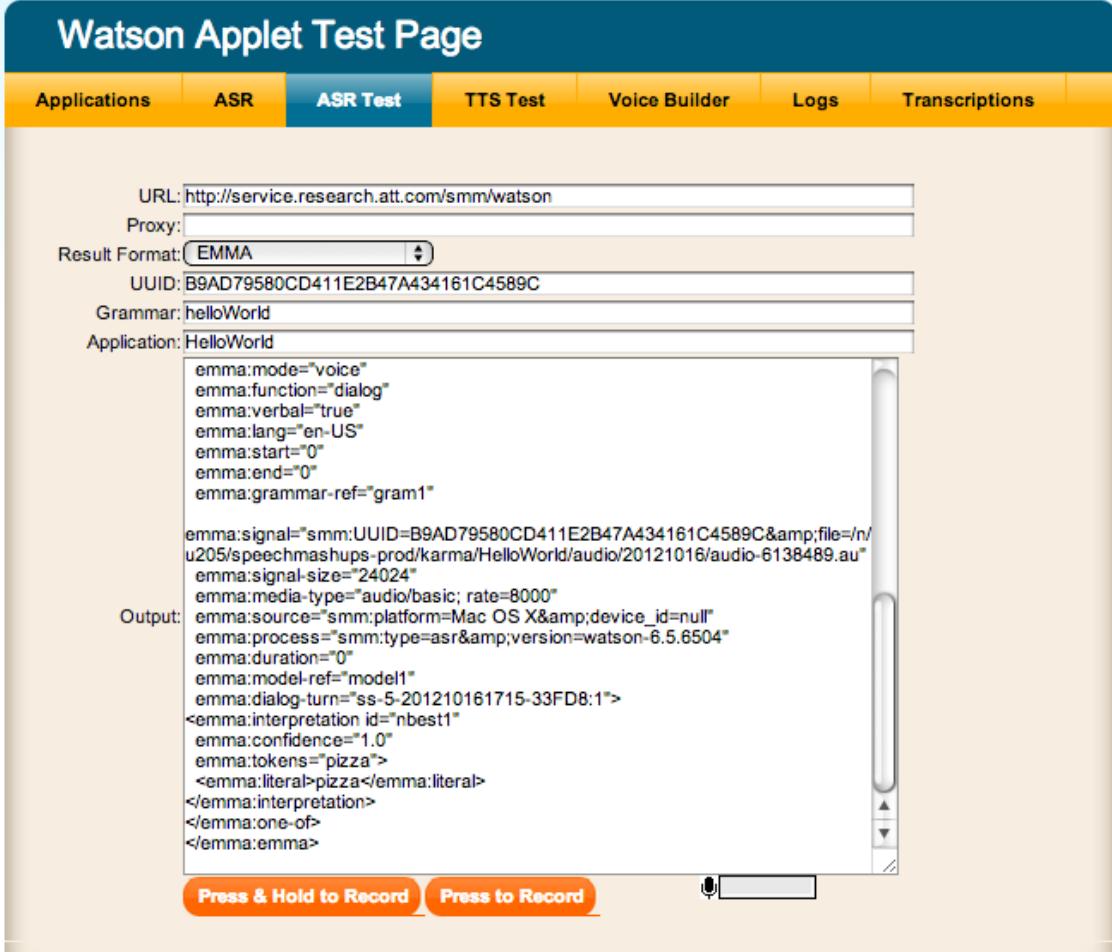
Grammar: helloWorld

Application: HelloWorld

```
emma:mode="voice"
emma:function="dialog"
emma:verbal="true"
emma:lang="en-US"
emma:start="0"
emma:end="0"
emma:grammar-ref="gram1"

emma:signal="smm:UUID=B9AD79580CD411E2B47A434161C4589C&file=/n/u205/speechmashups-prod/karma/HelloWorld/audio/20121016/audio-6138489.au"
emma:signal-size="24024"
emma:media-type="audio/basic; rate=8000"
emma:source="smm:platform=Mac OS X&device_id=null"
emma:process="smm:type=asr&version=watson-6.5.6504"
emma:duration="0"
emma:model-ref="model1"
emma:dialog-turn="ss-5-201210161715-33FD8:1"
<emma:interpretation id="nbest1"
emma:confidence="1.0"
emma:tokens="pizza">
<emma:literal>pizza</emma:literal>
</emma:interpretation>
</emma:one-of>
</emma:emma>
```

Press & Hold to Record Press to Record



# AT&T Speech Mashup

## Into details: TTS Tools

The TTS Test Portal section permits to test the TTS to decide which voice is better to use and which special tag can be added to make the voice as much real as possible.

The input fields are:

- The application to use
- The text to speech
- The voice to use (Crystal and Rosa are female voices, Mike and Alberto are male; Crystal and Mike are English while Rosa and Alberto are Spanish. The versions whose names end in 16 are 16-bit voices; the others are 8-bit)

- Use or not SSL Markup
- Notification to send
- The container type of the generated audio file

There is also an output field:

- The returned message from bookmarks

# AT&T Speech Mashup

## Into details: TTS Tools

We can try the TTS just by inserting the text and pressing the “Play Prompt” button.

Let's try something:

- Now *<prosody volume="x-loud">I can speak loudly</prosody> but <prosody volume="x-soft">I can speak even softly</prosody>*.
- *I can do even break <Break time="3s"/> while speaking or <prosody rate="x-slow">change my rate while speaking</prosody> <prosody rate="0.01">from slow</prosody><prosody rate="1.5">to the highest speed ever</prosody>*
- *I want to emphasize that <emphasis level="strong">I am really happy today</emphasis>while<emphasis level="reduced">yesterday I was really really sad </emphasis>*
- *Do I have to pronounce father in this way: <phoneme alphabet="darpa" ph="f aa 1 dh er 0"/> or in this way <phoneme alphabet="darpa" ph="p aa 1 t er 0"/>?*
- *I can say a string in different ways: as a date: <say-as interpret-as="date" format="dmy"> 10/12/2012 </say-as> or as a math calculation <say-as interpret-as="math"> 10/12/2012 </say-as>*
- *Would you like to know when I'm saying this <mark name="this\_bookmark"/> so that you can start with another process?*

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# AT&T Speech Mashup

## Into details: Logging Tools

In the logs it's possible to select the day to check the log of, and to read all the informations about the gotten processes.

In the log are available the following informations:

- Date and time of the request
- Passed parameters (like appname, cmd, grammar, resultFormat, uuid)
- WATSON response and logs (like debug, version, session\_id, hypothesis, audio file with the link to download it and so on)
- The EMMA response (or another response type if another one has been selected)
- The post processing detail and logs
- The TTS logs in case of a TTS request

# AT&T Speech Mashup

## API and Clients developing

AT&T Speech Mashup provide also API to develop applications for quite all clients.

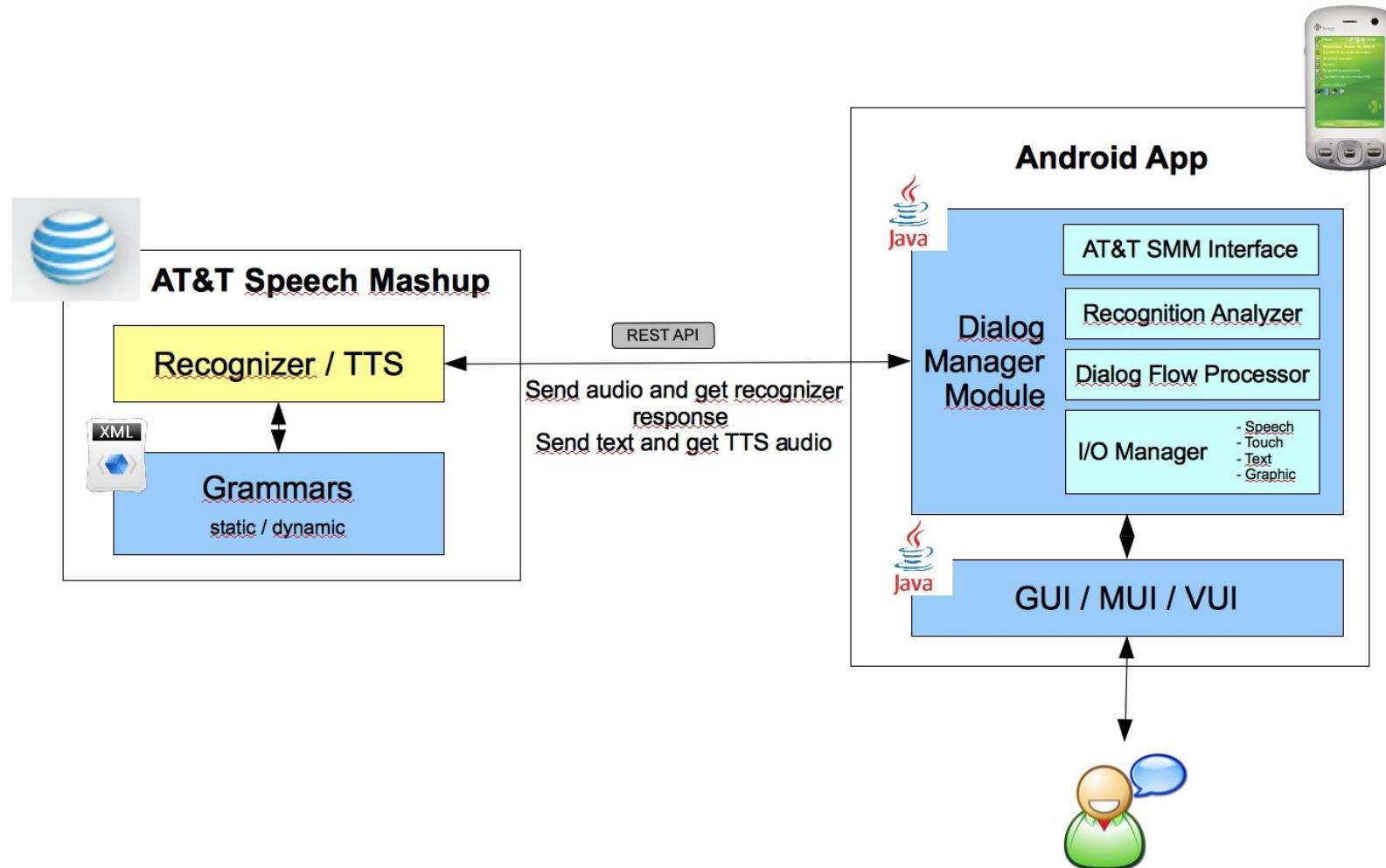
The API functionalities are the same explained before for the web portal.

Supported platforms are iPhone, Android, Blackberry and other devices supporting Java, web browsers supporting java like Safari, Firefox, Chrome and Internet Explorer.

The API can be used using the REST principles, even with the simple use of the command wget. Example:

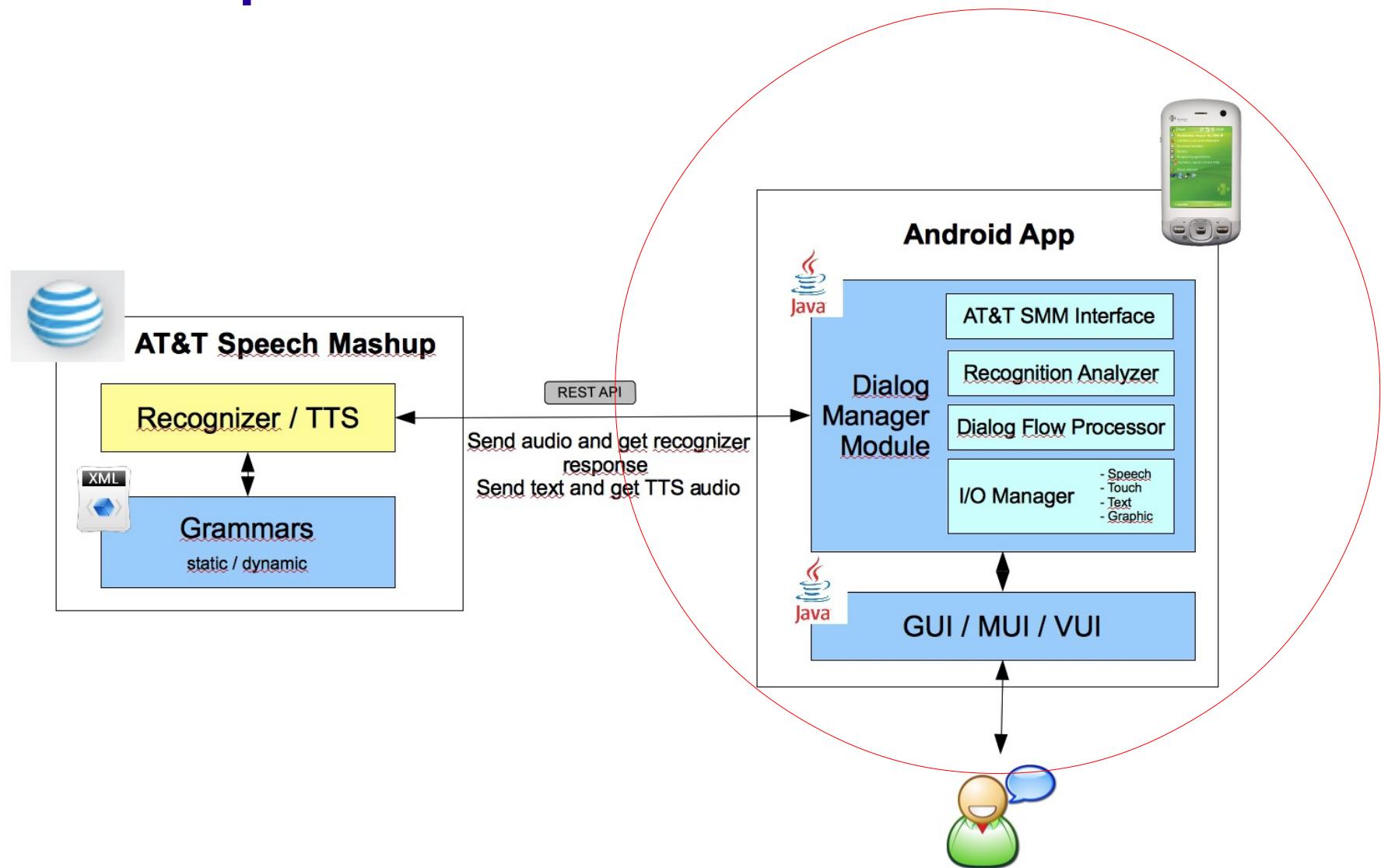
```
 wget \
  --post-file=sample-date.amr \
  --header 'Content-Type: audio/amr' \
  --server-response 'http://service.research.att.com/smm/watson?cmd=rawoneshot&grammar=en-us-date&uuid=<your own
  UPnP & appname>&application ID>&resultFormat=emma' \
  -O response.emma
```

# A possible architecture



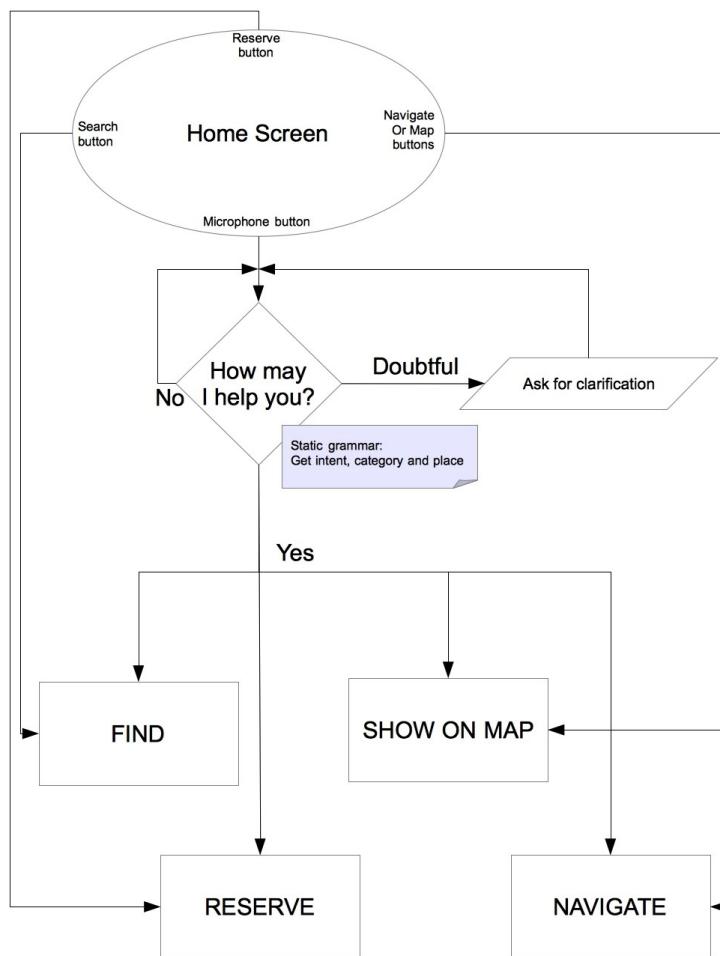
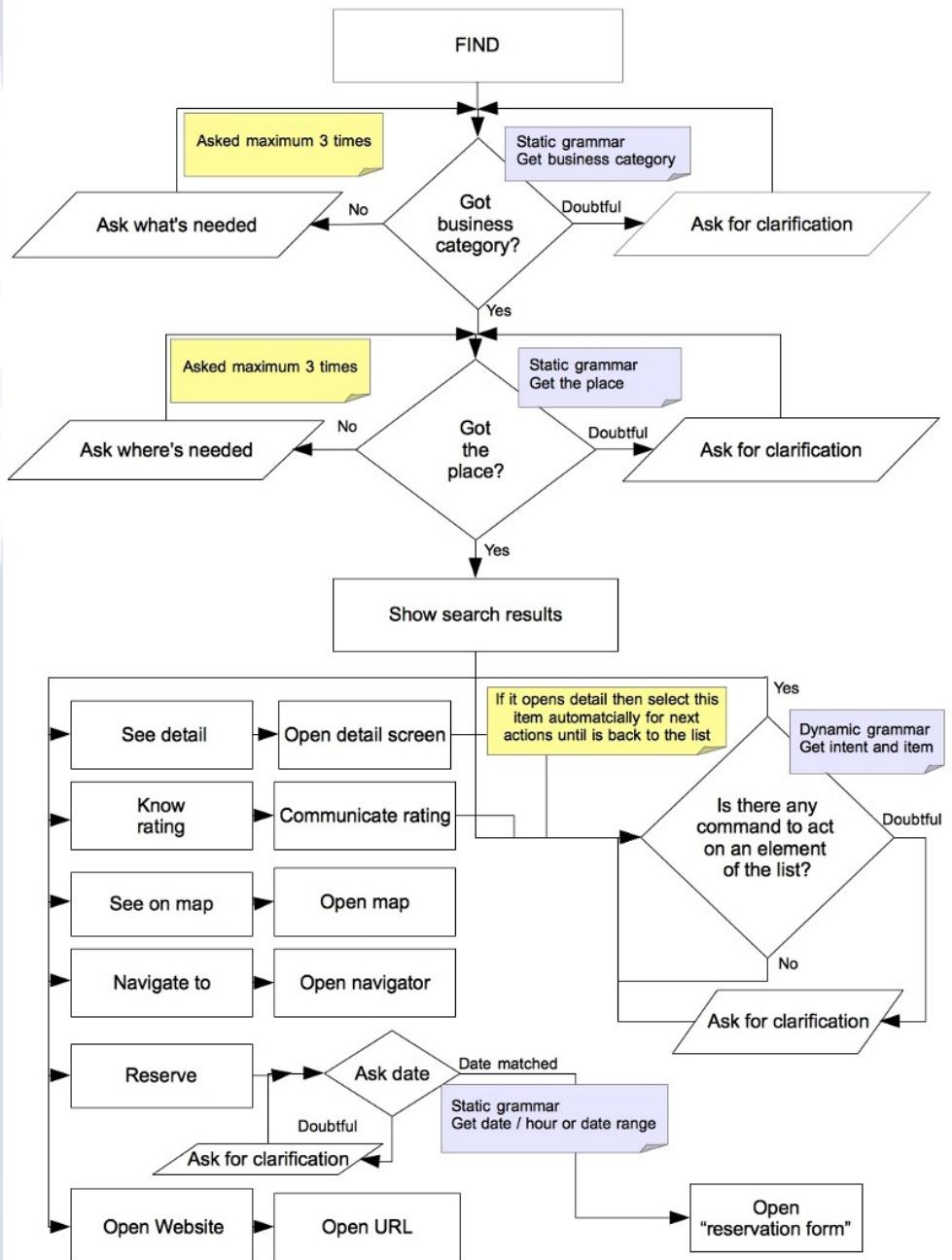
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# A possible architecture

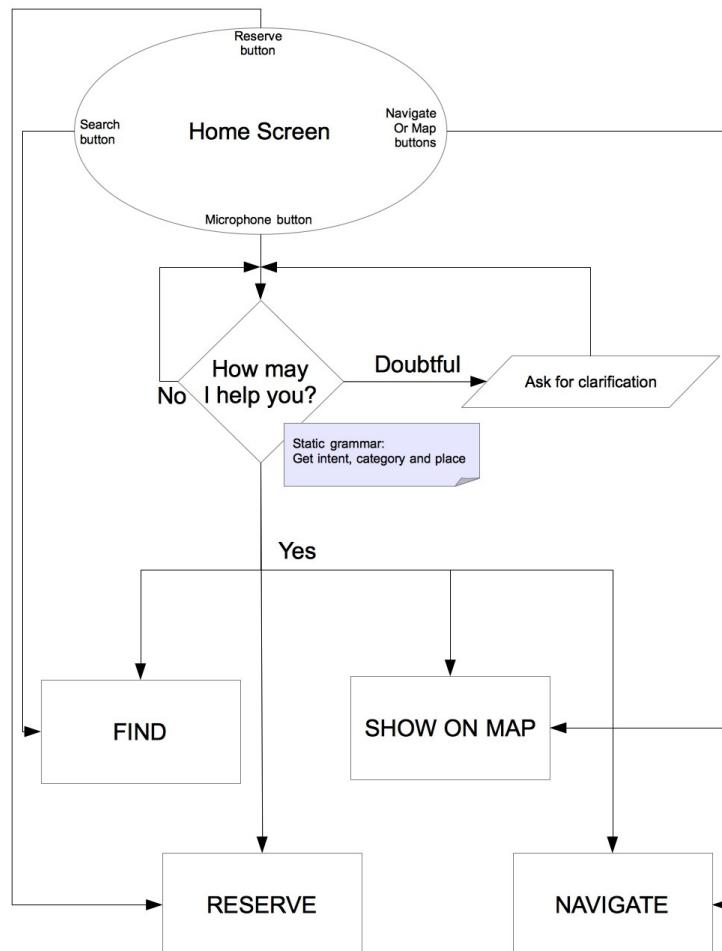
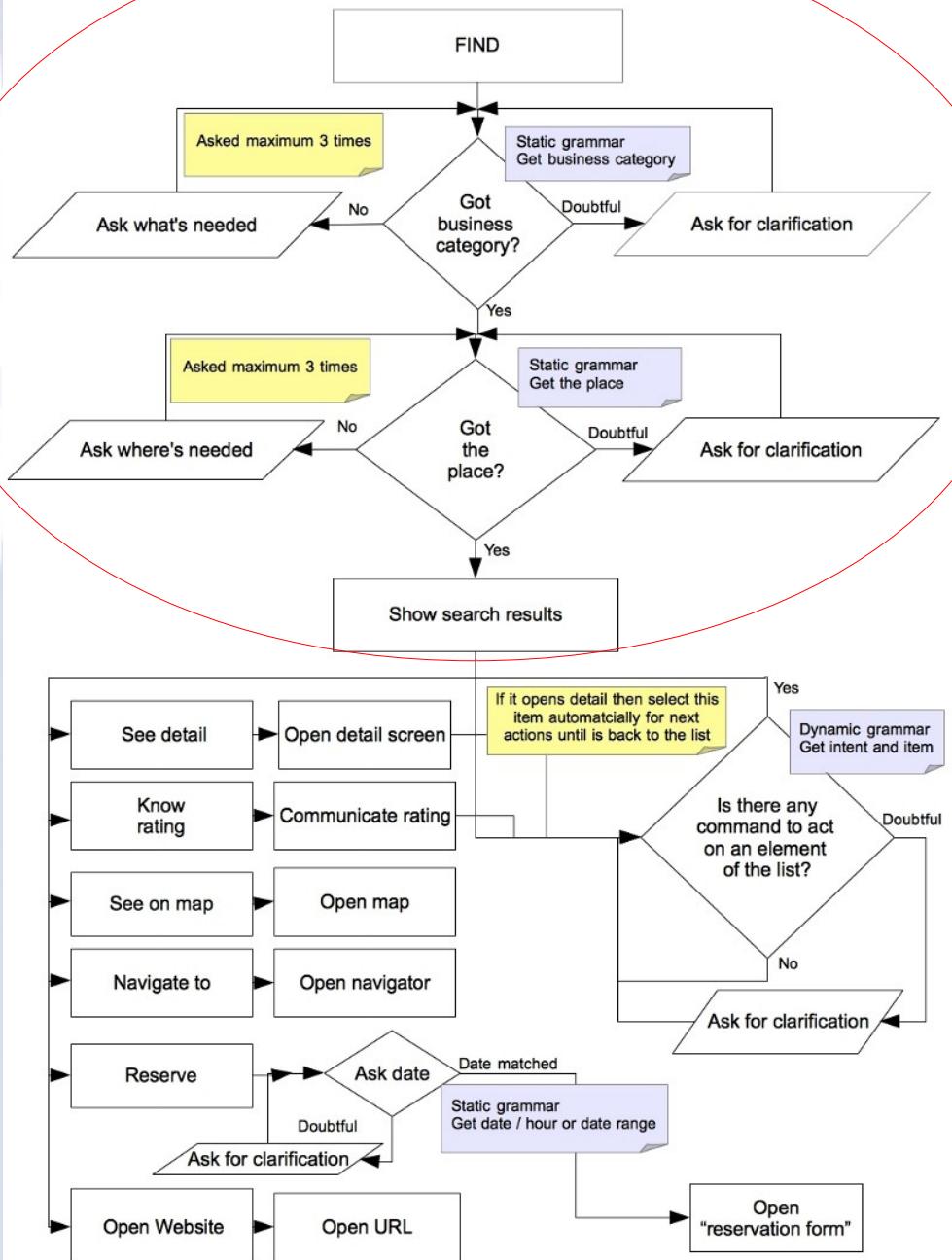


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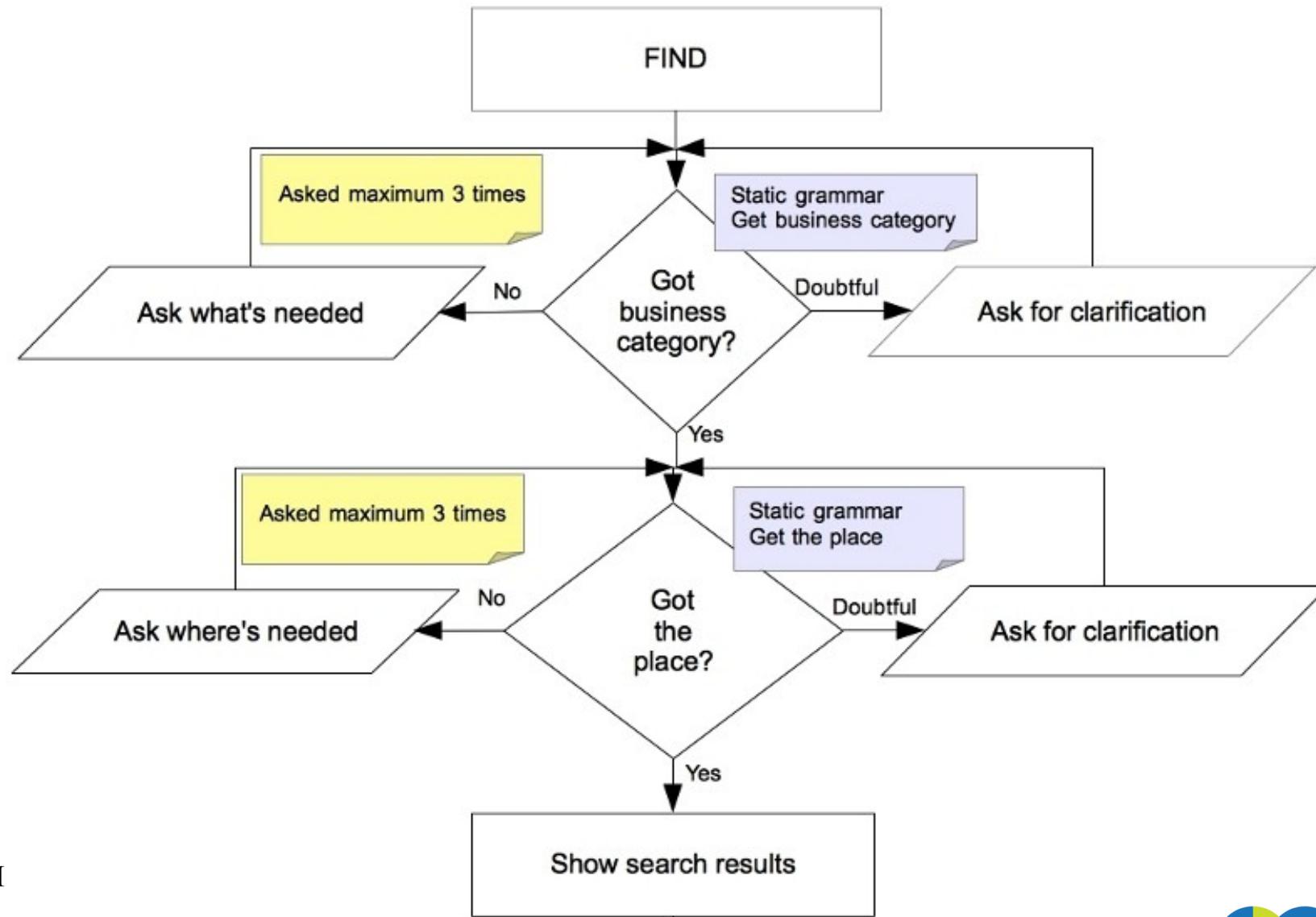
# Dialogue flow example



# Dialogue flow example

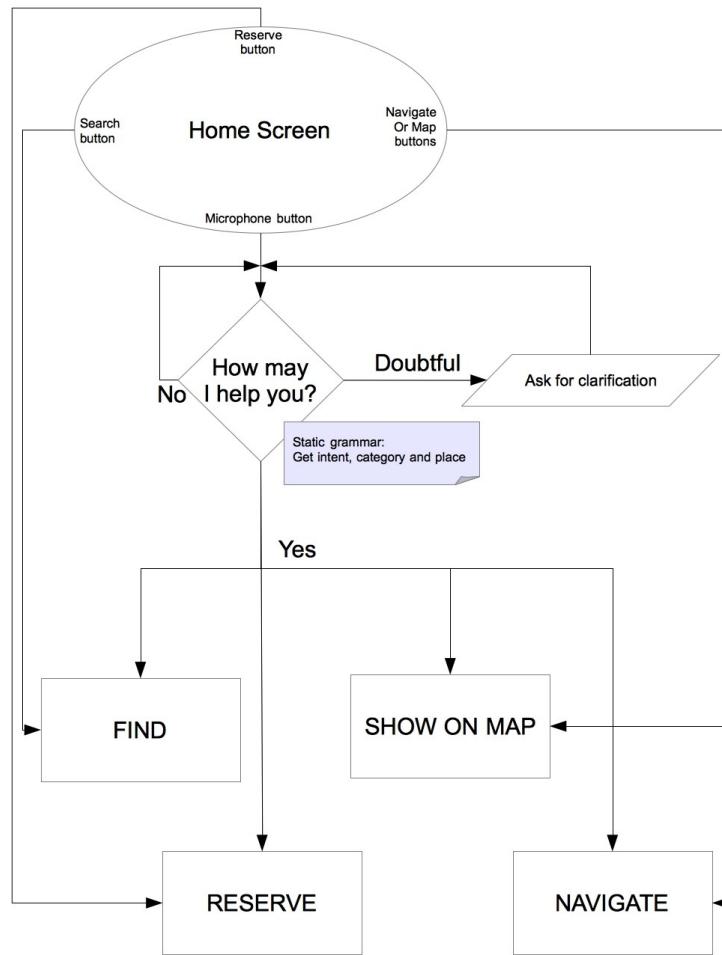
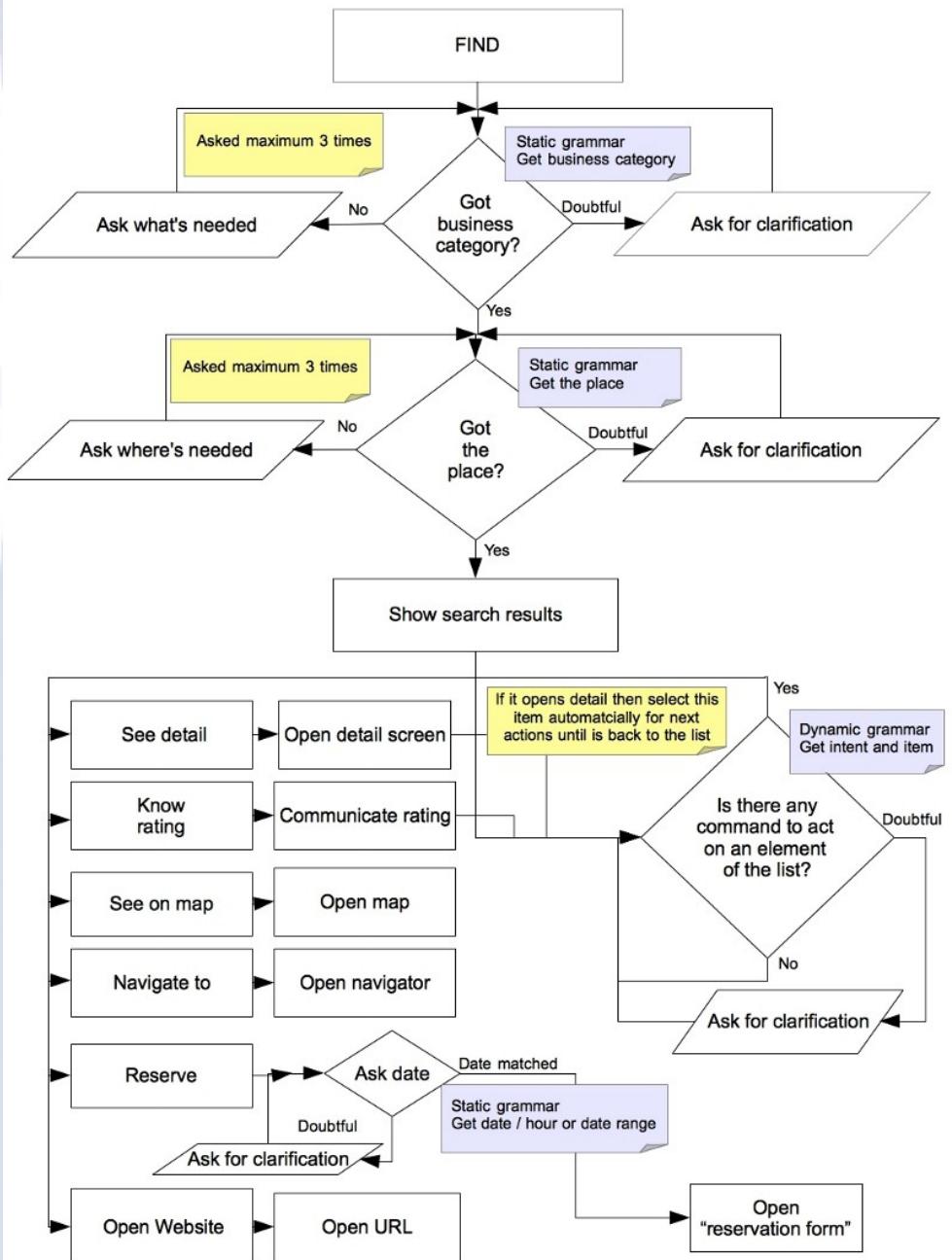


# Dialogue flow example

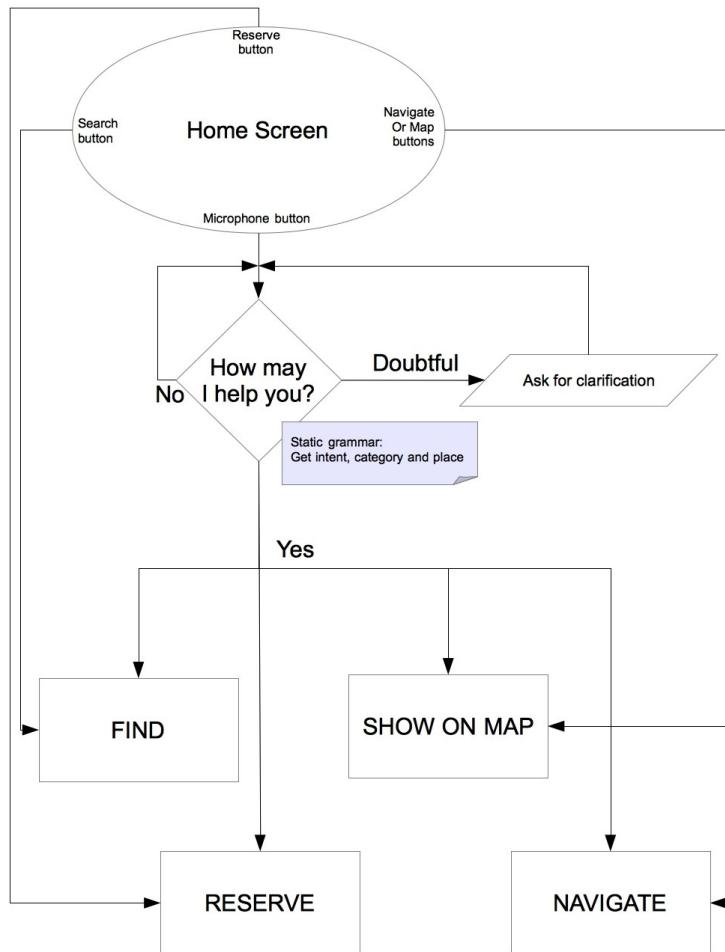
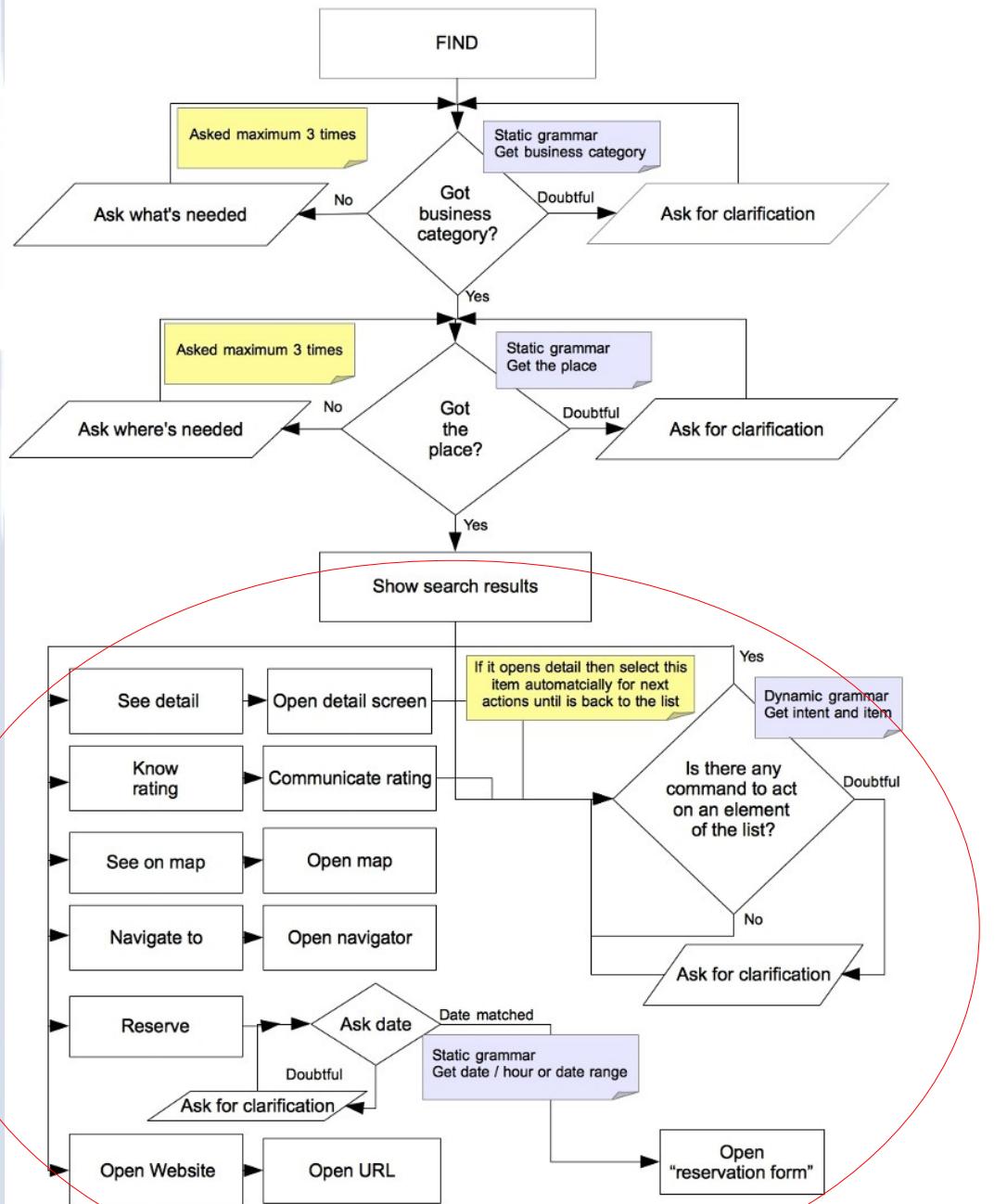


LPSM

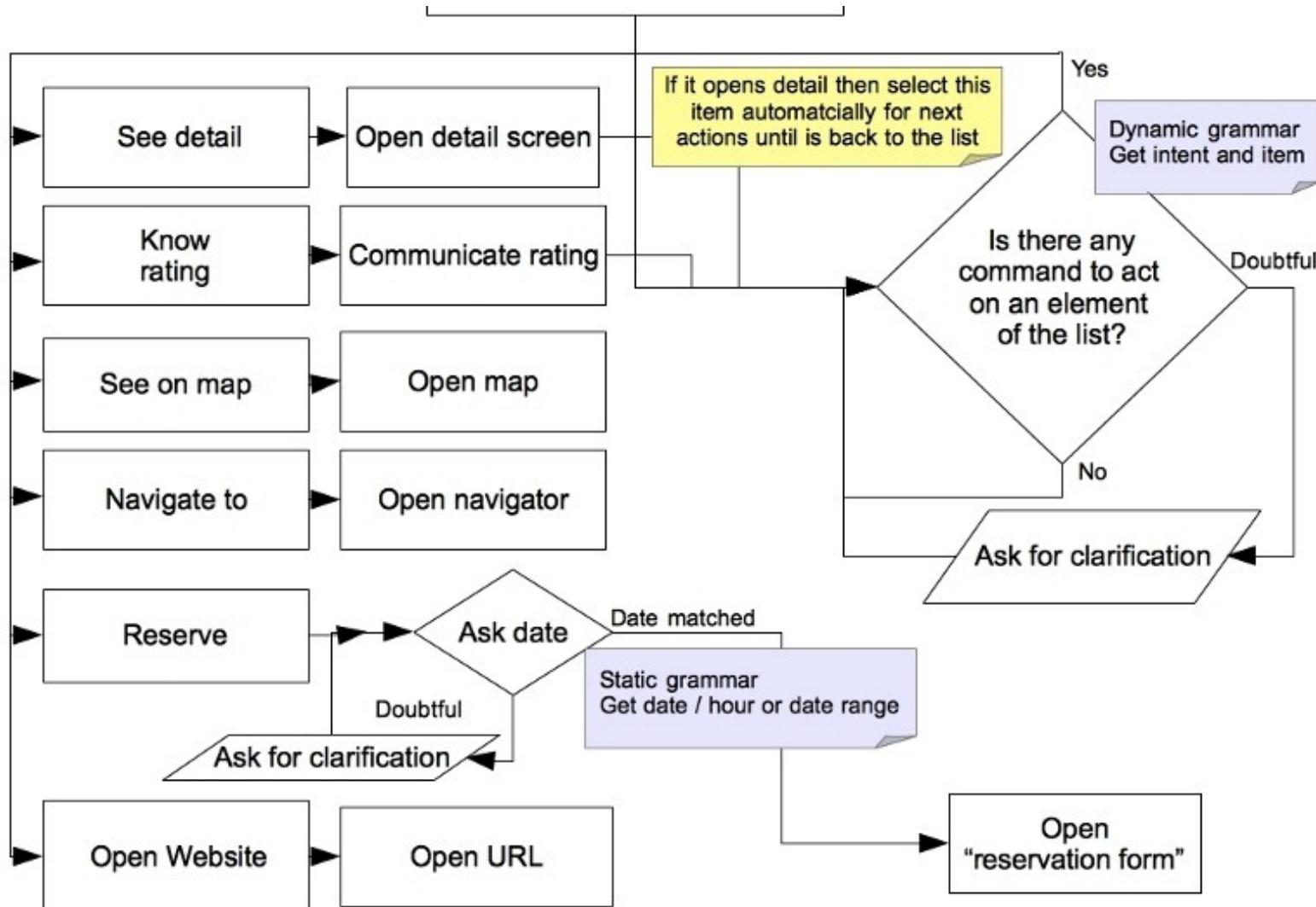
# Dialogue flow example



# Dialogue flow example



# Dialogue flow example



LPSM

# “Hello Lab” tutorial – Web Portal

- Subscribe to the AT&T Web Portal
- Create the HelloLab Application
- Create grammar and upload it
- Compile the grammar
- Test the grammar through the ASR Test
- Edit the grammar until you get the results you need, recompiling it everytime and testing the ASR after each modification
- Start developing your application on the mobile device

# “Hello Lab” tutorial for Android - Eclipse

- Create a new Android project
- Download HTTPComponents from <http://hc.apache.org/downloads.cgi> , import the .jar files inside the project in a folder named libs. Configure the build path to get these files.
- Edit the manifest and add the uses permission for INTERNET, WRITE\_EXTERNAL\_STORAGE e RECORD\_AUDIO
- Edit the main.xml with the visualization you prefer (like a textView, a button and another textView)
- Start modifying the main Activity: add inside the object the properties you'll need in all the project to be sure they are going to be seen in all your code.
- Set in these variables the URLs for the AT&T ASR and TTS putting in the arguments even your UUID
- Add a listener to the button. To have the hold effect use the OnTouchListener and check if the event corresponds to MotionEvent.ACTION\_DOWN or MotionEvent.ACTION\_UP
- On ACTION\_DOWN create the HelloLab directory in the storage and start recording the voice

# “Hello Lab” tutorial for Android - Eclipse

- On ACTION\_UP stop recording and save the result in a temporary file
- Send the file to the AT&T WATSON ASR and get the response in a String variable
- Parse the variable to get what you need from the results and set the text of your TextView, or add an element in a ListView
- If you want to say something send the request to the AT&T TTS, save the returning file and play it
- Now you have the basis! Enrich the code and application as you prefer...