Introduction to Telephony Application Development Using Asterisk, Asterisk-Java, and SIP

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I racked my minds, no gems to be found, for an impressive opening line....
So, out of frustration I write...

Frankly, I have no idea for the opening paragraphs of this booklet. All I have is a set of concepts / ideas roaming the empty space in my head.... Well, I'll just let them all out now.

This booklet is about all of them and none of them in particular (huh?). I mean, it touches the subjects drawn above to various levels of detail. Some are mentioned briefly, and some get quite an extensive coverage (for an introduction). I use a program that I wrote as a vehicle [Res. 1], and I'll start my explanation from the objectives (the requirements of the program). It reflects my learning style that usually starts with “what? tell me”, followed by “how? show me”, and ends up in “why? theories please”.

**Without further ado:** In this article I'm going to introduce you to techniques, libraries, and tools for developing telephony services. We'll use all open source stuffs here (cool, huh?), like Asterisk, Asterisk-Java Library, Ethereal, SIPp, and NIST JAIN SIP library.

**Audience**

Well, I can think of at least 2 groups of people. First: those who've never been in touch with telephony and CTI stuffs. I was in that group. But once I got my hands on Asterisk, an open source PBX, I got excited. It was like: “Wow, kewl, I can do stuffs with this thing... quite easily!”. I hope after reading this article, they'll find telephony application development interesting, and would like to give it a try.

The second group is: those who've been in this domain for some time, have heard of the word Asterisk and SIP buzzing around them, but not quite sure where and how to start making something out of them. I hope this article can be somekind of a guidemap.

I started from zero, I think. I studied the SIP specification [Ref. 5], some relevant books, installed and played around with Asterisk and related stuffs, scrolled through lots of packets captured using Ethereal, etc. I enjoyed the process, and it worked quite well for me. At least I think now I'm sufficiently oriented for my further endeavor. I'd like to share that approach with you through this article.

**By the way: if you're looking for the table of contents, it's at the end of this booklet.**
All You Need Is...

love..., and several softwares! Here's the list of things that you need to have in your computer:

- **Asterisk**, that will serve as our backend. Among other things, it switches incoming calls to registered extensions. In this occasion, we're going to monitor the events happening in an Asterisk so that we know when a call is started and ended.

You wouldn't find detailed information about how to install, run, and administer an Asterisk box here. References such as [Ref. 2] and [Ref. 6] do that job a lot better. However, I will give you an explanation just enough to quickly setup your asterisk box and configuring it in order to make our program works.

- **Asterisk-Java**, a programming library that we'll use to interact with Asterisk from the program that we're going to develop with Java. The kind of interactions possible are:
  a) Listening to events sent by Asterisk. Asterisk send all *events* happening in it to all the registered listeners. It's a socket based communication. Our program will open a socket, through which it registers itself to Asterisk (via a manager interface) so that *events* generated by Asterisk will arrive on that socket.
  b) Sending *actions* to Asterisk through the connection between our program and the manager interface. We'll receive a *response* for an action that we send to Asterisk.
  c) Sending *commands* to Asterisk through the so called "agi channel". We'll get a reply for a command that we send to Asterisk.

*Sidebar*. The distinction between *command* and *action* is not yet clear for me. I guess action is more general, in the sense that we can send an action to Asterisk any time. On the other hand, we send command through an *agi channel*, that's only available in the context of an execution of an *AGI script*. Execution of an AGI script, in turn, is triggered by a call made through the Asterisk box. So, I guess, the scope of a command is limited to a particular call.

- **SIPp**. It is a program that we can use to generate SIP signaling traffic. Typically it is used to stress-test a SIP-based telecommunication system. In our case, we need to make sure that our program performs correctly in a more-or-less realistic situation (calls coming in and out simultaneously). Therefore, we use SIPp to simulate multiple calls.

- **Ethereal**, an open source GUI network protocol analyzer. It lets you interactively browse packet data from a live network or from a previously saved capture file. As I told you earlier, I walked my way from the bottom up in understanding these stuffs. Hmm..., well, not exactly. Of course, first I read some references about how things work (for example: communication between SIP endpoints), then I moved on to API documentations to get some practical insights. But maybe because I'm a skeptic, I felt the need to really see what's being exchanged through the wire, the sequence, etc. For that reason, I opened my Ethereal, start sniffing, and try to confirm what I've read.

- In our program, particularly whenever creation of conference room is detected, we need to sneak in a dummy participant into that room for a reason that will be explained later. That dummy participant
basically is a software agent that dials in to the conference, and stays there until everybody else leave
the room. I've decided to implement the establishment (and tear-down) of the call session between that
software agent and Asterisk using SIP.

**Sidenote.** Actually we have at least 2 choices of call signalling protocol with an Asterisk box: IAX2
and SIP. I choose SIP for two reasons:

- At the time of writing there was no IAX programming library in Java, at least not the open sourced
  one. In this occasion I limit the choice only to open source softwares and libraries.

- The purpose of this article is to give the readers somekind of *birdview* over the world of telephony
  application development, using emerging open standards and open source softwares. We're not going
  for the *depth* right now, but instead for the *breadth*. Following the “more is more” principle, I bring on
  SIP to the table. The talk about “which one is more suitable for particular case, SIP or IAX?” is better
  left for another article, I guess.

SIP is interesting and promising, I can tell you. If you check big telecommunication vendors' websites,
you'll find that they all have SIP related stuffs in their product line (be it software or hardware). A
scoop of knowledge on SIP might be beneficial for you to play in that market.

We need JAIN-SIP. It is a Java-standard interface to SIP signaling stack (whatever that means :). From
practical point of view, our Java program constructs, sends, receives, and reads SIP messages using
classes and interfaces specified in JAIN-SIP. There's an open source implementation of JAIN-SIP
specification from National Institute of Standard and Testing [Res. 7].

Does JAIN-SIP vaguely remind you of JDBC...? Exactly!
**Mission Briefing**

First, let's define an objective. We want to create something like in Fig. 1. Let's call it CallWatcher.

![Fig. 1. CallWatcher's main window—single party calls](image1)

The application has 3 panels:

- **Single Party Calls**
  Calls which are answered directly by Asterisk should show up in this panel (Fig. 1). For example: in our *dialplan*, we specify that calls coming in to extension 600 will be answered by an Asterisk's built-in IVR that simply echoes whatever the caller is saying. That call should show up in the *Single Party Calls* panel.

- **Two Parties Calls**
  Calls which are terminated at another *user agent* (UA) outside of Asterisk should show up in this panel (Fig. 2). For example: we have configured in our *dialplan* that incoming call to extension 101 must be switched to agent #01. That call should show up in the *Two Parties Calls* panels. In this panel, the id of the caller – that could be her telephone number – is shown in the *Caller ID* column. The dialed extension is shown in the *Called ID* column.

![Fig. 2. CallWatcher's main window – two parties calls](image2)

- **Conference Calls:**
  Conference calls will show up in this panel (Fig. 3). The number of participants in the conference is shown in the *Participants* column. For example: we have configured that those who dial the extension 5400 will be joined to conference room #400.

  A conference room with that number will be automatically created in case no such room currently exists. As a result, we should see a new entry (row) is added to the *Conference Calls* panel.

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1 Something that originates calls / where a call is terminated. A (SIP) phone is an example of user agent.
panel. Otherwise (if the room exists), the value of Participant column in the corresponding entry in the panel should be incremented by one. Similarly, whenever someone leaves a conference the the value of Participant column should be decremented by one.

![Fig. 3. CallWatcher's main window – conference calls](image)

On all the panels there is a column named State. That columns shows the state of the call. A call can be in any of this state:

- **IDLE**: initial state
- **CONNECTING**: when Asterisk is able to locate the other party – to which the call should be connected, in the case of Two Parties Call – and is in the process of connecting both parties.
- **ACTIVE**: when the call is active (involved parties are talking to each other).
- **INVALID**: when the call has been torn-down. In the case of Conference Call, it is invalid when everyone (including the dummy participant) has left the conference room. The conference room is also automatically disposed.

Additionaly, user of CallWatcher can perform an action on an active call. She does that by selecting an active call, and click the button that represents the action she wants to execute. As you can see in Fig. 1 through Fig. 3, there are two possible actions:

- **Drop**: this will terminate the selected call. In the case of conference call, everyone in the room will be kicked out first.
- **Monitor**: this will instruct Asterisk to record the selected call.

In the case of Single Party call, it's clear that we're interested in what the caller said and heard. So we'll tap in the channel between the caller and the Asterisk box.

In the case of Two Parties call, we're interested in what either the caller or the callee said and heard. There's no difference, but you have to pick one (I pick the caller).

In the case of Conference call, we're interested in what the dummy participant “heard”. Why? Because the dummy participant was there during the whole conference session. Other participants might left and rejoined, thus missed some parts of the discussion. That's not the case with the dummy participant. For that reason we'll tap in the channel between the dummy participant and the Asterisk box.

Well, that's it. Quite rough a requirement. We are going to refine our requirements as we move on with the design and – further – with the implementation.
The Design..., a Static View

The first step in designing a system, particularly a system developed in an object-oriented way, is understanding the domain. This involves identifying the concepts within the domain, how they’re related to each other, and how they work together to accomplish a set of objectives. The outcome of that process is something called domain model.

A good place to start is an existing standard, like JTAPI (Java Telephony API). I adopted several concepts that I learned from JTAPI [Res. 6]. Overview of the design is shown in Fig. 4. Please keep in mind, we’re not implementing JTAPI here (it’s not easy). The model that I present here works in our specific situation, but might not work in other / more general situations. It is something simple enough to understand, implement, and explain..., but still conceptually correct (or at least makes enough sense).

**Fig. 4. Class diagram**

Let’s start with the central concept: 📞 Call. It represents a telephone call, the information flowing between the service provider and the call participants\(^2\). The information flow between a call participant and the provider through an logical conduit that we call 📱 Channel. Each participant is represented by its address; we call it 📱 CallEndpoint in our model.

Another very important concept in our model is the 📤 Provider. It’s an abstraction of telephony service provider software. Let’s take a look at some of the things that we can do with a Provider:

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\(^2\) Definition is borrowed from one of JTAPI Whitepapers (The Java Telephony API – An Overview).
- drop(Call call);
- monitor(Call call);

Abstraction effectively means that the one who uses it doesn’t really know (nor care) how those operations are carried out. All she needs to know is the effect of the operation, which in the case of dropping a call (for example) is: all the participants will be disconnected from the call and the call will become invalid. We simply delegate that dirty job to our underlying telephony system..., Asterisk.

**Sidenote.** Please notice that in our model Provider is an interface, not a class. There's actually a concrete class – not shown in the diagram – that implements Provider. Its name is AsteriskProvider. It contains the logics of communicating with Asterisk.

Eventhough we have no other implementation of Provider, and neither do we have any plan to create another implementation of Provider (that talks with PBX other than Asterisk), still it's a good idea to keep that Provider interface. I resist the temptation (or a harrasment from a friend of mine) to convert Provider into a class and move all the logics from AsteriskProvider into Provider (and consequently purge AsteriskProvider).

The general rule that I apply is: an abstraction of an external system – such as Asterisk – should be interface-ified. It eases the testing because then we can do somekind of scenario based testings, even without the existence of that external system. This interface based programming gets along very well with test driven development (particularly using mock object approach).

For a comparison, take a look at the following diagram that I stole from JTAPI Whitepaper [Ref. 1]. The Channel in our model is more or less equivalent to Connection in JTAPI's model. The CallEndpoint in our model is a ripped off version of Address in JTAPI's model.

![Fig. 5. JTAPI's call model](image-url)
Let's go back to the telephone Call. We have 3 types of call:

- Single Party Call
- Two Parties Call
- Conference Call

**Single Party Call**

*Single Party Call* is a call with only one participant (thus one channel). It happens when someone calls an extension in Asterisk which is configured to switch the call to an Asterisk's built-in IVR. There is a flow of information only between the caller and the Asterisk. The class ✴️ *SinglePartyCall* in our model represents this concept.

When I called the extension 600 – mapped to *Echo* application – the log that I observed on Asterisk's Command Line Interface (CLI) was like the one shown in *Fig. 6*. That log confirmed me that there are indeed cases when a call has only one channel.

**Two Parties Call**

*Two Parties Call* is a call with two participants – two channels – just like a normal phone call that most of us would expect :). The class ✴️ *TwoPartiesCall* in our model stands for this concept.

In our *asterisk dialplan*, a call to certain numbers will be switched to another party outside of Asterisk. For example if someone calls the extension 101, she will be connected to someone named “Agent 1”. In that case the participants of the call are: Mrs. Jane Doe (the caller) and Agent 1. Take a look at *Fig. 7* (so that you're convinced :):

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3 Warning: they are purely product of my imagination. You will not find them in JTAPI specification.
Conference Call

Conference Call is a call that can have more than two participants (therefore more than two channels). The class ConferenceCall in our model corresponds to this concept.

Asterisk has a built-in application for conferencing. Its name is MeetMe. There’s a slight difference between Asterisk's model and ours (regarding conference). Take a look at Fig. 8.

Fig. 8. Asterisk CLI – conference call

Looks like in Asterisk's model a conference is not a really call. It's more like a collection of single party calls which are destined to an extension which is configured to switch the call to a particular conference room, where the information from the callers are mixed and then distributed to every participants.

I decided not to follow Asterisk's model. I wanted to be as consistent as possible with the model that I learned in JTAPI. None of the subtypes of javax.telephony.Call in JTAPI has a set of calls as its property. Instead, an instance of javax.telephony.Call has a set of instances of javax.telephony.Connection. Therefore, in our model ConferenceCall is modeled to have a set of instances of Channel.
The Design..., the Dynamics

Let's start with a simple question: “Who creates instances of Calls?” Let's see... a call is actually a concept within the telephony system. I mean, a call is established in the telephony system. Take a look again at Fig. 6 to Fig. 8. It's understandable if someone says “Ah, currently there are 5 calls going on in the Asterisk box” after seeing the report in the Asterisk CLI that resulted from an execution of “show channels” command.

In our model the telephony system is represented by an instance of Provider. Therefore it makes sense to assign the responsibility of instantiating Calls to Provider. Currently there's no mechanism that lets users of our program to instruct Provider to create a call (call creation is not on-demand). Instead, calls are created automatically within the Provider on notifications received from the underlying telephony system.

Instance of Provider is a stateful object – it holds instances of Call. Because of memory usage consideration, we decided to make the Provider holds only the active calls. We refer to those calls as attached calls\(^4\). To get the list of calls attached to a provider simply invoke the method \(\text{getAttachedCalls}()\) on it.

A boy cried: “but..., I also want to be notified when a Call is created, and – at the same time – obtain the reference to that instance of Call“.

So we invented an interface named \(\text{ProviderListener}\) to calm that boy. It has 2 methods:

- callAttached(Call call)
- callDetached(Call call)

Accordingly, we add methods into Provider to register and unregister a listener to it. Those methods are:

- addListener(ProviderListener listener)
- removeListener(ProviderListener listener)

In our program the class that controls the user interfaces – net.raka.agiexp.gui.Main – implements ProviderListener. Whenever a new instance of Call is attached to the provider that it listens to, a new row – that shows information about the call – will be added to the table in the appropriate panel (depending on the concrete type of the call).

Call construction is a multi-steps process. It's not like the Asterisk simply notifies our Java application once whenever a new call is created within the Asterisk, and passes along all the information needed to create an instance of Call in our Java program. The library that we're using to communicate with Asterisk – Asterisk-Java – works at the lower level. What the library passes to our Java application are notifications about the events that take place in the Asterisk, in the form of instances of \(\text{ManagerEvent}\).

\(^4\) When a call becomes invalid it will be detached from the provider to which it was previously attached.
Even something as simple as calling the extension 600 – that is mapped to the *Echo* application – will trigger a sequence of manager events as shown in Fig. 8. Loosely speaking, that sequence of events builds up the single party call.

```
class: net.sf.asterisk.manager.event.NewChannelEvent
dateReceived: Sat Feb 25 02:19:42 CST 2006
getPrivilege: call, all
callerId: customer01
callerIdName: customer01
channel: SIP/customer01-5afa
state: Ring
uniqueId: 1140855582.4

class: net.sf.asterisk.manager.event.NewExtenEvent
dateReceived: Sat Feb 25 02:19:42 CST 2006
getPrivilege: call, all
appData: null
application: Answer
channel: SIP/customer01-5afa
context: inbound stress
extension: 600
priority: 1
uniqueId: 1140855582.4

class: net.sf.asterisk.manager.event.NewStateEvent
dateReceived: Sat Feb 25 02:19:42 CST 2006
getPrivilege: call, all
callerId: customer01
callerIdName: customer01
channel: SIP/customer01-5afa
state: Up
uniqueId: 1140855582.4

class: net.sf.asterisk.manager.event.NewExtenEvent
dateReceived: Sat Feb 25 02:19:42 CST 2006
getPrivilege: call, all
appData: null
application: Echo
channel: SIP/customer01-5afa
context: inbound stress
extension: 600
priority: 2
uniqueId: 1140855582.4
```

**Fig. 8. Manager events generated when 600 is dialed**

A *NewChannelEvent* – like the first event in Fig. 8 – gives an indication to our application that someone / something is connecting to the *Asterisk*. But at that point we can not figure out if it is a single party call, two parties call, or conference call. More information is needed, that is the value of the next events in the sequence. Only when the sequence of events turn out to be like the one in Fig. 8, for example, we can conclude that a new single party call has been created.

Analysis of sequence of manager events will be discussed thoroughly in the *implementation* section.

The point here is: we need something that keeps track of the events, and instantiates a *Call* at an appropriate point in time. In our model that “something” is ✗ *CallConstruction*. You can think of an instance of *CallConstruction* as a *finite state machine*. You feed manager events to it, sequentially. If it accepts the event (for processing) then it will switch its state, depending on the type and the properties
of the event. When it arrives to a specific state – lets call it “call instantiation state” – it will instantiate the correct type Call.

So, we have a manager event being passed around. First off, a manager event is raised by Asterisk. Then it travels through a TCP socket to AsteriskProvider that acts as ManagerEventHandler that has only one method, namely handleEvent(ManagerEvent event). See arrow #1 in Fig. 9.

![Fig. 9. Call construction](image)

AsteriskProvider will first iterate through the attached calls, to see if any of them would “accept” the manager event. It does that by calling the process(ManagerEvent event) on each instance of Call (see arrow #2). The method will return true if the call accepts the event, or false otherwise.

An instance of Call also keeps track of ManagerEvents in order to be able to decide if it should update its state. Of course not all events passed in to it will be accepted by the Call. A call determines acceptability of an event based on its current state and the previous events that it has accepted.

If none of the calls accept the event then AsteriskProvider will iterate through the living CallConstructions. Again, it calls a method named process(ManagerEvent event) on each instance of CallConstruction (see arrow #3). If none of them accepts the event then AsteriskProvider will inspect the type of the event. If it is a NewChannelEvent and the state of the channel is “Ring”, then a new instance of CallConstruction will be created and added to the list of living CallConstructions.

Otherwise – if there’s a CallConstruction that accepts the event – and the event causes the CallConstruction to move on to the “call instantiation state”, a new instance of Call will be created and attached to the AsteriskProvider (see arrow #4). Then, the CallConstruction will be removed from the list of living CallConstructions.

Another concept needs explaining is the java interface CallListener. A call will notify its listeners of interesting events that happen in it. CallListener has the following method, among others:

- stateChanged(int oldState, Call call)

The class that controls the user interfaces in our program – net.raka.agiexp.gui.Main – also implements CallListener, so that it can update the display on the table of calls (each time any of the calls that it keeps track of changes its state).

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5 An interface defined in the Asterisk-Java library that handles events received from an Asterisk server.
**Setting Up the Environment**

Let's set up our playground before jumping in to the fun part that is coding...., starting with the installation of Asterisk. Oh, did I tell you that we need to work on Linux OS? I'm using OpenSuse 10 at home, but Asterisk can run in (m)any flavor of Linux.

**Asterisk**

The installation is easy. Once you have it downloaded from [http://www.asterisk.org](http://www.asterisk.org) just do the following steps:

1. Unpack that file (i.e.: `tar xzvf asterisk-1.2.4.tar.gz`)
2. Go to the directory where you unpacked the file
3. Type in the command line: `make`
4. Then type: `make install`

Now we can move on to the configuration stage. Asterisk will look for its configuration files under the directory `/etc/asterisk/`. There are more than a dozen of configuration files (mucho), each of them controls particular aspect of an Asterisk server. If you're lazy (like me), you wouldn't want to write those files from the scratch. Rather, you'd simply type `make sample` in the command line. That will copy sample configuration files to `/etc/asterisk`.

Among those configuration files, we will be concerned with only four of them:

- `sip.conf`
- `extensions.conf`
- `manager.conf`
- `meetme.conf`

And let's start with `sip.conf`...

**sip.conf**

Among other things, here we define the locally connected SIP phones, which can be of any of these types: `user`, `peer`, and `friend`.

- For a UA that can only place calls to Asterisk we set the type to `user`. The UA will be asked authenticate itself everytime it places a call. The authentication user name and the password sent by the UA (in response to the challenge) will be matched against the name of the entry and the password specified for that entry, respectively.

- For the UA that Asterisk can place calls to – in addition to receiving calls from – we set the type to `peer`. That UA is kind of trusted by Asterisk, meaning that it will not be asked for authentication (password) when placing a call.

However, for the UA to be able to receive calls, Asterisk needs to know its location. Registration – by the UA – is a way of notifying Asterisk about its location. Only then Asterisk will ask for authentication user name and password.
The last type, *friend*, is *user* + *peer* in one.

Most of the time I set the locally connected phones to *friend*, though. However, to let people in the office to place or receive a call to outside parties through another SIP server (e.g.: *FreeWorldDialup*), we define an entry of type *peer*.

Let’s experiment a little. First, copy and paste the following lines to the end of your *sip.conf*.

```conf
[johndoe]
type=user
secret=p@55w0rD
host=dynamic
canreinvite=no

[goldenboy]
type=peer
secret=p@55w0rD
host=dynamic
canreinvite=no
```

(And...?) Let’s read about *extensions.conf* before running anything :).

**extensions.conf**

This file contains the *dialplan* of an Asterisk box. It's a text file where we – in a simplest scenario – specifies something like “if the caller dials 103 then switch the call to Jeniffer the CEO’s assistant”. The book [*Ref. 2*] succintly describes it as:

> ... the heart of any Asterisk system, as it defines how Asterisk handles inbound and outbound calls. In a nutshell, it consists of a list of instructions or steps that Asterisk will follow.

Dialplan are broken into sections called *context*, which is a named group of extensions. Some of the contexts defined in the sample *extensions.conf* are: international, longdistance, local, default, and demo. Now let’s take a look again at *sip.conf*. You should find, in the *general* section, something like in Fig. 10:

```
[general]
context=default ; Default context for incoming calls
allowguest=no ; Allow or reject guest calls (default is yes, this can also be set to 'osp')
```

**Fig. 10. A portion of sip.conf**

The highlited line says that: the calls which come through the SIP channel, from a caller / user whose context is unknown / undefined, will be put in the context named *default*. The *default* context in the sample *extension.conf* is defined the following way:

```
[default]
;
; Default context for incoming calls
;
; probably don’t want to have the demo there,
;
include => demo
```

**Fig. 11. The default context in the sample extensions.conf**

Ok, it leads us to another context named *demo* (grr...). There are several extensions defined in the demo
context, with the extension 600 being one of them (see Fig. 12).

```
; Create an extension, 600, for evaluating echo latency.
;
exen => 600.1,Playback(demo-echotest) : Let them know what's going on.
exen => 600.n,Echo : Do the echo test.
exen => 600.n,Playback(demo-echodone) : Let them know it's over.
exen => 600.n,GoToFs(6) : Start over.
```

Fig. 12. The extension 600 defined in the demo context

The user johndoe doesn’t have its context defined (in sip.conf). Therefore the call from johndoe will be placed in default context. Furthermore if the extension called by johndoe is 600, then it will be (1) greeted with the playback of an audio file named demo-echotest, then (2) handled by an application named Echo (that simply echoes back whatever the caller says), and (3) greeted again with the playback of an audio file named demo-echodone. Let's prove it, by making a call.

**Your first call**

First, start the Asterisk by typing `asterisk` in the command line, followed by typing `asterisk -vvvvvvvvv` to get into the Asterisk CLI (so that we can see what's going on). To exit from the CLI – without killing Asterisk's process – type `exit` in the CLI. To stop the Asterisk type `stop now`.

Next, let’s configure your softphone. There are at least 2 free SIP softphones that runs on Linux: KPhone (open source) and Xien-Xlite. On this occasion we’ll use Kphone. With the Kphone, first, you need to configure the identity that it will use when placing a call. Follow the steps displayed in Fig. 13. You have to set the “Host Part of SIP URL” to the IP address of your Asterisk box.

**Fig. 13. KPhone identity configuration**
Once you're done with step 2, click the OK button, then you will have to restart your Kphone (exit and rerun). To place a call, type the SIP URL of the destination – which in this case **600@90.0.0.15** – in the Kphone address bar (see Fig. 14), followed by hitting the Enter key. When you’re prompted for password type **p@55w0rD**. That's it..., are you connected now?

![KPhone](image)

**Fig. 14. KPhone**

**Your second call**

Now let's try the following scenario: “if someone (such as johndoe) calls the extension 666, he will be transferred to goldenboy...”. The first thing you'll have to do is registering the extension 666 to the **default** context. Copy the following lines...

```plaintext
exten => 666,1,Dial(SIP/goldenboy)
exten => 666,n,Hangup
```

... and paste them into **extensions.conf**, inside the definition of **default** context, such that your **extensions.conf** looks like Fig. 15.

![Definition of default context, after the modification](image)

**Fig. 15. Definition of default context, after the modification**

Now you have to run another instance of Kphone for the peer **goldenboy**. If you have more than one machine, you're lucky, you just have to repeat the identity configuration steps explained above on the other machine, specifying **goldenboy** as the authentication user name. Otherwise you'll have to do two things:

1. Create another linux user on your machine, open a new instance of **Konsole**, login as that new user from that console, and run Kphone from the console. Follow the steps exemplified in Fig. 16.

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6 Change the host part of the SIP URL to the IP address your Asterisk box.
2. Do a little trick with the Kphone because by default KPhone will try to have an exclusive access to the soundcard during a phone call. The trick basically is configuring at least one of the instances of KPhone to use null device (/dev/null). Follow the steps exemplified in Fig. 17.

In our scenario, this instance of KPhone is going to receive a call. Therefore it has to register itself to the Asterisk. Let's register the phone under the name goldenboy. Follow the steps shown in Fig. 18.
To verify if (location of) the phone has been registered in the Asterisk, go to Asterisk CLI and type `sip show peers`. From the example output of the command – see Fig. 19 – we can tell that the phone is located on the machine 90.0.0.15 and is bound to port 5062.

At last, you can make your second call now. From the first instance of Kphone (johndoe), dial 666@90.0.0.15 (change the IP address). That's it..., are you connected now? That's all about the installation and configuration of Asterisk. I hope you didn't run into any troubles. Next, let's get dirty with the codes!
Implementation

Before anything else, I suggest that you download the source code of your application from [Res. 1]. Then you import the project into your Eclipse workspace and try to resolve the library dependency issues by yourself. Here's how the project structure looks on my Eclipse:

Let's begin our journey from the constructor of the Main class. There are three things that Main does when its being instantiated, they are: 1. opening a connection to the Asterisk box, 2. registering itself to the AsteriskProvider as one of its listeners, and 3. registering AsteriskProvider as the handler of manager events received from the Asterisk box (through the connection established in step. 1). Please take a look at Fig. 21 to see the corresponding lines of code.

```java
ManagerConnectionFactory mgrConnFactory = new ManagerConnectionFactory();
ManagerConnection mgrConn = mgrConnFactory.getManagerConnection(asteriskIP, user, password);  
asteriskProvider = new AsteriskProvider(mgrConn);
asteriskProvider.addListener(this);  
mgrConn.addEventListener(asteriskProvider);  
mgrConn.login();
```

Fig. 20. Eclipse screenshot (... a teaser ;)

Fig. 21. Snippet from constructor of Main class
Now let’s see how, roughly, the display (user interface) is updated to show the information about the new call everytime a new call is established in the Asterisk box. You must have understood that when a call is established in the Asterisk box, a corresponding instance of `Call` will be attached to the `AsteriskProvider`\(^7\). Being one of the listeners of the provider, `Main` will receive a notification of that event (\(\circ\)).

![Fig. 22. Snippet from constructor of Main class](image)

The provider pass in the call to the listeners (when doing the notification). `Main`, in this case, will inspect the concrete type of the instance of `Call` that it receives in the notification. Depending on the concrete type, `Main` will add the call to any of the following lists that it maintains: `singlePartyCalls`, `twoPartiesCalls`, or `conferenceCalls` (\(\bullet\)). The lines of code in `Main` that corresponds to this step is shown in Fig. 23.

```java
public void callAttached(Call call) {
    if (call instanceof SinglePartyCall) {
        singlePartyCalls.add(call);
    } else if (call instanceof TwoPartiesCall) {
        twoPartiesCalls.add(call);
    } else if (call instanceof ConferenceCall) {
        conferenceCalls.add(call);
    }

call.addListener(this);
    presentationModel.callAttached(call);
}
```

![Fig. 23. Implementation of callAttached in Main class](image)

The `Main` class pushes the notification further to an instance of `PresentationModel` by calling the `callAttached(...)` method on that presentation model (\(\circ\)). `PresentationModel` represents the state and behavior of the presentation independently of the GUI controls used in the user interface. For example, instead of having the logics that decide whether or not the monitor button in the panel which shows single party calls should be enabled, mixed with the codes that define things like dimension and laying

\(^7\) Which is the reason why I draw a projection light from the Asterisk box to the `AsteriskProvider` – to indicate that the `AsteriskProvider` is representation of the Asterisk box in the world of our program.
out of the GUI controls, we keep them in two separate classes:

- Definitions of GUI controls go to a class named `AgiExp`.
- The logics mentioned previously go to `PresentationModel`.

The idea is that the presentation model can be used with various presentation technologies (be it Swing, web, or others). Of course, then we have to provide a binding between the presentation model and the user interface that lies on top of it, somehow. For more about presentation model, go to [Ref. 3].

Our presentation model pushes the notification even further, to its listeners (that are instances of `PresentationModel.Listener`), with `AgiExp` being one of them (1). In response, `AgiExp` will ask the appropriate table to refresh itself to reflect the latest content of the corresponding list of calls pulled out from the presentation model (2). The following figure shows the section of codes in `AgiExp` from where the request for a refresh is sent.

```java
public void callAttached(PresentationModel model, Call call) {
    if (call instanceof SinglePartyCall) {
        fireTableDataChanged(singlePartyCalls_tblModel, dropSinglePartyCall_btn, monitorSinglePartyCall_btn,
                              PresentationModel.SINGLEPARTY_CALLTYPE);
    } else if (call instanceof TwoPartiesCall) {
        fireTableDataChanged(twoPartiesCalls_tblModel, dropTwoPartiesCall_btn, monitorTwoPartiesCall_btn,
                              PresentationModel.TWOPARTIES_CALLTYPE);
    } else if (call instanceof ConferenceCall) {
        fireTableDataChanged(conferenceCalls_tblModel, dropConferenceCall_btn, monitorConferenceCall_btn,
                              PresentationModel.CONFERENCE_CALLTYPE);
    }
}
```

**Fig. 24. The definition of method callAttached in AgiExp.**

Responding to the request, the table will first ask its model for the number of rows that should be displayed (see point 1 in Fig. 25). Then, for each column in the row it will again ask its model for the value to be displayed (2).

---

8 This is a typical way of implementing a `TableModel`. I assume that you're quite familiar with Swing programming.
So that was how “a new row is added to the table”. Now I’m going to explain the “how an entry – in a table – is updated”. When you run the program (later), you will observe that the state of the call changes over the time, until finally it ends up in `INVALID` state. The following is a scientific explanation for that phenomena.

Take a look again at *Fig. 23*, there the `Main` registers itself as a listener to an (new) instance of `Call` passed to it. Being a ✶ `CallListener`, the main class implements the following methods:

- void `stateChanged(int oldState, Call call)`
- void `channelAdded(ConferenceCall conferenceCall, Channel channel)`
- void `channelRemoved(ConferenceCall conferenceCall, Channel channel)`

Everytime the state of the call changes, for example, the call will notify its listeners by invoking the method ✶ `stateChanged(...)` on each one of them. The following figure shows how `Main` reacts to the changes of state of a call:

```java
public void stateChanged(int oldState, Call call) {
    presentationModel.callStateChanged(oldState, call);
    if (call.getState() == Call.INVALID_STATE) call.removeListener(this);
}
```

*Fig. 26. The definition of method stateChanged in Main*
[Finally... ] We're about to go through another interesting part of our journey, interacting with Asterisk that is. I'm going to show you, first, how to receive and handle manager events that come through a manager connection. I'll put it in its own section, for it's quite extensive. So, please, bear with me.

**Receiving and handling manager events**

Any object that wants to receive manager events from Asterisk has to satisfy the following requirements:

- Implements a java interface named `ManagerEventHandler`, which is part of the Asterisk-Java library that we're using. The `AsteriskProvider` class does it.
- Be registered to the manager connection as one of its handlers. A manager connection is an instance of `ManagerConnection` – also part of Asterisk-Java library – that represents a (network) connection to the Asterisk box. The `Main` class does it for the `AsteriskProvider`, by invoking the method `addEventHandler(...) on the manager connection (see Fig. 21 point 📌).

The only method declared in `ManagerEventHandler` is `handleEvent(ManagerEvent event)`. The following figure shows how it is implemented in `AsteriskProvider`. I suggest that you also take a look back at Fig. 9, you'll notice that the lines of code in Fig. 27 is the other form of the diagram in Fig. 9. You can compare them point-by-arrow, e.g.: point 📌 in Fig. 27 corresponds to arrow #1 in Fig. 9.

```java
public void handleEvent(ManagerEvent event) { 1
    boolean processed = false;
    for(Iterator iter = attachedCalls.iterator(); iter.hasNext();)
        call = (Call) iter.next();
        processed = call.process(event); 2
        if(processed) break;
    
    if(!processed) processEvent(event);
}

private void processEvent(ManagerEvent event) {
    boolean processed = false;
    for(Iterator iter = callConstructions.iterator(); iter.hasNext();)
        callConstruction = (CallConstruction) iter.next();
        processed = callConstruction.process(event); 1,1
        if(processed) break;
    
    if(!processed) {
        if(event instanceof NewChannelEvent) {
            NewChannelEvent nce = (NewChannelEvent) event;
            if("Ring".equals(nce.getState())) {
                CallConstruction callConstruction = new CallConstruction(this, nce);
                callConstructions.add(callConstruction); 1,2
            }
        }
    }
}
```

*Fig. 27. The definition of method stateChanged in Main*

By now you might be wondering how do I know that a `NewChannelEvent` in a “Ring” state indicates a new call. My answer: “from analysis of sequence of manager events”. Here, I'll show you....
Analysis of sequence of manager events

The way for an external program to know what's going on in Asterisk is by monitoring it through manager interface. Well, actually, manager interface serves two purposes: (1) monitoring and (2) controlling Asterisk from external programs. We'll focus on the monitoring part in this section.

There's nothing fancy here; Asterisk simply sends manager events to every external program that has a manager connection to it. The manager events received are ordered in time. However, an event is not necessarily related to the other event that came before it. Therefore, the program needs somekind of a state machine to sort out and make sense of the events.

For every distinct event in Asterisk there's a unique type of manager event (sorry). The following figure lists all the types of manager event. In our program we're going to deal only with a small subset of it, namely $ChannelEvent$ (and all its subtypes), $LinkageEvent$, and $MeetMeEvent$.

![Fig. 28. The type hierarchy of class ManagerEvent](image)

I have created a simple program that connects to the Asterisk manager interface, and dumps all the events it receives to the screen. The name of the program is $EventDumper$. Before you can run it we need to make sure that manager interface is enabled in Asterisk and create a manager account in Asterisk.

We do that by modifying the file `/etc/asterisk/manager.conf`. Go to the general section and set the enabled property to yes. It is also a good idea to uncomment the displayconnect properties (so that you can confirm if your program really is connected). You should come up with something like in Fig. 29.

![Fig. 29. General section in manager.conf](image)

To add a manager account, add the following lines to the end of the file. Remember to change the IP
address – 90.0.0.15 in the example below – to the IP address of the machine where you're going to connect from (i.e.: the IP address of the machine where our program will be ran on).

```
[agiexp]
secret=password
deny=0.0.0.0/0.0.0.0
permit=90.0.0.15/255.255.255.0
read = system,call,log,verbose,command,agent,user,action
write = system,call,log,verbose,command,agent,user,action
```

Here are the steps for obtaining sequence of manager events generated by a call:

- Start or **restart** Asterisk (it wouldn't pick up the change you've made to the `manager.conf` unless you restart it).
- Run **EventDumper** from your Eclipse as java application. The program requires that you pass three arguments to it: (1) the IP address of the Asterisk box, (2) the user name of the **manager account** for the connection, and (3) the password of the manager account. You should see something like below in the output console of the **EventDumper** application:

```
Mar 10, 2006 1:20:12 AM net.sf.asterisk.util.impl.JavaLoggingLog info INFO: Connecting to 90.0.0.15 port 5038
class: net.sf.asterisk.manager.event.ConnectEvent
dateReceived: Fri Mar 10 01:20:12 CST 2006
getPrivilege: null
protocolIdentifier: Asterisk Call Manager/1.8
```

- Then start the kphone, this time let's authenticate as user **johndoe**. First, let's find out manager events generated by a **single party call**. Dial **600@90.0.0.15**. All the events during the call are shown in **Fig. 30**.
Let's not jump in to any conclusion so quickly. We yet have to see the events generated by a two parties call. Try making a call from *johndoe* to *goldenboy*. All the events generated are shown in **Fig. 31**.

Try both experiments – making a single party and two parties call – several times, just to be sure. From those logs one thing you can quickly spot is: they all start with `NewChannelEvent` whose state is RING. That's how came to the conclusion that the *AsteriskProvider* should know that when that event came in, a new call should be constructed (i.e. a new call construction should be commenced)\(^9\). Therefore, we have the lines of code as is shown in **Fig. 32** in *AsteriskProvider*.

---

\(^9\) I'm not showing you the events from a conference call. Too noisy. But, believe me, it's the same. Crescent my heart.
Fig. 32. Manager events generated for a single party call

When instantiated the *CallConstruction* will go directly to its first state, that is waiting for *NewExtenEvent* to come in. The Fig. 33 shows the state diagram of a call construction. To be more exact, it's limited to the area within the green border. The events are related to each other by their unique id, with the unique id of the *NewChannelEvent* as the basis. In other words, a particular instance of *CallConstruction* will only process events whose unique id matches the basis.

Fig. 33. State diagram of a call construction
Right after it reaches any of the “creation” states (the ones with red border), the call construction will go directly to its final state. From that point on it no longer expects for events (because subsequent related events will be handled by the call it just created). Therefore it will remove itself from the list of living call constructions maintained by the AsteriskProvider. The lines of code that correspond to Fig. 33 is shown in Fig. 34 (they're within the `processEvent(...)` method defined in class `CallConstruction`).

You might be asking right now how do I decided, for example, to instantiate a `SinglePartyCall` when an instance of `NewStateEvent` whose state is UP and application is not `MeetMe` is received. Well, firstly, I've made comparisons between various call scenarios. So I just know (believe it like you believe in god10 – ha!). Second: it's because at that moment the program will already have all the information it needs to instantiate a `SinglePartyCall`.

---

10 If you don't believe in god, you're lost dude. I have no time to convince you and do a salvation for you. Hehehe....
Ok, no more questions. Let's see how a call handles manager events passed to it. We'll start with single party call. The following figure shows the state diagram of a single party call.

![State Diagram of single party call](image1)

Fig. 35. State Diagram of single party call

And... Fig. 36 below shows how a two parties call handles manager events passed to it.

![State Diagram of two parties call](image2)

Fig. 36. State Diagram of two parties call

One thing in common between the way a single party and two parties call filter the events (such that only “relevant” events could affect its state) is that they check the name of the channel of the event – by calling `getChannel()` on the event – and compare it with the id of the descriptor of the channel of the call (phew!).

The following figure shows the java code representation of Fig. 36. It's the code within the method `processEvent(...)` in the class `TwoPartiesCall`. To get a very good grasp of the concepts and mechanics, I
encourage you to play with the debugger and experiment with various scenarios such as where (1) goldenboy doesn’t register to Asterisk, (2) johndoe hangs up the call before goldenboy picks up the call, and (3) goldenboy rejects the call.

Fig. 37. Source code representation of Fig. 36
Testing (#1)

Now let's try running our program (yay!). We'll first do functional testing manually. Automatic testing will be explained later. Here are the steps you should follow:

1. Make sure that Asterisk is running..., with the manager interface enabled.

2. Run the main class of our program, namely `net.raka.agiexp.gui.Main`. You can do that directly from Eclipse. Don't forget to pass in the following program arguments:
   1. The IP address of the Asterisk box.
   2. The account name in manager interface (e.g.: `agiexp`).
   3. The password of the account in manager interface (e.g.: `password`).
   4. The port number where Asterisk's SIP service is bound to (by default it's 5060. Check the value of property named `bindport` in the general section in `/etc/asterisk/sip.conf`).
   5. The IP address of the machine where you're running this program. Don't use `127.0.0.1` or `localhost`, use actual IP number. This value is used by conference call monitor, which I'll explain later.
   6. The port number where the conference call monitor will be bound too. Use any port which is not occupied (e.g.: 7000). This value is also used by conference call monitor.

3. Make the calls:
   1. Try calling 600 from `johndoe`, and watch the table in the tab named “Single Party Calls” in our program.
   2. Try calling `goldenboy` from `johndoe` (dial 666). Watch the table in the tab “Two Parties Calls”. Experiment with the scenarios I mentioned previously.
How was it? I hope you were satisfied. Did you notice that when you clicked on the row that displays an active call, the two buttons at the bottom right corner of the panel became active?

- Clicking the Drop button, the call will be, well, dropped (all the parties connected to the call will be kicked out, and at the end the call will become invalid).
- You can click the Monitor button to record the conversation within the call. I specified in the code that the file will be saved in /var/recordings.

Umm..., you've guessed it. We're now getting to the other use of Asterisk manager interface, for controlling. So let's get back to code (we have much to learn).
Implementation Again (Controlling Asterisk)

The action handler of those buttons are defined in the class AgiExp. The clicking on the drop button, for example, will end up in the `drop(Call call)` method declared in Provider. The following diagram shows the path that the clicking will go through to reach the `drop(...)` method in Provider.

Fig. 38. Members calling 'drop(Call)' – in workspace

So now let's take a look at the implementation of the `drop(...)` method in AsteriskProvider.

```java
public void drop(Call call) {
    if (call instanceof SinglePartyCall) {
        SinglePartyCall spc = (SinglePartyCall) call;
        HangupAction hangupAction = new HangupAction(spc.getChannel().getDescriptor().getId());
        try {
            managerConnection.sendAction(hangupAction);
        } catch (Exception e) {
            throw new RuntimeException(e);
        }
    } else if (call instanceof TwoPartyCall) {
        TwoPartyCall tpc = (TwoPartyCall) call;
        HangupAction hangupAction = new HangupAction(tpc.getDialingChannel().getDescriptor().getId());
        try {
            managerConnection.sendAction(hangupAction);
        } catch (Exception e) {
            throw new RuntimeException(e);
        }
    } else if (call instanceof ConferenceCall) {
        ConferenceCall cc = (ConferenceCall) call;
        for (Iterator iter = cc.getChannels().iterator(); iter.hasNext()) {
            Channel channel = (Channel) iter.next();
            HangupAction hangupAction = new HangupAction(channel.getDescriptor().getId());
            try {
                managerConnection.sendAction(hangupAction);
            } catch (Exception e) {
                throw new RuntimeException(e);
            }
        }
    }
}
```

Fig. 39. Members calling 'drop(Call)' – in workspace

So... controlling Asterisk from external program is as simple as sending out an instance of `ManagerAction` through the manager connection. In the case of dropping a call, we send an instance of `HangupAction`. For recording a call, we send an instance of `MonitorAction`. The following figure displays all the standard manager actions. But it doesn't mean all of them can be used. The number of actions available are determined by the modules presently loaded in the Asterisk.
In most cases, maybe, the standard actions provided by Asterisk-Java suffice to get your job done. But, in case your fancy requirement requires (ugh) you to send an action that’s not in the list above, you can have your custom action by making a class that extends `AbstractManagerAction`. Of course, on the Asterisk side, there must also be a module loaded that can handle your custom action.

*(What’s the guideline for implementing a custom action – other than extending `AbstractManagerAction`?)*

Not much. I see that an instance of manager action that we throw into the manager connection is sent through the wire as a plain text, in a very simple format..., key-value pairs. The following is what I got from Ethereal when the program sent a `MonitorAction`:

```yaml
action: Monitor
actionid: 421988_3#
file: /var/recordings/-1059416815
channel: SIP/johndoe-d54a
mix: true
format: wav
```

That text is constructed out of the properties of the action. You can verify this information by taking a look at the source code of `MonitorAction`. The class has the following methods: `getAction()`, `getActionId()`, `getFile()`, `getChannel()`, `getMix()`, and `getFormat()`. If you’re still unconvinced – what an infidel – then check out the implementation of the method `buildAction(...)` in class `ActionBuilderImpl` (part of Asterisk-Java).

*Conclusion: if you have a third party module that provides externally accessible extra functionalities installed in your Asterisk, and you want to be able to access them from your Java program, then ask for the documentation that explains the key-value pairs accepted / required by those functionalities. Then, on the Java side, you only have to make sure that your custom manager action(s) has the appropriate getters (for all the keys specified in the documentation)\(^{11}\).*

\(^{11}\) Just follow the JavaBean convention for the naming of the getters.
Implementation Returns (Conference Call)

Conference call is the final piece in our program. I put it the last, separated from the explanation about the two other calls, because (1) there are other configuration things to do in order to make it work (*a tiny hassle!*), and (2) there are other new concepts and techniques that I'd like to introduce.

Zaptel installation

To make the conference feature – using MeetMe – available on your Asterisk, you have to do several things. First of all, *MeetMe* requires a timing device in order to operate (Asterisk won't even compile it if no timing device is found). All Digium PCI hardware provides a 1-kHz timing interface. If you lack the PCI hardware required to provide timing, the *ztdummy* driver can be used as a timing device\(^\text{12}\). So..., you need to install zaptel drivers on your Linux. Here's a quick instruction:

1. **Download** [zaptel-1.2.x.tar.gz](http://asterisk.com) from Asterisk's website [Res. 2], and unpack it.
2. I assume that you don't have Digium PCI hardware. So, you need to have ztdummy compiled too. *Edit* the *Makefile* (of zaptel), and look for the following section (and remove the hash that precedes ztdummy):

   ```
   MODULES:=zaptel tor2 torisa wcusb wcfxo wctdm wctdm24xxp \ 
   ztdynamic ztd-eth wctlxxp wct4xxp wctellxp pciradio \ 
   ztd-loc # ztdummy
   ```

   *) If your Linux uses kernel 2.6.x, you don't have to do this.
3. Finally do the: (1) *make clean*, (2) *make*, (3) *make install*, and (4) *make config*.
4. Re-compile and re-install Asterisk.

That's all there is to it. Well, not really. At least not in my case. I don't know what is wrong with my Linux (or zaptel installation), I have to do the following ritual before starting Asterisk:

```
rahaome$ whoami
root
rahaome$ # modprobe zaptel
rahaome$ # mkdir /dev/zap
rahaome$ # mv /dev/zapchannel /dev/zap/ctrl
rahaome$ # mv /dev/zapchannel /dev/zap/ctrl
rahaome$ # mv /dev/zapchannel /dev/zap/ctrl
rahaome$ # modprobe ztdummy
```

*Fig. 41. My silly ritual*

Meetme configuration

The configuration options for the MeetMe conferencing system are found in `/etc/asterisk/meetme.conf`. Inside the configuration file, you define conference rooms and optional numeric passwords (if a password is defined here, caller will be required to enter a password for entering the conference room).

To define a conference room (with number 400, and without password), simply add the following line to the end of `meetme.conf`:

\(^{12}\) Again, I copied-and-pasted several sentences from [Ref. 2] – Chapter 3, on Compiling Zaptel.
Now, to support such “requirement” as: “if the caller dials the number 5XXX, then she should be joined to conference room #XXX”, we have to add the following lines in the *extension.conf* (put them in the section for *default* context):

```
exten => _5XXX,1,MeetMe(${EXTEN:1},i)
exten => _5XXX,n,Hangup
```

There you have your conferencing system set up. Re-start Asterisk, followed by making a call (from *johndoe*) to the number 5400 and once you’re connected to the conference room, just follow the instruction on the phone (*after the tone bla bla bla...*)\(^{13}\). Well, I guess a conference wouldn't be very fun with only one person in the room. So, try making a call from other UA (e.g.: *goldenboy*) to 5400\(^{14}\). Observe! You can start with checking Asterisk’s status from the CLI (see Fig. 42).

---

![Fig. 42. Checking channels & meetme status](image)

### Back to our program

The logic of creation of a conference call goes like this (this is to clarify the diagram in Fig. 33):

- Incoming calls to a conference room go through the same state transitions, that is: *waiting for NewExtenEvent* → *waiting for NewStateEvent* → *waiting for MeetMeJoinEvent* → *creation of conference call*.

- However, only the first call should cause the creation of an instance of *ConferenceCall* (1). As long as the first call hasn’t left the conference room, the subsequent calls should be added – as a new channel – to the instance of *ConferenceCall* created previously (2).

The logic above translates to the following java codes in the class *CallConstruction*:

---

\(^{13}\) You need to install a package named *asterisk-sound* to be able to hear the instruction. Asterisk-sound is simply a collection of audio files.

\(^{14}\) It’s better to carry on this experiment in a local area network with several computers (so that you can really play with sound – remember? the sound card exclusive access problem). But if you’re only interested in the signalling, it’s ok to run all the phones in a single computer.
Well, there was nothing new in the above explanation (just a reiteration of processing of manager events). The two new interesting things that I promised are integration using Asterisk Gateway Interface (AGI) and programming with SIP stack library. Let's begin with AGI.

```java
case MEETMEJOIN_STATE:
    if(event instanceof MeetMeJoinEvent) {
        MeetMeJoinEvent mje = (MeetMeJoinEvent) event;
        NewStateEvent nse = (NewStateEvent) eventStack.pop();
        NewExtenEvent nee = (NewExtenEvent) eventStack.pop();
        NewChannelEvent nce = (NewChannelEvent) eventStack.pop();
        String roomId = mje.getRoomId();
        boolean confCallExist = false;
        for(Iterator iter = provider.getAttachedCalls().iterator(); iter.hasNext();) {
            Call attachedCall = (Call) iter.next();
            if(attachedCall instanceof ConferenceCall) {
                ConferenceCall confCall = (ConferenceCall) attachedCall;
                if(confCall.getRoomId().equals(roomId)) {
                    confCallExist = true;
                    Channel.ChannelDesc newChannelDesc = new Channel.ChannelDesc(nce.getChannel(),
                        nce.getDateReceived(), new CallEndpoint(nee.getExtension()));
                    confCall.addChannel(newChannelDesc);
                    provider.removeCallConstruction(this);
                    return true;
                }
            }
        }
        if(!confCallExist) {
            Channel.ChannelDesc channelDesc = new Channel.ChannelDesc(nce.getChannel(),
                nce.getDateReceived(), new CallEndpoint(nee.getExtension()));
            ConferenceCall confCall = new ConferenceCall(roomId, mje.getDateReceived(), channelDesc);
            provider.removeCallConstruction(this);
            provider.attachCall(confCall);
            return true;
        }
    }
    break;
```

Fig. 43. Conference call creation logic
Sorry for the interruption, but I have to give some words of caution to you. It's about the state machine. I don't want to (mis)lead you to think that the state machine in this program is reusable (can be used in all kinds of situation involving detection of creation and destruction of a call in Asterisk).

I would say that a state machine is highly related (coupled?) to the dialplan. Even a little modification to the dialplan might render the state machine useless. For example, if you used a function other than `Dial` to make an outbound call from your Asterisk, then you would not see the two parties call show up in the user interface. It's because our state machine assumes that the lifecycle of a two parties call begins when a `NewExtenEvent` – whose `application` property is set to `Dial` – is received (see again Fig. 34, blue shaded area).

So, I think, in developing a telephony application (using the approach similar to the one we use here) first you have to understand the requirements for the dialplan. Then you have its structure well defined, so that you know all the possible sequence of events. Finally you design your state machine based on that.
Asterisk Gateway Interface (AGI)

The Asterisk Gateway Interface, or AGI, provides a standard interface by which external programs may control the Asterisk dialplan.

You must have realized that the “language” we use in the `/etc/asterisk/extensions.conf` to craft the dialplan is very simple (and limited). It supports branching in a very limited way, through the use of *dialplan functions* named *GotoIf*. It doesn't even seem to support looping. We can do simple string pattern matching, though, like we did with the extension to access a conference (_5XXX).

Clearly other – additional – ways of controlling a dialplan is required..., with other languages which are more capable (and more popular). That's the case for AGI; it's what CGI is to webserver to Asterisk. You can program an AGI in languages such as C, python, Java, and maybe some others.

The AGI script can be deployed in the same machine as the Asterisk (as is the common case with C or python-based AGI script), or on a remote machine (in which case, the communication is conducted between Asterisk and the AGI server that hosts the script through a socket, using FastAGI protocol).

In our program we take the later approach, and the programming language that we use to create the script is Java. The library *Asterisk-Java* comes with classes that allow us to run an AGI server in embedded mode (in addition to a standalone AGI server program). Now let's take a look how we call our remotely deployed AGI script from the Asterisk. Replace the following lines of code (that you put earlier) from `extensions.conf`:

```plaintext
exten => _5XXX,1,MeetMe(${EXTEN:1},i)
exten => _5XXX,n,Hangup
```

...with...

```plaintext
exten => _5XXX,1,Agi(agi://90.0.0.15/conferenceMonitor?roomId=${EXTEN:1})
exten => _5XXX,n,MeetMe(${EXTEN:1},i)
exten => _5XXX,n,Hangup
```

Based on the configuration above, when a call enters the extension 5XXX, a request will be sent to an AGI server running on the machine 90.0.0.15 to execute a script which is mapped to an identifier *conferenceMonitor*. A parameter named *roomId*, whose value is set to XXX, will be passed along the request. On successful return from the execution of the script, the next line in the dialplan will be executed.

To have a custom AGI server, you have to create a class that extends `DefaultAGIServer`. Furthermore, the instance of that server must have its *mapping strategy* set. A mapping strategy determines which instance of `AGIScript` has to be called to service a given `AGIRequest`.

We set the mapping strategy by calling `setMappingStrategy(...)` on that instance of AGI server\(^{15}\). In our case, we assign an instance of `ConfMonitorMappingStrategy` to our server, that returns an instance of `ConferenceMonitorScript` for the requests made to *conferenceMonitor*.

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\(^{15}\) The mapping for standalone AGI server is specified in a properties file named `agiserver.conf`. 
Our server is launched during the initialization of our program. We do that by starting a new thread – from the `run()` method in our `Main` class – that instantiates and run the server (see Fig. 45).

An AGI script, in our java program, is an instance of a class that extends `BaseAGIscript`. The class has to implements the `service(...)` method. The following figure shows the implementation of the `service(...)` method in `ConferenceMonitorScript`. 

```java
public class ConfMonitorAgiServer extends DefaultAGIserver {
    private ConferenceMonitorScript confMonitorScript;

    public ConfMonitorAgiServer(ConferenceMonitorScript confMonitorScript) {
        if (confMonitorScript == null) {
            throw new IllegalArgumentException("confMonitorScript can not be null");
        }
        this.confMonitorScript = confMonitorScript;
        setMappingStrategy(new ConfMonitorMappingStrategy());
    }

    class ConfMonitorMappingStrategy implements MappingStrategy {
        public AGIScript determineScript(AGIRequest request) {
            if ("conferenceMonitor".equals(request.getScript())) {
                return confMonitorScript;
            }
            return null;
        }
    }
}
```

**Fig. 44. Source code of ConfMonitorAgiServer**

**Fig. 45. Starting embedded AGI server**
Fig. 46. The service(...) method of ConferenceMonitorScript

Our script simply delegates the call to an instance of ConferenceMonitor, asking it to monitor a conference with the given roomId.

Why all these intricacies (with the script, conference monitoring, and all)?

Well, with the script, basically we want to intercept calls to a conference room, such that we can sneak in a dummy (silent) participant to the room (if we haven't already done so). Sneaking in that dummy participant is the task that we assign to conference monitor. How does it accomplish the task? It's the topic for the next section....
A Sip of SIP

First of all..., what is SIP? It’s an end-to-end, client-server signalling protocol. Signalling in telephone system is the key mechanism by which telephone calls are set up and terminated. Many value added services in a telephone system are made possible by having programs that mingle with the signalling process. Examples of value added services are: PBX features, group calling, click to connect, and many kinds of intelligent network (IN)\textsuperscript{16} services.

Signalling on SIP is carried through IP network (therefore it’s a VoIP protocol). It is specified in an RFC numbered 3261. The kinds of entity in the network that can deal with signalling messages are\textsuperscript{17}:

- User Agent (user application), that can be further classified into 3 types:
  - UA Client (originates calls)
  - UA Server (listen for incoming calls)
  - B2B (back-to-back) UA
- SIP Proxy Server: relays call signalling, i.e. acts as both client and server.
- SIP Redirect Server: redirects callers to other servers.
- SIP Registrar: accepts registration requests from users.

One thing from SIP that attracts me most is the fact that it uses plain text for the messages used in the signalling. The format is also simple, attribute-value-pair (that’s looks very much like HTTP messages). So, as long as you understand the protocol specification, and know how to parse and construct those messages in your program, you’re good to go\textsuperscript{18}. The SIP entities mentioned above are basically (SIP) message processors, that behave in such way that they conform to a specification. The following figure shows an example of SIP message.

\begin{center}
\includegraphics[width=0.5\textwidth]{sip-message.png}
\end{center}

\textbf{Fig. 47. An example of SIP message}

\textsuperscript{16} Intelligent network is a collection of servers and other resources used to control call setup and to provide media services such as announcements, voicemail, etc.

\textsuperscript{17} Here “deal with” means sending, accepting, modifying, or passing / redirecting.

\textsuperscript{18} Well, maybe in the harsh reality it’s not that simple.
In our program, we implement a very simple UA client. To help us with the construction and parsing of SIP messages (among other things) we use NIST-SIP that is an open source implementation of a specification named JAIN SIP issued by Java Community Process.

This UA client represents a dummy participant that we put in a conference room. For each active conference we have a distinct instance of UA client. As a client, the only things it does are: (1) making a call to the conference room, and (2) disconnecting itself from the conference room when the conference ends. The details of those operations are encapsulated in a class named `NoteTaker`, that has the following methods:

- `joinConference(String roomId)`
- `leaveConference()`

Instantiation of `NoteTaker` and invocation of its `joinConference(...)` method take place inside the execution of the `monitorConference(...)` method of `ConferenceMonitor` (see Fig. 47).

```java
public void monitorConference(String roomId) {
    if(roomId == null) {
        throw new IllegalArgumentException("roomId can not be null");
    }
    synchronized (noteTakers) {
        NoteTaker noteTaker = new NoteTaker(roomId);
        noteTaker.joinConference();
        noteTakers.put(roomId, noteTaker);
    }
}
```

**Fig. 48. Implementation of monitorConference(...) in ConferenceMonitor**

You can say that `ConferenceMonitor` is all ears, because it implements so many listener interfaces: `SipListener`, `ProviderListener`, and `CallListener`.

- As a SIP listener, it processes response events it receives from the SIP peer, inside the `processResponse(...)` method (where it has the chance to send the SIP ACK message, that marks the completion of a call set up, back to the peer).
- As a provider listener, it registers itself as listener to a conference call whenever it receives a notification that a (conference) call has been attached to the provider.
- As a call listener, it listens to the dynamic of a conference call. Inside the `channelRemoved(...)` method, it has the chance to check the number of remaining participants in the room, and ask the `NoteTaker` to disconnect itself from the room if the number is 1 (see Fig. 49).

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19 If you're a regexp guru, for example, you might think you don't need library just for text processing. However, there are some more high level concepts above SIP messages, such as `dialog` and `transaction`. If you'd like to work with them easily, you'd want to use specialized library like JAIN-SIP.

20 An interface defined in JAIN-SIP.

21 The other SIP entity that it communicates with, which in our case is the Asterisk box.
We get to the important questions now: (1) what messages are exchanged during call set up and tear down, (2) how do we construct / parse them, and (3) how do we send them. Let's begin with the messages. The following figure shows the signalling of a normal case of session setup and tear down.

In SIP parlance, making a call is “sending an invitation” (to the callee). The caller does that by sending a SIP request whose method name is set to INVITE (point 1 in Fig. 50). The answerer will send the response to the location destination specified in the first Via field in the request, which in our example it is port 5062 on machine with IP 90.0.0.15 (point 2).

The location specified in the Via field is not necessarily the same as the location of the caller. It's possible that the request has travelled through proxy(s) before it reached the destination (where each proxy in the signalling path inserted a Via field before the first Via field found in the request). However, we can always tell the ultimate destination for the response from the value specified in the Contact field in the request (point 3).
The INVITE request contains a *session description* that indicates the desired communication means (audio, video, etc), parameters of those means (such as codec types), and address addresses for receiving media from the party who will answer the call. In our example, the sender specifies that it supports codecs PCMA, iLBC, GSM, and PCMU. It also specifies that it's waiting for the media on port 1032 (point 4).

The OK response from the answerer also contains a session description. In our example it says that it supports GSM, PCMU, and PCMA, and is receiving media on port 16454 (point 5).

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**What's the role of Asterisk in a SIP call?** The answer is: it can be either a back-to-back user agent (B2B UA) or a SIP proxy. As a B2B UA it stays in the media path of the call (which means, there will be 2 media connections in the call: (1) between caller and Asterisk, and (2) between Asterisk and the answerer). As a SIP proxy, there will be direct media connections between the caller and the answerer. (see Fig. 51).
Asterisk will act as a proxy if both the caller and the answerer supports Re-INVITE, which is a mechanism of modifying the session description during the call. For UA that we register in sip.conf, we can set its re-invitiability by setting the canreinvite property to no. In our program we have them all set to no. The reason is because we use Asterisk's built-in Monitor application for recording the call. That application requires that Asterisk be in the media path.

Another valid reason for operating Asterisk as B2B UA is to free us from worrying too much about compatibility issues (esp. regarding codec supports) between the UAs involved in the call, because Asterisk will iron out the incompatibility (if any) by performing transcoding.

I'm not going to explain how to use JAIN-SIP API in detail. I suggest that you take a look at the implementation of joinConference() and leaveConference() method in NoteTaker and decipher the code by yourself. Hints:

- From inside the joinConference() we send the INVITE
- From inside the processResponse() we send the ACK (when appropriate).
- From inside the leaveConference() we send the BYE.

**A bit more configurations for the conference...**

First, we need to register a UA to the Asterisk, that can only makes calls to it. This is for the notetaker. So, first, copy and paste the following lines to the end of your sip.conf.

```
[notetaker]
type=friend
host=90.0.0.15
port=7000
canreinvite=no
context=notetaker
```

We set the host to a specific IP address instead of dynamic. The port, too, is set to a specific number. The reason is because we want to keep the implementation of NoteTaker very simple (we don't want to implement registration step in our code). Therefore, it's important that we run our program on a
machine whose IP address matches the *host* we specify here, and we must also tell our program – through execution parameters – to bind the *notetaker* to the port we specify here (7000).

We also specify that calls from *notetaker* be put in a separate context named *notetaker*. This is to make sure that calls from *notetaker* – that should be directed toward a conference room – will not be confused with calls from other UAs (that might go to somewhere else). I’ll explain, but first copy and paste the following lines to the end of *extensions.conf*.

```plaintext
[notetaker]
exten => _XXX,1,MeetMe(${EXTEN},i)
exten => _XXX,n,Hangup
```

Remember that in the *extensions.conf*, for calls that enter the extension 5XXX, we drop the first number from the extension (5) and pass the rest it (XXX) to the *conferenceMonitor* script as a parameter named *roomId*? ... Right, you do remember it :). Well, the *notetaker* will send an INVITE request to *XXX@your.asterisk.host*.

Now, let's suppose we didn't put the two lines above – anything except *[notetaker]* – in a separate context that is exclusive for calls from UA *notetaker*. And also, suppose that we had a conference room with number 666 (registered in *meetme.conf*). Then (bummer!) the call from *notetaker* would directed to *goldenboy* (assuming that we also have defined extension 666 for *goldenboy* in the same context).

That's all for the SIP.
Let's move on to the last section (yay!)..., stress testing with SIPp.
**Stress Testing with SIPp**

SIPp is an (open source) performance testing tool for SIP protocol. We use it to simulate multiple UAs making a call at the same time. You must have realized that a call session (1) starts with the sending of an invitation (or accepting invitation), (2) is terminated with tear down (sending / receiving a BYE), and (3) anything that happen in between. Together, they make a call scenario.

With SIPp, we specify the scenario in an XML file that contains the SIP requests to be sent, and the expected responses. They're processed in document order – from top to the bottom of the document – when the scenario is being played. In the file, a SIP request is enclosed within a `<recv>` tag, while a `<resp>` tag contains information about an expected response. For more information about the content of a scenario file, please consult SIPp documentation [Ref. 4].

In our experiment we have two scenario files. One of them represents the scenario from the standpoint of the caller\(^ {22}\), and the other one represents the scenario of the same call from the standpoint of the answerer\(^ {23} \).

\[\text{Sidenote. How do I know what to write in the scenario files? The answer: copy-and-paste (and a bit of editing) from SIP traffic of a real call, made from SIP phone that I know works (e.g.: Kphone and Xlite), captured using Ethereal.}\]

For example: I know that Asterisk SIP service was bound to port 5060, and KPhone was bound to another port (let's say 5062). What I did was a copy-and-paste of SIP messages originating from port 5062 and going port 5060 into `caller.xml`, and vice versa for the `answerer.xml`.

Easy, eh? Believe me, Ethereal is a (network) programmer best friend :). Here's a nice screenshot of it:
In order to make things simpler for the testing, we need to lift the requirement for the caller to authenticate. We do that by deleting / commenting out the secret line in the user agent configuration parameters in sip.conf (see Fig. 52).

**Fig. 52. Comment out the secret line**

The other thing we need to do is dropping the requirement for the user agents which are assigned to receive calls to register. This is done by changing the host property of the user agent from dynamic to an IP number, and by specifying a port number in the sip.conf. And, just like the caller, we also drop the requirement for the answerer to authenticate. See Fig. 53.

**Fig. 53. Change to static registration**

Now you're set! I have prepared two script files: caller.sh and answerer.sh. The following figure shows the content of caller.sh.

```
4 /usr/local/bin/sipp 9.0.0.15:5060 -l 9.0.0.15 -inf caller.csv
  -r 5 -rp 1000 -sf caller.xml -t un -trace_err -n 5 -trace_msg
```

**Fig. 54. The content of caller.sh**

According to the script: the instance of SIPp will play the scenario specified in caller.xml. The maximum number of calls that the SIP instance will generate is 5, and each user agent associated with a call will be bound to a randomly selected distinct port. The SIP request will be sent to SIP service running on port 5060 in machine 9.0.0.15, which in our case is the Asterisk's SIP service. Finally, the values to replace the “fields” in the scenario file are read from a file named caller.csv, see Fig. 55.

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24 This is due to current limitation in SIPp, which I'll explain at the end of this section.
The other side – `answerer.sh` – is configured to run in UDP mono socket mode (`-t ul` command line parameter) that means all the user agents it starts will be bound to the same port number (specified through `-p` command line parameter), which in our case is 600125.

Now we're ready to (stress) test the application. Follow these steps:

1. Run our application from Eclipse (or from the command line by execution the `runapp.sh` script)
2. Run the `answerer.sh`
3. Run the `caller.sh`

Good luck, and enjoy. For your reference, here's a screenshot of execution of `answerer.sh` & `caller.sh`.

25 It must match the port number we specify in `sip.conf` for user agents that we assign to answer the calls (e.g.: `goldenboy`).
Analysis of problem with registering from SIPp.

There are at least 2 possible modes of execution of SIPp: server mode and client mode. In the testing scenario for our software, specifically for the case of inbound call, we would like the SIPp instance that simulates the UA (user agent) of a contact center agent to be running in server mode. On the other hand, the SIPp instance that simulates the UA of a customer who makes a call to the contact center should be running in client mode.

When running in server mode, SIPp maintains map of active calls that it is handling. Each call in the map is identified by its Call-ID. Whenever SIPp receives SIP message whose Call-ID doesn't have corresponding entry in the map, it will create a new instance of call (in the memory) and store that call in the map (see sipp.cpp line 2060). I use SIPp version 1.1 (unstable) snapshot 2006-01-20 (... and this problem continues to exist in the latest snapshot, 2006-03-15).

When running in client mode, that message will be considered as out-of-call message, and will be treated as such (discarded); see sipp.cpp line 2091.

Unfortunately, we can't set the mode through somekind of execution option(s)..., SIPp will figure it out by itself, in an "interesting" way. It will set its mode of execution to server mode only if the first instruction in the scenario file is a <recv> (see scenario.cpp line 521). Previously we had a problem related to this, because in our old answerer.xml the first message is a <send> that sends a SIP message for registering the location of the UA of the agent to the Asterisk (such that Asterisk knows where to forward the SIP message to whenever it receives an incoming INVITE from the UA of the customer).

We need to adapt to this situation by removing the need for the UA of the agent to register itself, so we can comment out all the instructions that precedes the <recv> for INVITE in the answerer.xml.

This can be achieved by modifying /etc/asterisk/sip.conf. Previously, for each agent in that file, we set the host property to dynamic. Now we need to switch to static mode, by specifying an IP address to that property. Additionaly, we need to specify the port (to which the UA of the agent is expected to listen for SIP messages). In our case, we specify that all agent UAs will be listening to the same (UDP) port: 6001. Correspondingly, we need to set the transport mode of the instance of SIPp for the agent's UA to UDP Mono Socket (in answerer.sh we now have the -t u1 -p 6001 switch).

Analysis of problem with authenticating from SIPp.

The auth_uri included by SIPp in the response to the proxy authentication challenge doesn't match the one which is expected by Asterisk. The one generate by SIPp is composed of remote_ip:remote_port (e.g.: 90.0.0.15:5060), while Asterisk expects it to be service@remote_ip:remote_port.

The auth_uri sent by SIPp can be forced to a value specified through -auth_uri command line parameter when running SIPp. However, with this trick all the user agents generated by SIPp can only make a call to one SIP URI, that is the value of auth_uri parameter.

That's not appropriate in our situation because we want to have multiple agents making calls to different destinations: some of them call 600, the other ones call 666, and the rest call 5400 & 5500.
References

5. RFC 3261: http://www.ietf.org/rfc/rfc3261.txt

Resources

1. Source code of the program: http://www.simitel.com/resources/booklet1/article_1.tar.gz
2. Asterisk website: http://www.asterisk.org
5. Ethereal: http://www.ethereal.com
6. JTAPI: http://java.sun.com/products/jtapi/index.jsp
7. NIST-SIP library: http://jain-sip.dev.java.net

About the author...

Cokorda Raka Angga Jananuraga, a balinese born in 1978. He moved from Bali to Mexico in 2005, and currently is working for a telecommunication software company named Simitel. He loves to eat the greasy suckling pig (babi guling), a traditional balinese food.... What else? Hmm... I guess that's all.

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Troubleshooting

1. **Q:** My softphone freezes when I tried making call from it.
   
   **A:** Is your computer disconnected from the network?
   
   If yes, make sure that in the *routing configuration* of your network device (ethernet) the *default gateway* is set to none. I haven't got a chance to find out why. It happened to me once, and setting default gateway to none solved my problem.
# Table of Contents

1. Preface  
2. Audience  
3. All You Need Is...  
4. Mission Briefing  
5. The Design, a Static View  
   1. Single Party Call  
   2. Two Parties Call  
   3. Conference Call  
6. The Design..., the Dynamics  
7. Setting Up the Environment  
   1. Asterisk  
   2. sip.conf  
   3. extensions.conf  
   4. Your first call  
   5. Your second call  
8. Implementation  
   1. Receiving and handling manager events  
   2. Analysis of sequence of manager events  
9. Testing (#1)  
10. Implementation Again (Controlling Asterisk)  
11. Implementation Returns (Conference Call)  
   1. Zaptel Installation  
   2. MeetMe configuration  
   3. Asterisk Gateway Interface  
   4. A Sip of SIP  
   5. A bit more configurations for the conference  
12. Stress Testing with SIPP  
13. References  
14. Resources  
15. About the Author  
16. Troubleshooting
I hope you enjoyed reading the document as much as I enjoyed writing it. Also, I hope that you find the document useful, for you, or for member of your team. Please send your feedback (critics, suggestions, etc.) to rakabali78@yahoo.com. Because I need to know whether or not I've made myself clear enough, in an efficient way. I'm looking forward to your feedback! In the meantime, I'll be with these guys preparing satay. Cya.