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# Rapid VoIP Development

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# Agenda

Sip Servlets Model

Seam and Seam Telco Framework

Torquebox Telco Framework

Other useful frameworks

IDE Tools

Examples

# Sip Servlets 1.1 Programming Model

Extends HTTP Servlets API

Enables reusability

- .. of Web Framework Concepts

- .. of Web IDE Tools

- .. of Web Deployment Patterns

Application Composition and Routing

**Easy to Learn**

Hides SIP complexities

Lacks Media

# JBoss Seam Overview

An application framework to simplify Web and Enterprise Java Development

Independent core (IoC container, bijection, scoping, events, interceptors, etc)

Enhances JSF

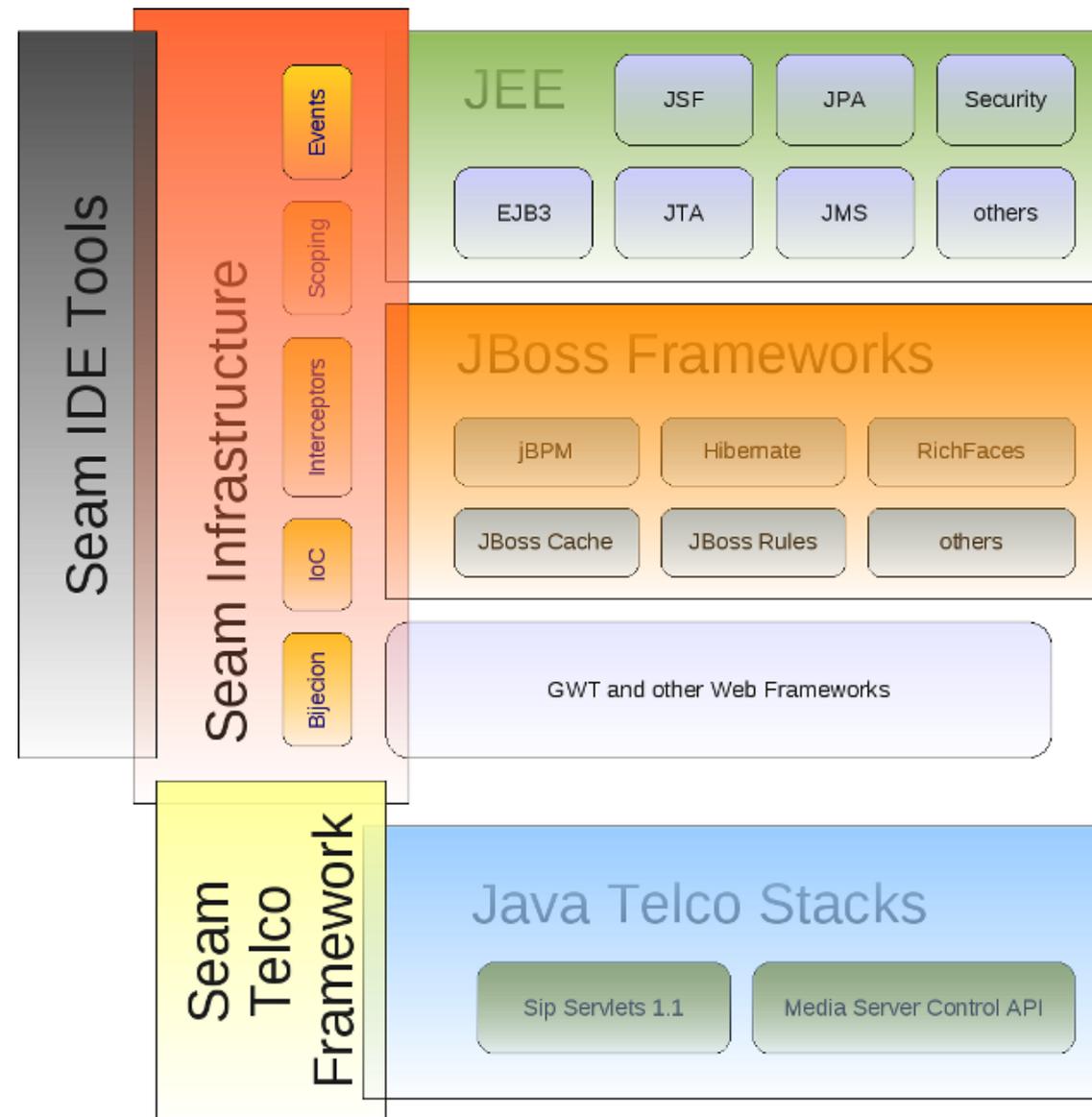
Integration with Java EE components

Integration with JBoss frameworks (jBPM, JBoss Rules, RESTeasy and others)

Integration with third parties (Spring, GWT, JSF libraries)

# Seam Telco Framework (STF)

- Aiming at unification and flat learning curve
- Combine SIP, Media and JEE
- Minimum new APIs.
- Wire the Telco stacks to the rest of the application through the Seam core infrastructure.
- Reuse the Seam development tools (no plugins needed)
- Fast hot-deployment



# STF Technical Overview

The Session scope is mapped to SipSession (roughly spans one single phone call)

One Application → One Seam Application Scope

All SIP and Media events occur in the context of the Session scope for the call

Protocol-agnostic application logic

Cooperation with the Web parts of the application

Reduce code volume for common applications -  
Conferences, IVR, Converged Web+Telco applications

# STF and Converged Applications

Shared components.xml for the SIP and the Web part

Shared application scope to store application-wide state/data

Push technology for desktop-like interface

GWT

Richfaces and A4J

Good for shared authentication of SIP and Web users

Allows calling SIP-aware components from JSF EL

```
<h:commandLink value="Dial"  
  action="#{callAction.dialParticipant(someParticipant)}"/>
```

```

@Name("conference")
@Scope(ScopeType.STATELESS)
public class Conference {
    @In MediaController mediaController;
    @In SipSession sipSession;
    @In(scope=ScopeType.APPLICATION, required=false)
    @Out(scope=ScopeType.APPLICATION, required=false)
    String conferenceEndpointName;
    @Observer("INVITE")
    public void doInvite(SipServletRequest request) throws Exception {
        String sdp = new String((byte[]) request.getContent());
        request.createResponse(180).send();
        sipSession.setAttribute("inviteRequest", request);
        if (conferenceEndpointName == null)
            conferenceEndpointName = "media/trunk/Conference/$";
        mediaController.createConnection(conferenceEndpointName).modify("$",
            sdp);
    }
    @Observer("connectionOpen")
    public void doConnectionOpen(MsConnectionEvent event) throws IOException {
        conferenceEndpointName = event.getConnection().getEndpoint().getLocalName();
        SipServletRequest request = (SipServletRequest) sipSession.getAttribute("inviteRequest");
        SipServletResponse response = request.createResponse(200);
        response.setContent(event.getConnection().getLocalDescriptor(), "application/sdp");
        response.send();
    }
}

```

```
@Observer("DTMF")
// Called when users presses a button
// ( any of Inband, SIP , RFC2833)
public void doDtmf(String digit) throws Exception{

    // log to the logger
    log.info("User pressed " + digit);

    // play announcement to user
    ivrHelper.playAnnouncementWithDtmf(
        "welcome.wav");
}
```

# JRuby Torquebox Telco Framework

Based on JBoss AS 5 Torquebox

Uses Sip Servlets 1.1 API

Supports Media API (JSR-309)

Supports Ruby on Rails

Supports hot-deployment

100% pure JRuby Telco Applications

# JRuby Sample Code

```
# Note that the class extend a JBoss provided sip
# controller called JBoss::Sip::SipBaseHandler
# that mimic the Java Sip Servlet class the ruby way
class SipHandler < JBoss::Sip::SipBaseHandler
  # Handle INVITE request to setup a call by answering 200 OK
  def do_invite(request)
    request.create_response(200).send
  end
  # Handle BYE request to tear down a call by answering 200 OK
  def do_bye(request)
    request.create_response(200).send
  end
  # Handle REGISTER
  def do_register(request)
    request.create_response(200).send
  end
  # Handle a successful response
  def do_success_response(response)
    response.create_ack.send
  end
end
```

# Other frameworks useful in Telco Applications

JBoss Seam

JRuby (for AS 5)

GWT

Richfaces and A4J

JBoss Cache/Infinispan

JBoss Rules

JBPM

RESTeasy

Many others

# Development Tools

## JBoss Tools and JBoss Developer Studio

Reuse the Seam plugins for STF projects

<http://www.jboss.org/tools/>

<http://www.jboss.com/products/devstudio/>

## Mobicents Eclipse Sip Servlets Plug-ins

Core plugin — project creation and deployment of JSR-289 projects

SIP Phone plugin — for testing SIP and Media applications with real phone from the IDE

STF Plugin – creating STF projects

Java - MSS JBoss 5.0 Server - Eclipse SDK

File Edit Source Navigate Search Project Run Window Help

Package Expl Hierarchy MSS JBoss 5.0 Server

redhat mobicents sip servlets management console

Router Configurat Server Settings Deploy Applications

Default Application Router Configuration View source Save Log and errors Help

ALL INVITE REGISTER SUBSCRIBE OPTIONS MESSAGE

Application Name: org.mobicents.servlet.sip.exar

Options

Delete

Add application

Task List Find: All Uncategorized

Outline An outline is not available.

Overview Sip Servlets Management

Problems Javadoc Declaration Servers Console

MSS JBoss 5.0 Server [JBoss Application Server Startup Configuration] C:\Program Files\Java\jre6\bin\javaw.exe (Jun 12, 2009 4:50:52 PM)

To: <sip:test@127.0.0.1>  
 Via: SIP/2.0/UDP 127.0.0.1:7287;branch=z9hG4bKd59562e50f4f74032cc8a295d565d279  
 Max-Forwards: 70  
 User-Agent: Sip communicator1Windows XP  
 Expires: 3600  
 Contact: "display" <sip:test@127.0.0.1:7287;transport=udp;registering\_acc=127\_0\_0\_1>;expires=3600  
 Content-Length: 0

16:53:42,093 INFO [SipApplicationDispatcherImpl] sip session (sip:test@127.0.0.1,7f7106,sip:test@127.0.0.1,k  
 16:53:42,093 INFO [SipApplicationDispatcherImpl] session (sip:test@127.0.0.1,7f7106,sip:test@127.0.0.1,bdbce  
 16:53:42,093 INFO [SipApplicationDispatcherImpl] Sip session (sip:test@127.0.0.1,7f7106,sip:test@127.0.0.1,k

SIP Phone

Server: 127.0.0.1  
 Server Port: 5080  
 Local SIP Port: 5060  
 Dial URI: sip:1010@127.0.0.1:5080  
 Unregister Call  
 Ended

[Overview] One or more ports have an invalid value. This may adversely affect the tooling's ability to interact with the server.

4:53 PM

# Time for Demos

IPBX – STF PBX Application with Richfaces UI

Torquebox Telco Framework

Conferencing with server push

IDE Tools

Code

Media Server

Management, Debugging and Analysis Tools

**QUESTIONS?**

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