Web 2.0 Phone Home: Rapid Development of Telecom-Enabled Web Applications

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Talk Outline

> Whirlwind introduction to IP telephony and converged applications
> Tools for creating converged applications
  • E4SS
  • Grails
  • Converge
> Tools for testing converged applications (KitCAT)
> Increased productivity through application composition
Motivation
Urgent assignment from the boss (spouse?)

Need an application to record a message and deliver it via phone calls to a list of people
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The System Architecture

- Media Server
- SIP/HTTP App Server
  - App
  - App
  - App
- DB
- Browser
- HTTP
- SIP
The System Architecture: SIP

- SIP/HTTP App Server
  - Media Server
  - DB
  - Browser
  - App

- SIP connections:
  - SIP/HTTP App Server to Media Server
  - SIP/HTTP App Server to DB
  - SIP/HTTP App Server to Browser

- HTTP connection:
  - SIP/HTTP App Server to App
Introduction to IP Telephony

Major standards

> **SIP**
  - Session Initiation Protocol
  - used for *signaling*

> **SDP**
  - Session Description Protocol
  - used for *media description*

> **RTP**
  - Real-Time Protocol
  - used for *media transport*
Introduction to IP Telephony
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SIP basics

> Text-based protocol
> Peer-to-peer
> Request methods include \texttt{INVITE}, \texttt{ACK}, \texttt{BYE}, etc.
> Requests and responses can contain media descriptions in SDP

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<th>UAS</th>
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<td>(7) 200/BYE</td>
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INVITE sip:bob@biloxi.com SIP/2.0  
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bK776asdhds  
Max-Forwards: 70  
To: Bob <sip:bob@biloxi.com>  
From: Alice <sip:alice@atlanta.com>;tag=1928301774  
Call-ID: a84b4c76e66710@pc33.atlanta.com  
CSeq: 314159 INVITE  
Contact: <sip:alice@pc33.atlanta.com>  
Content-Type: application/sdp  
Content-Length: 142  

(SDP not shown)
SIP basics

Text-based protocol

Peer-to-peer

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The System Architecture: App Server and Apps
Introduction to SIP Servlets

- First standardized in JSR 116, updated in JSR 289.
- Extension of GenericServlet model, offering application-specific callbacks such as doInvite and doResponse.
- Supports session-oriented storage.
- SipApplicationSession can contain multiple protocol sessions (e.g., SipSession and HttpSession).
Introduction to Converged Applications

> JSR 289 supports applications that include SIP Servlets plus:
  • HTTP Servlets
  • Java EE components (e.g., EJBs)

> Permits use cases such as:
  • SIP calls initiated via HTTP
  • EJB updated to reflect SIP call state
  • etc.
Open-source Application Servers

> **Sailfin**
  - adds SIP Servlet capabilities to Glassfish
  - supports clustering, converged load balancing
  - Sailfin 1.0 Final Release - 23 Jan 2009

> **Mobicents SIP Servlets**
  - adds SIP Servlet capabilities to Tomcat and JBoss
  - supports load balancing, clustering, failover
  - Mobicents SIP Servlets 0.9 - 21 Apr 2009
The System Architecture: Apps

- Media Server
- SIP/HTTP App Server
- App
- App
- App
- Browser
- DB

Connections:
- SIP
- HTTP
- SIP
- SIP
- SIP
- SIP
Application Functions

> Record
  • Web: Initiate recording
  • Telecom: Set up call between subscriber and audio recorder

> Blast
  • Web: Initiate “blasting” recipients
  • Telecom: Set up calls between recipients and audio player
Record Message

Media Server

App

Browser
Record Message

Media Server

App

Browser
Record Message

Media Server

App

Browser
Record Message

Media Server

App

Browser
Blast Message

Media Server

App

Browser
Blast Message
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Media Server

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Media Server

App

Browser
Blast Message

Media Server

App

Browser
Open Source Tools for Rapid Development of Converged Apps

> ECharts for SIP Servlets (E4SS)
  • SIP servlet application development

> Grails
  • web application development

> Converge
  • converged application development

> KitCAT
  • converged application testing
E4SS (ECharts for SIP Servlets)

> Call processing logic expressed as state machines with ECharts programming language

> Includes catalog of reusable state machine fragments

> Provides catalog of complete, reusable telecom components distributed as jar files

> (Demonstration)
Grails

> Java-savvy Rails-like framework
> Builds on Spring and Hibernate
> Uses Groovy scripting language
> Plugin architecture with many third-party plugins available
Converge

- Two components
  - Application development framework
  - Core runtime applications
- Framework combines E4SS and Grails
  - Grails used to develop web and DB side of app
  - E4SS used to develop telecom side of app
- Core applications include an application router, a SIP registrar and an administrative control panel
Building the App

> Use Converge to generate a Grails plugin based on the E4SS reusable Click2DialFlow1 feature
> Use Grails to create an app that uses the plugin
> Flesh out the skeleton
> (Demonstration)
The System Architecture: Media Server

SIP/HTTP App Server

Media Server

DB

SIP

HTTP

Browser
Open-source SIP media servers

> Asterisk
> SIP Express Media Server
> Asterisk + OpenVXI + VoiceGlue
> sipXecs
> jVoiceBridge
Testing the App

> Grails supports unit/integration testing for the web side

> KitCAT supports functional testing for both the web and telecom side
KitCAT - Functional testing of converged applications

- Java-based testing framework supporting SIP, RTP, and HTTP (*via HtmlUnit*)
- Allows creation of JUnit-style test cases for converged applications
- High-level call control primitives to control test agents (e.g., call, answer, end) with options for low-level customization
- Send and receive RTP, send touchtones
- Flexible and extensible assertion primitives
Extending the example

> Review message before blasting
> Obtain blast numbers from an address book
> Blast at a future time
> View blast status for each number (successful/unsuccessful)
E4SS Convergence Framework

> Reusable E4SS features provide SIP-to-Java and Java-to-SIP interfaces to support interaction with non-SIP environment
Converge: E4SS/Grails Interfacing

> Converge generates Grails services to interface with an E4SS feature’s SIP-to-Java and Java-to-SIP interfaces.
Blast Status

> Create BlastStatus domain class and use Click2DialFlow1MachineToJavaService to update its state
> Grails scaffolding provides views
> (Demonstration)
Application Composition

> Unlike HTTP servlets, SIP servlets can be composed at runtime to create complex telecom services
> Accomplished using a JSR 289 application router
> Converge includes the E4SS DFC app router
For more information

> You don’t have to be the phone company to incorporate telecom into your apps

> SIP gateway services (DID inbound origination and outbound termination)
  • see voip-info.org

> Converge, E4SS, KitCAT, ECharts and lots of documentation
  • see echarts.org
Speakers: Greg Bond & Tom Smith
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