Java

JavaOne

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

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at&t

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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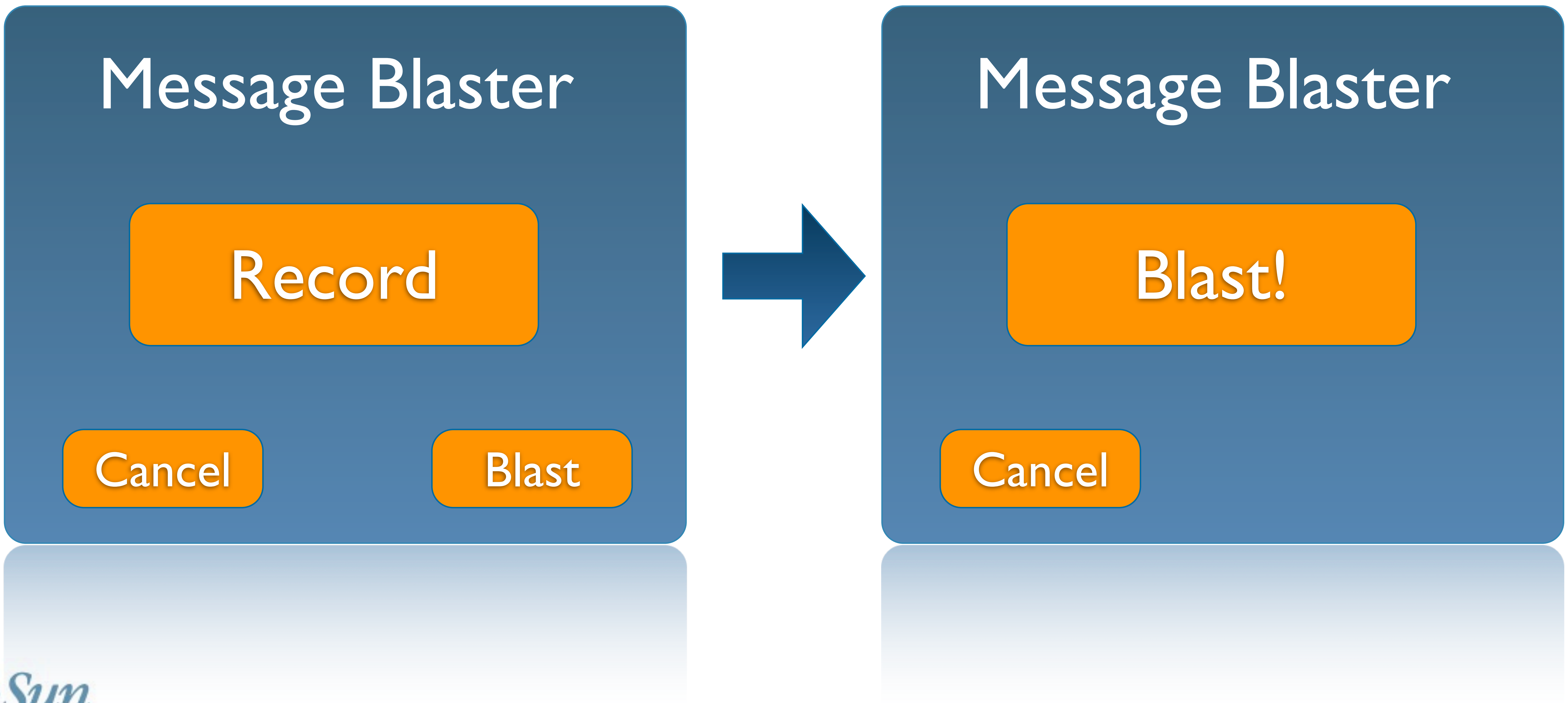
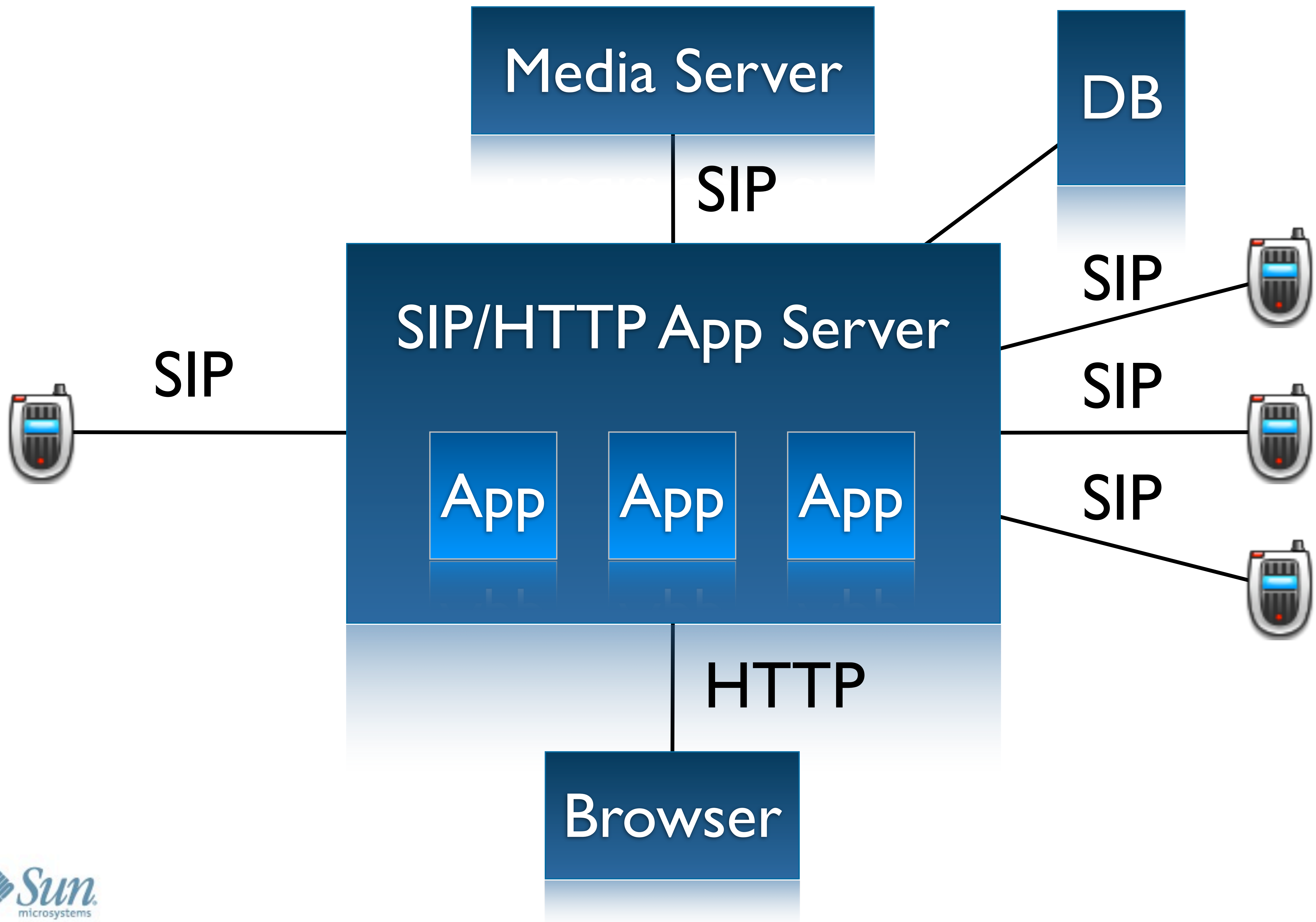
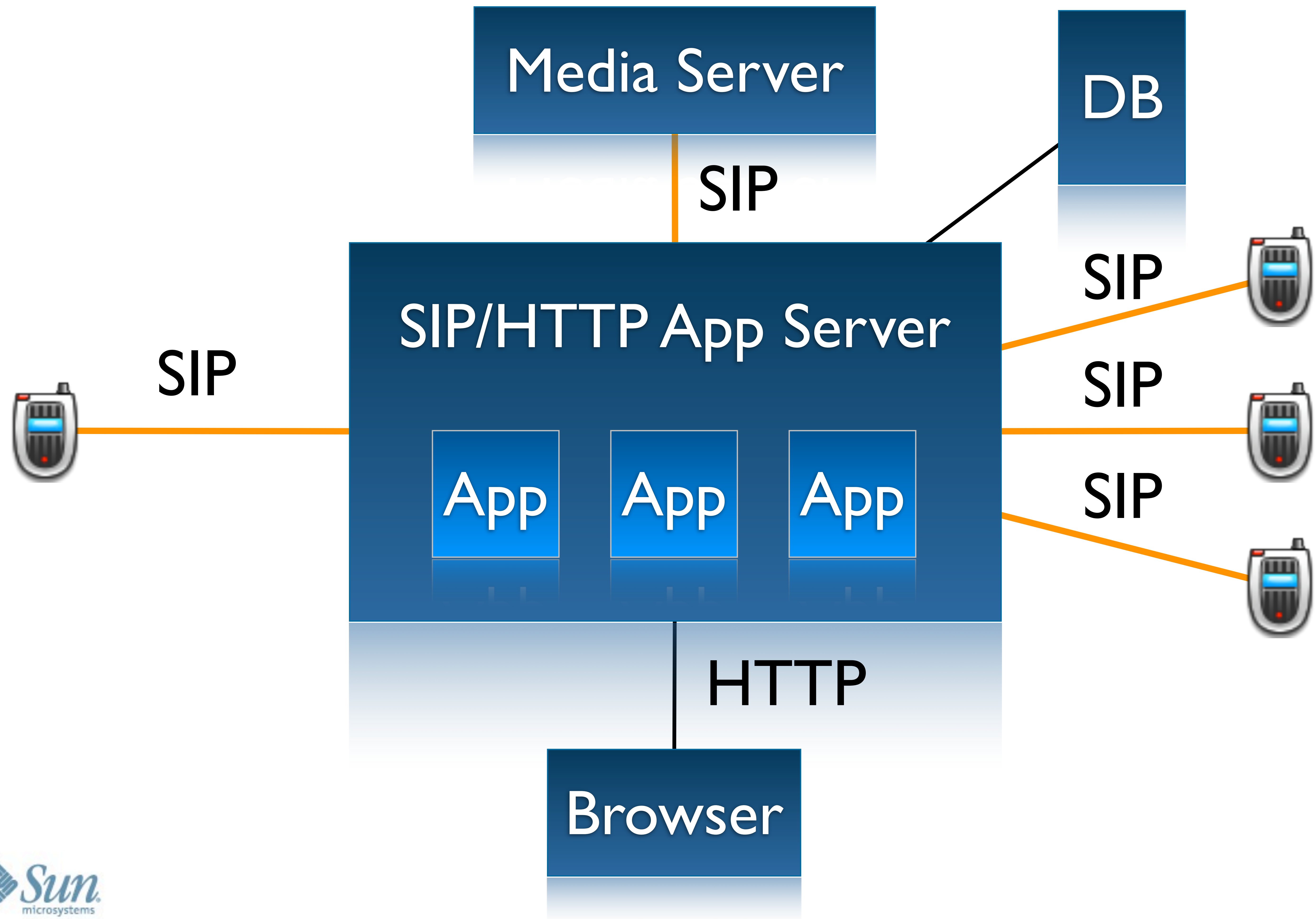


The System Architecture



The System Architecture: SIP



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*

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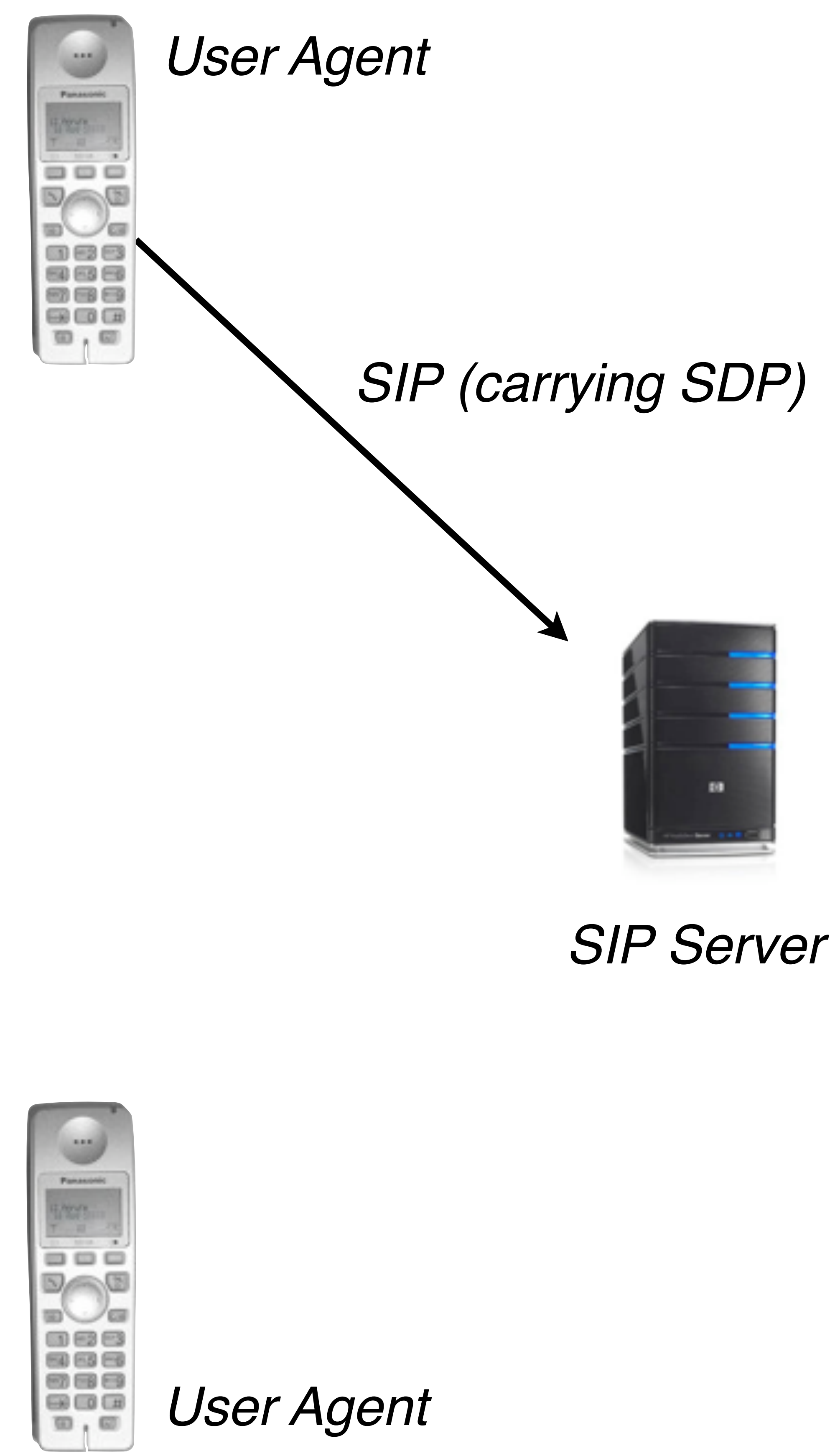

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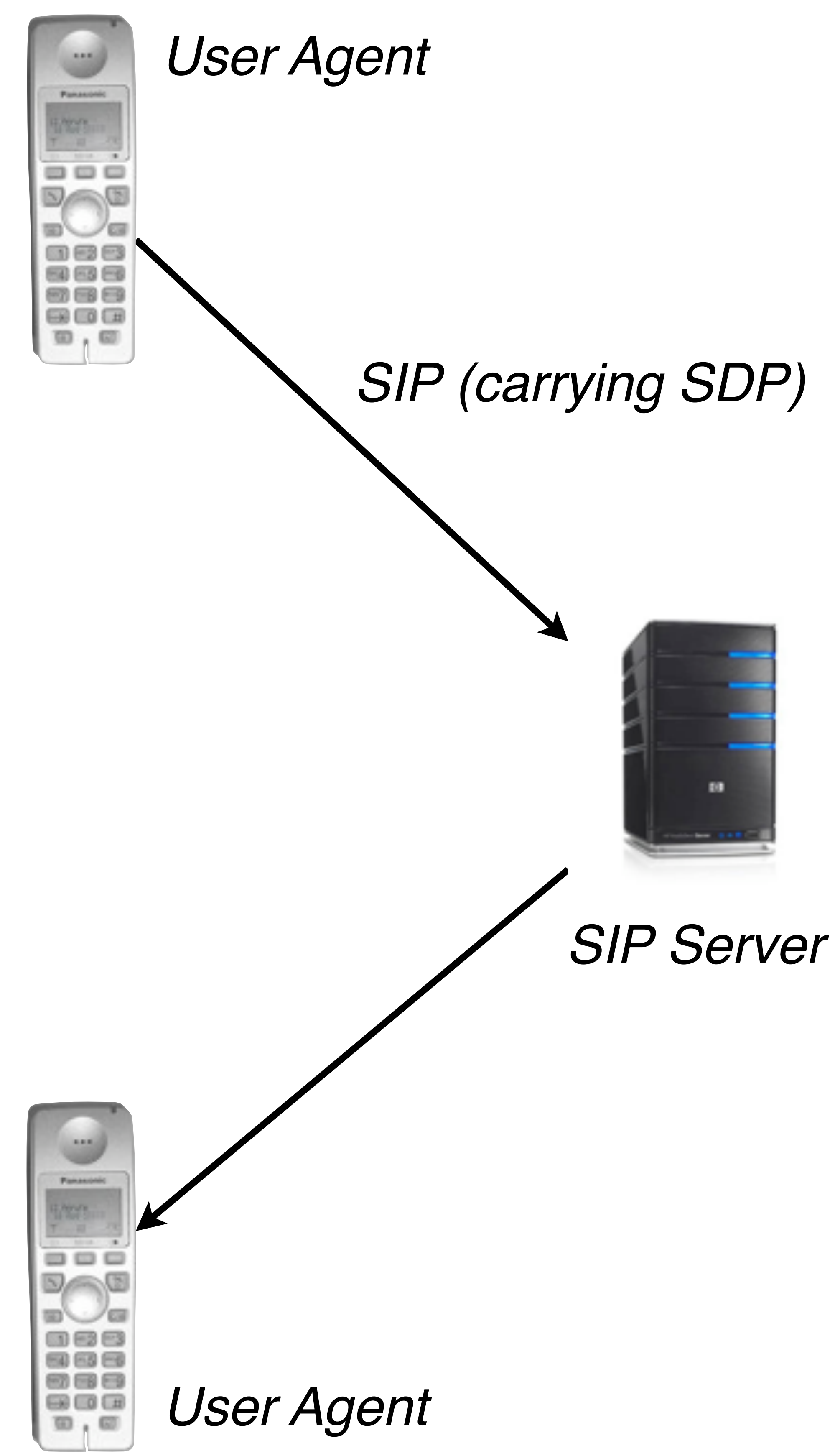

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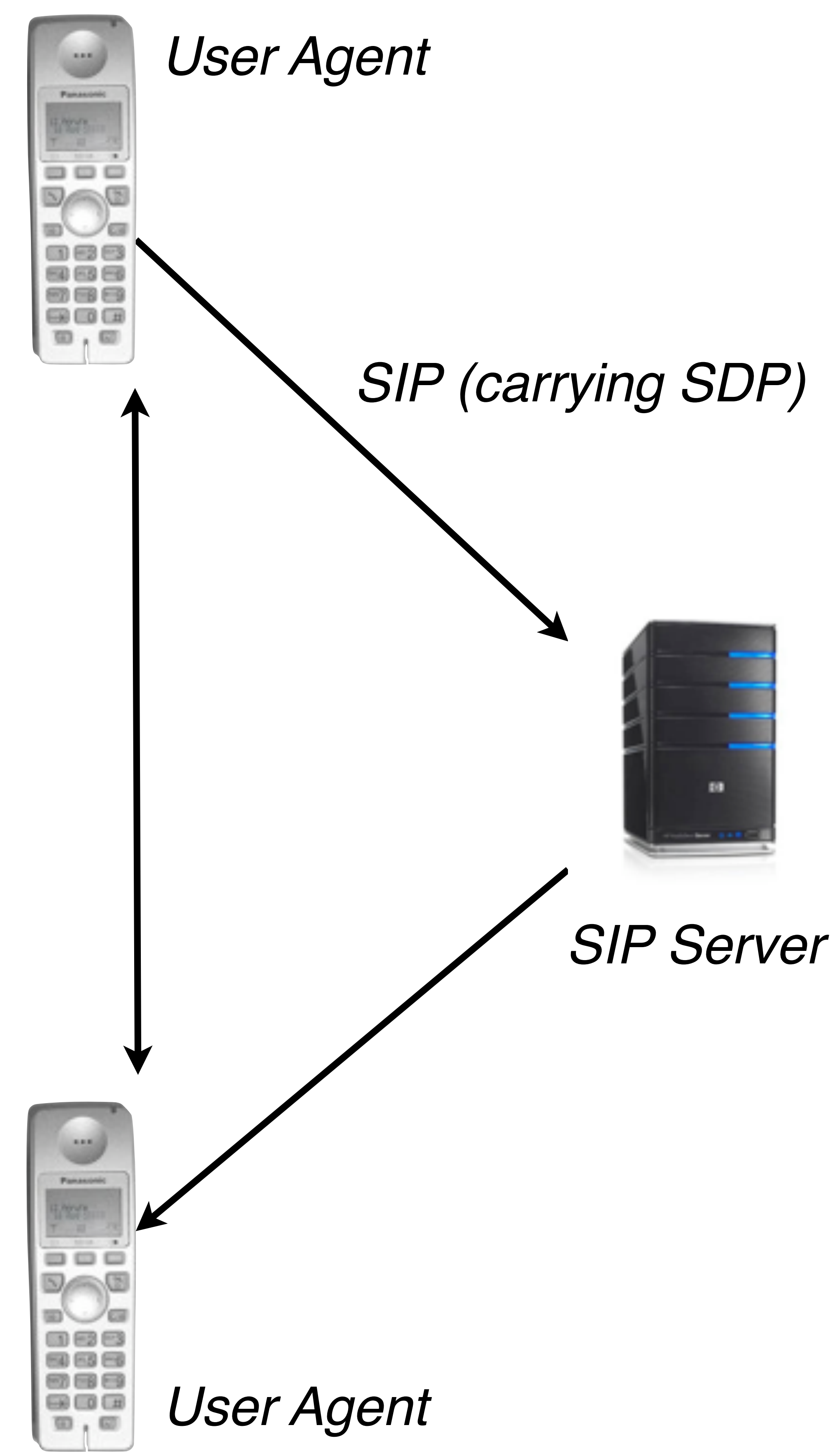

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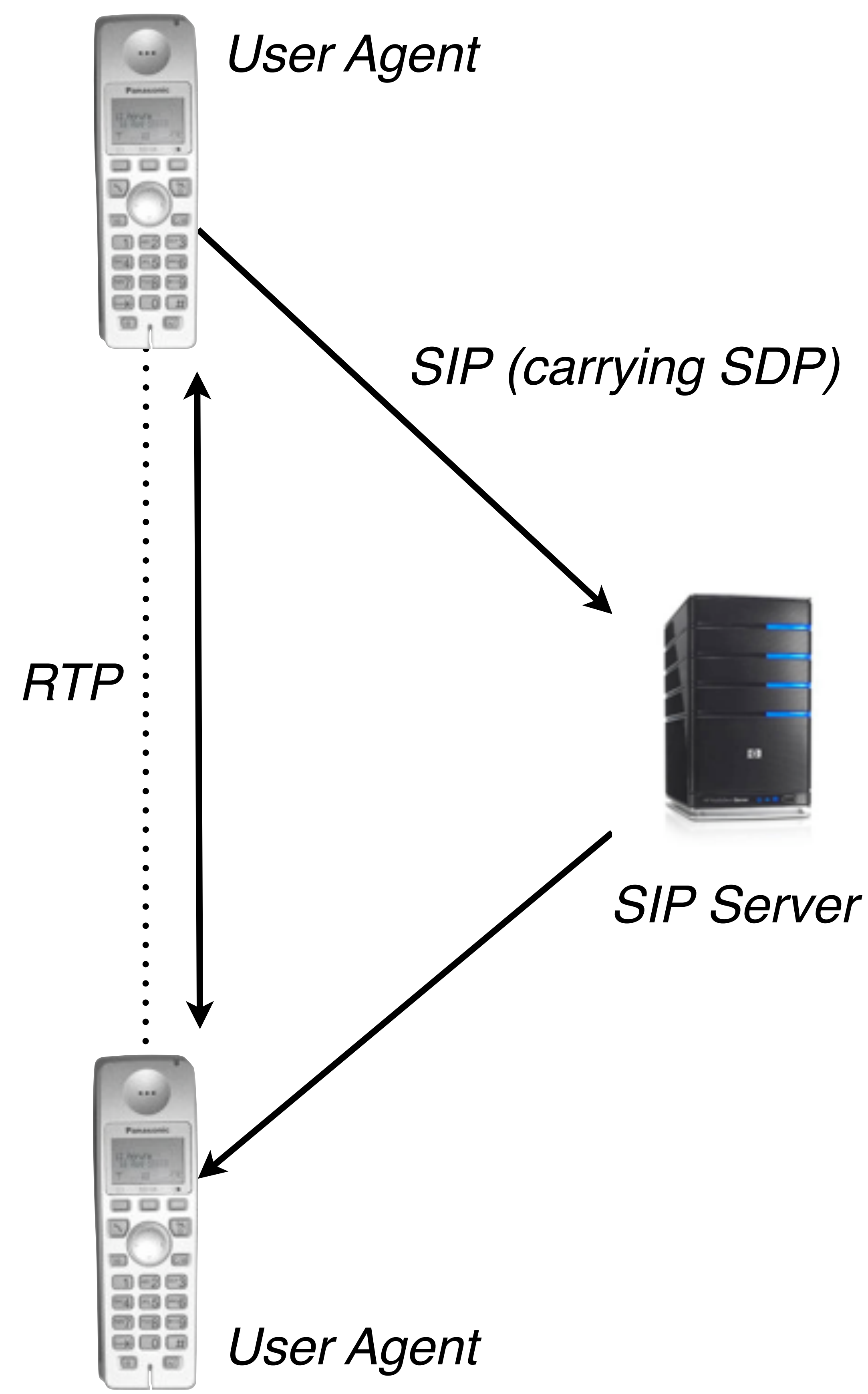

SIP basics

- Text-based protocol
- Peer-to-peer
- Request methods include **`INVITE`**, **`ACK`**, **`BYE`**, etc.
- Requests and responses can contain media descriptions in SDP

```
UAC                  UAS
   | (1)  INVITE       |
   |---------------->|
   | (2)  180/INVITE |
   |<----------------|
   | (3)  200/INVITE |
   |<----------------|
   | (4)  ACK          |
   |---------------->|
   | (5)  RTP          |
   |.................|
   | (6)  BYE          |
   |<----------------|
   | (7)  200/BYE      |
   |---------------->|
```

```
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)
```

include **INVITE**, **ACK**, **BYE**, etc.

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```
                    UAS
  NVITE              |
  ----------->|
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  -----------|
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|<-------------------|
|(4) ACK             |
|------------------->|
|(5) RTP             |
|...................|
|(6) BYE             |
|<-------------------|
|(7) 200/BYE         |
|------------------->|
```

```
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/2.0 200 OK
Via: SIP/2.0/UDP server10.biloxi.com
   ;branch=z9hG4bKnashds8;received=192.0.2.3
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710@pc33.atlanta.com
CSeq: 314159 INVITE
Contact: <sip:bob@192.0.2.4>
Content-Type: application/sdp
Content-Length: 131

(SDP not shown)
```

```
                  UAS
 NVITE             |
 ------------->    |
 30/INVITE         |
 -------------     |
| (3) 200/INVITE   |
 -------------     |
 CK                |
 ------------->    |
 TP                |
 .............     |
 YE                |
 -------------     |
 00/BYE            |
|---------------->|
```

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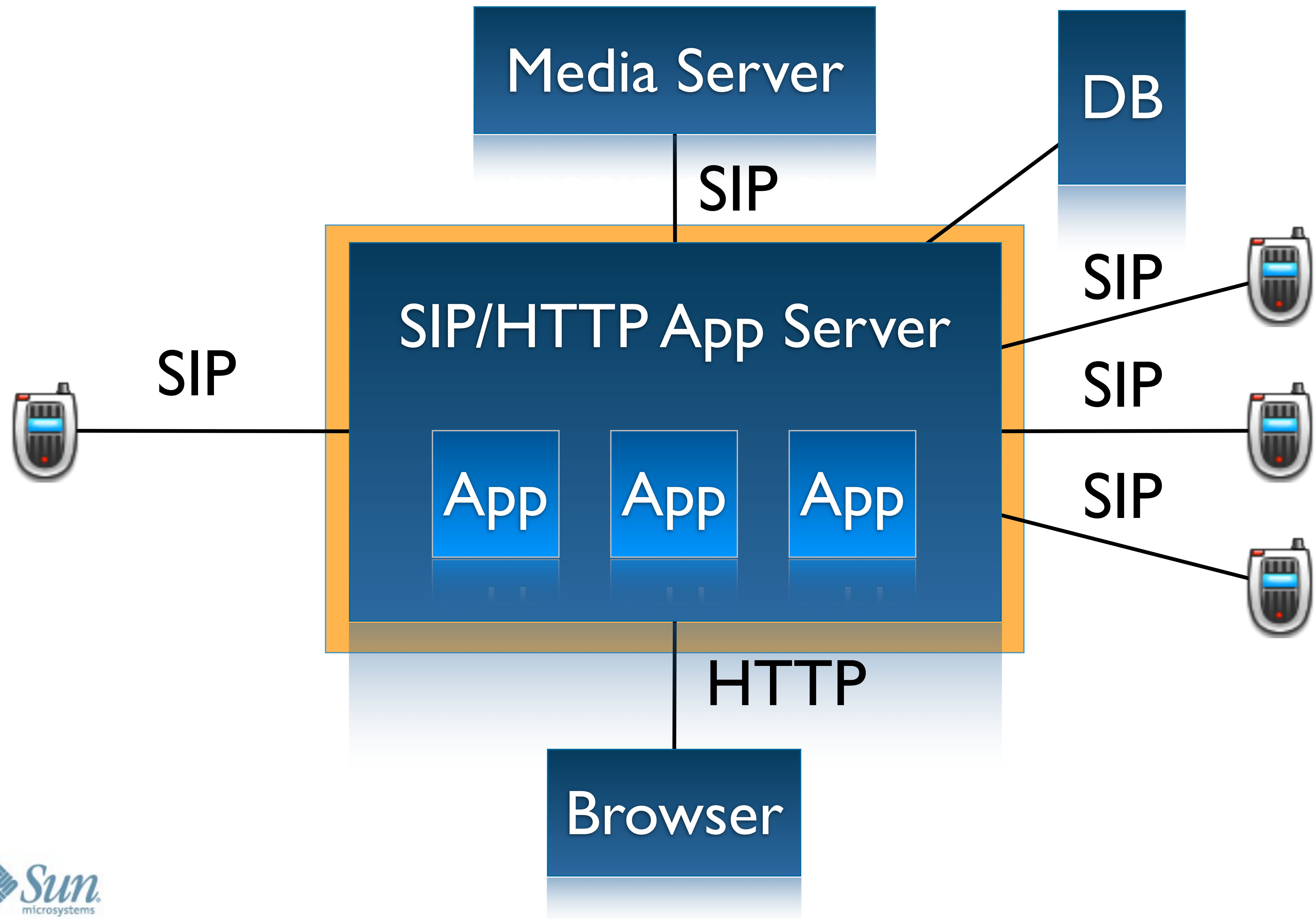
   |------------------>|
   |(5) RTP            |
   |...................|
   |(6) BYE            |
   |<------------------|
   |(7) 200/BYE        |
   |------------------>|
```

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 Servlet plus:
- HTTP Servlets
- Java EE components (e.g., EJBs)

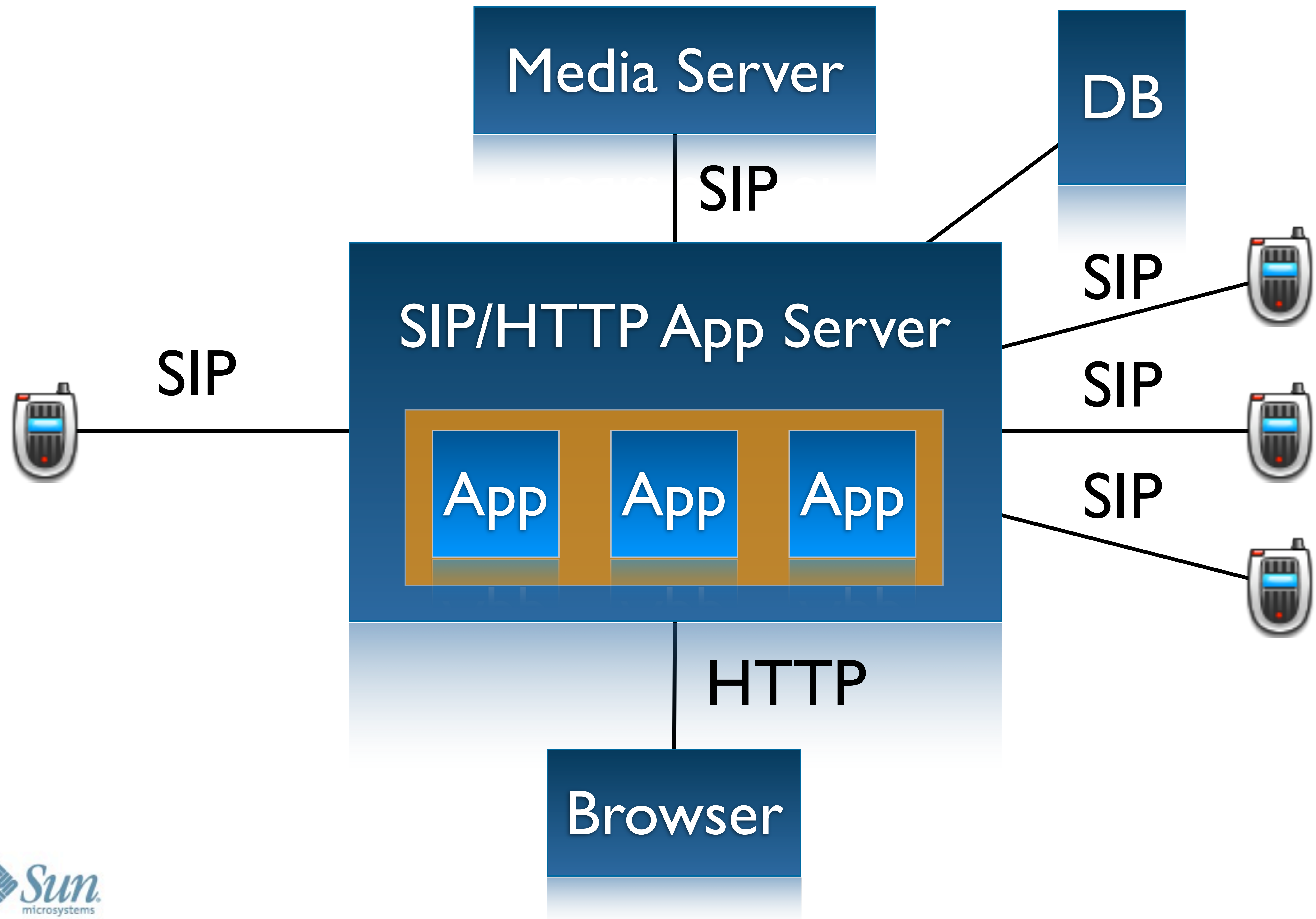
> Permits use cases such as:
- SIP calls initiated via HTTP
- EJB updated to reflect SIP call state
- etc.

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

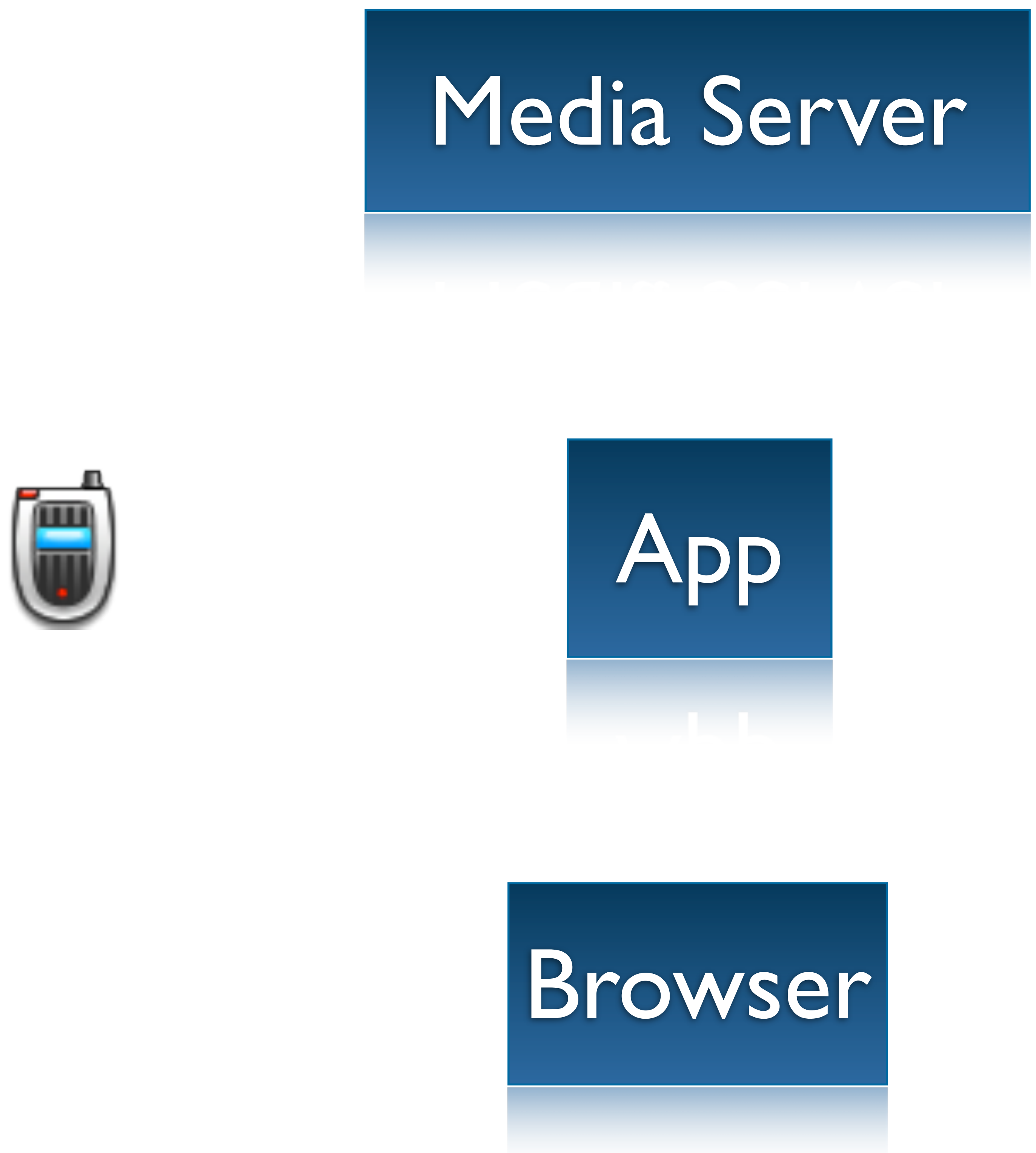


Application Functions

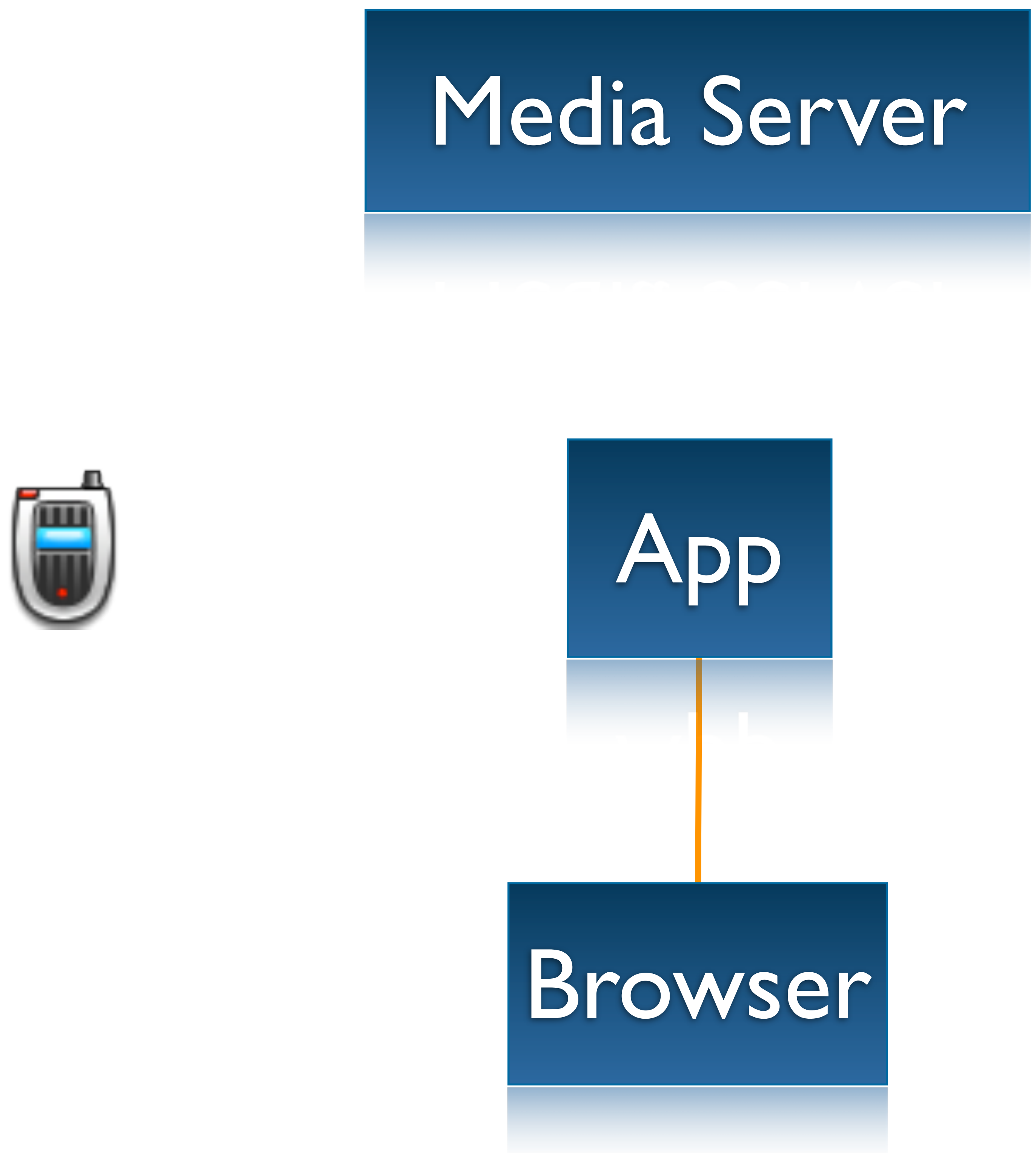
> Record
- Web: Initiate recording
- Telecom: Set up call between subscriber and audio recorder

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

Record Message



Record Message



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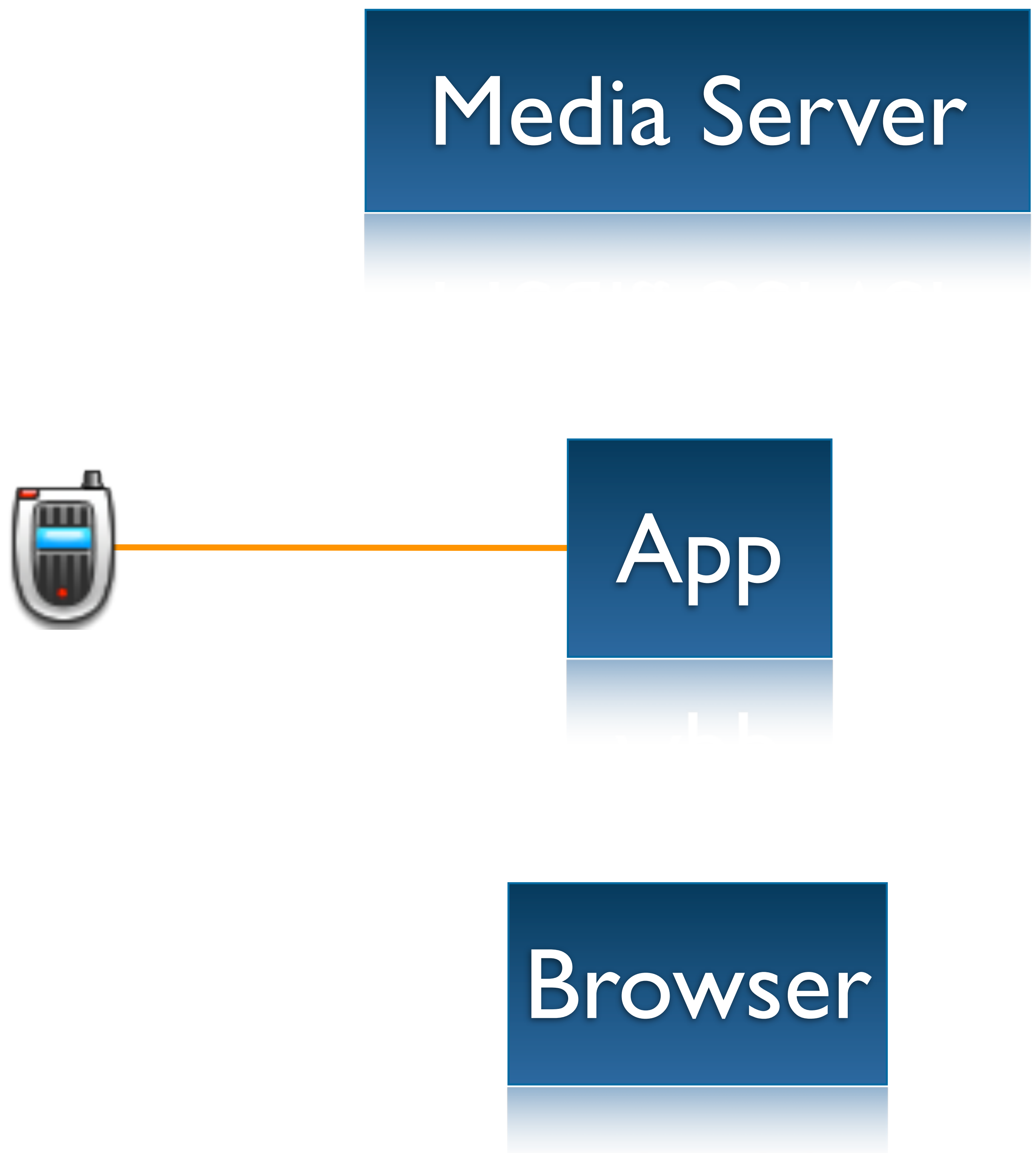
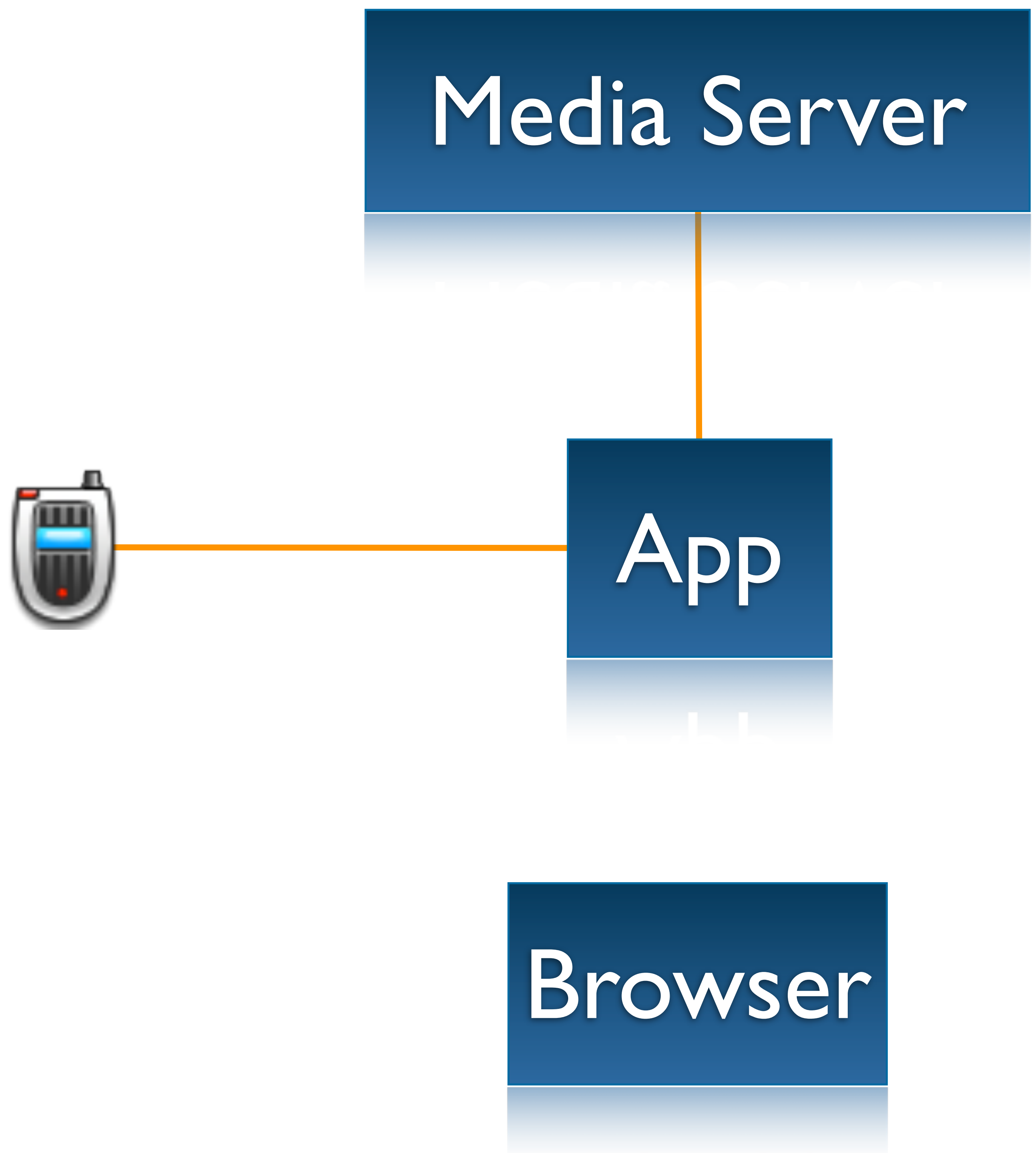
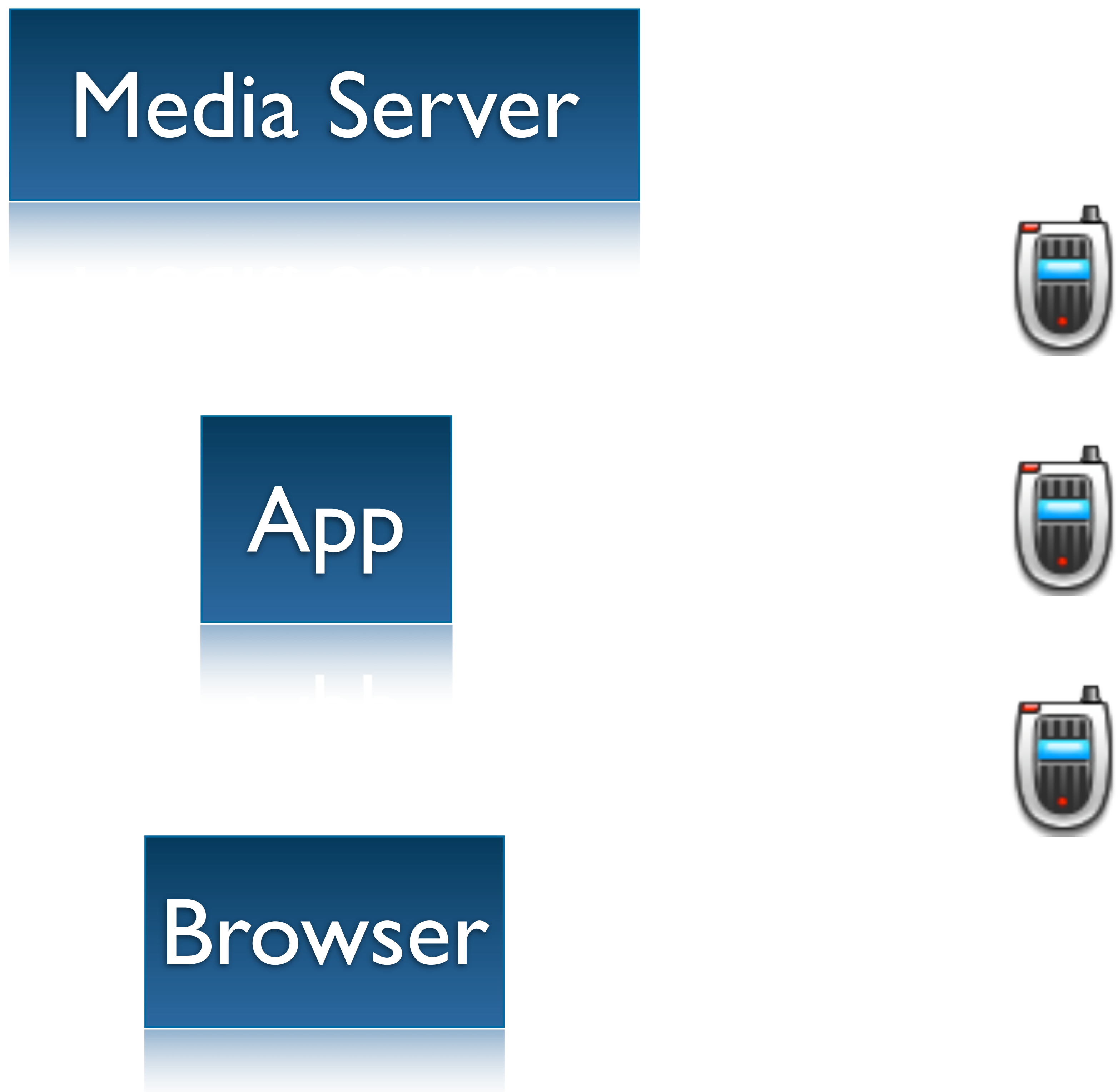


Thursday, June 4, 2009

Record Message



Blast Message



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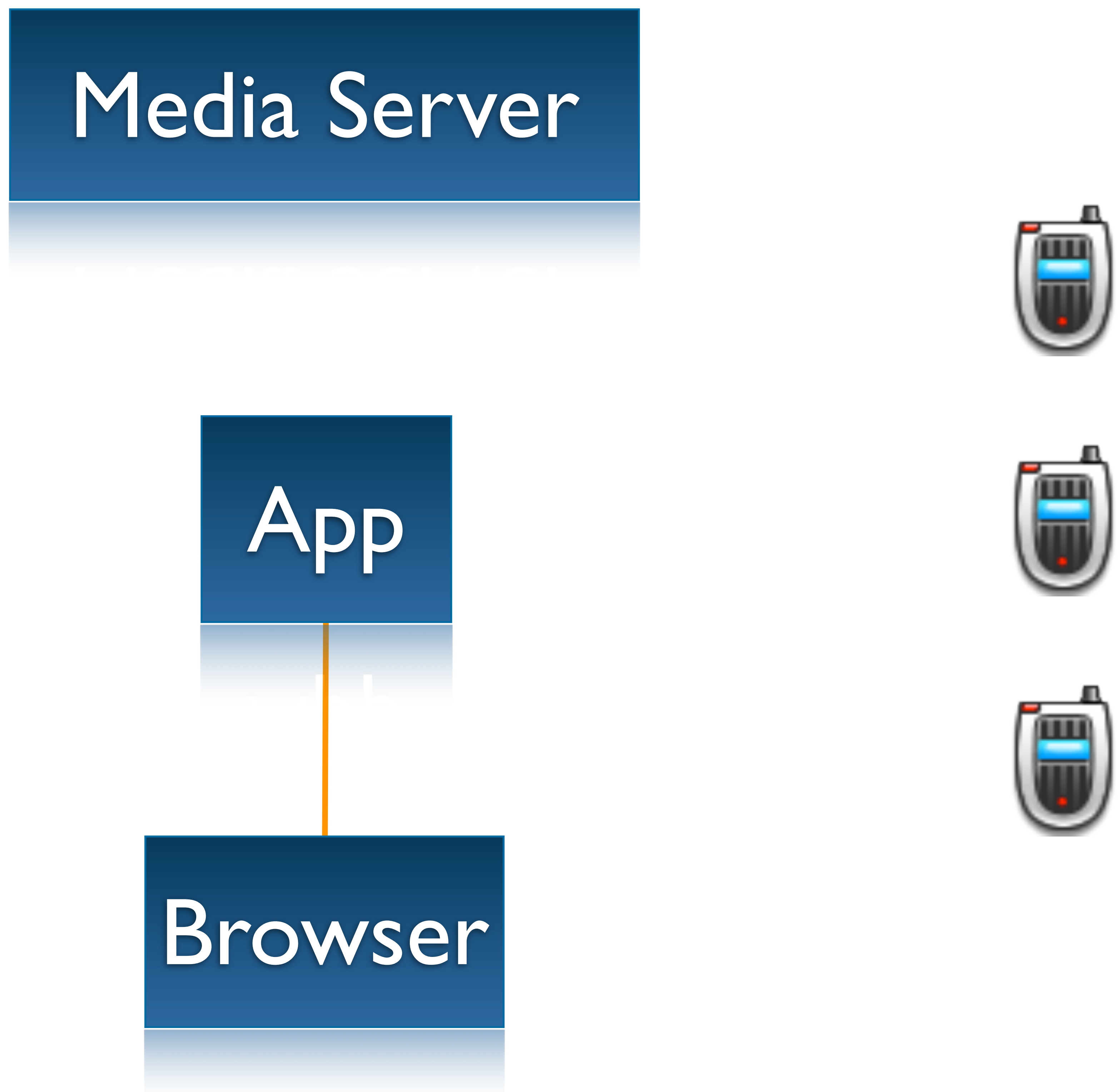

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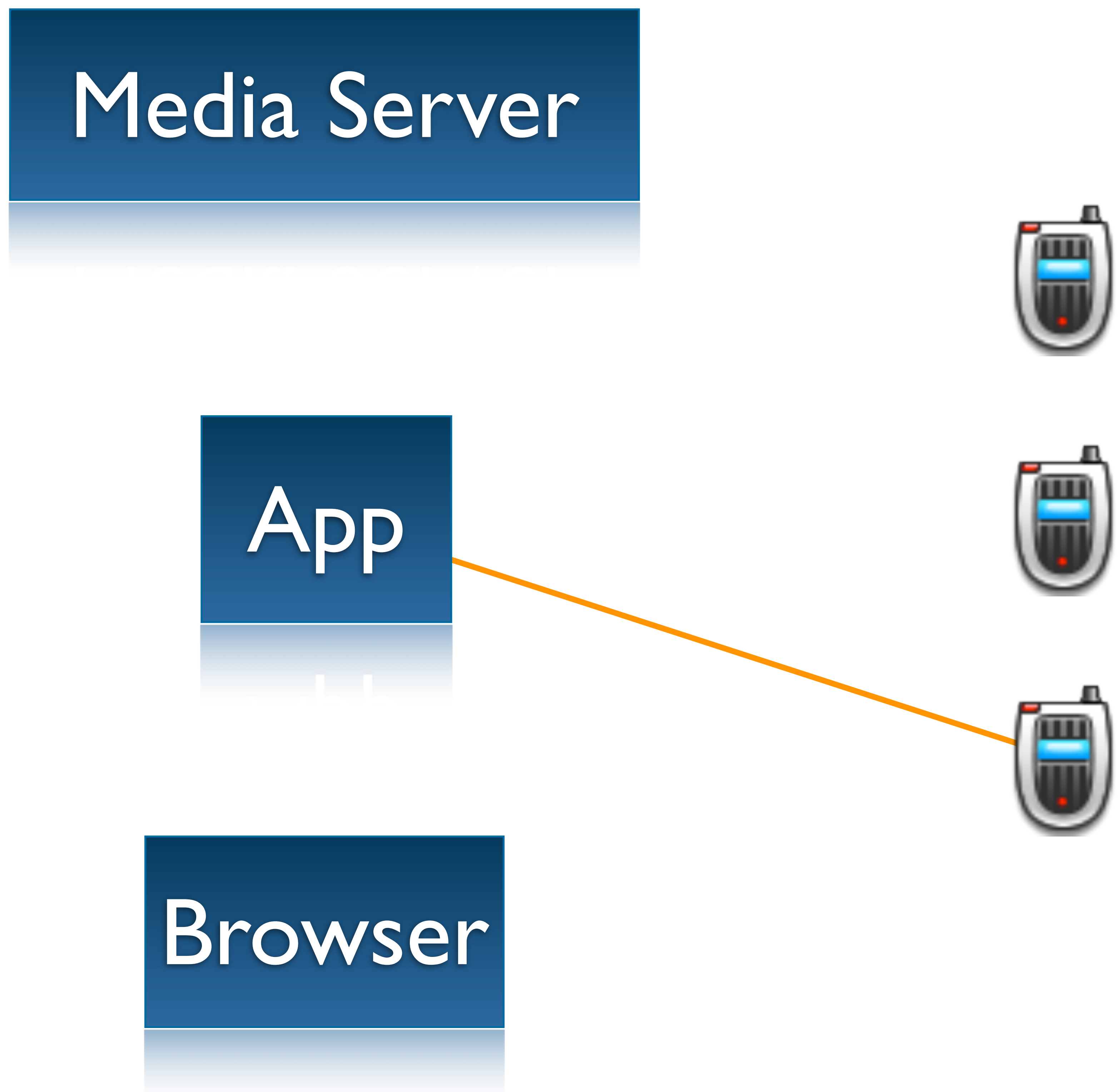

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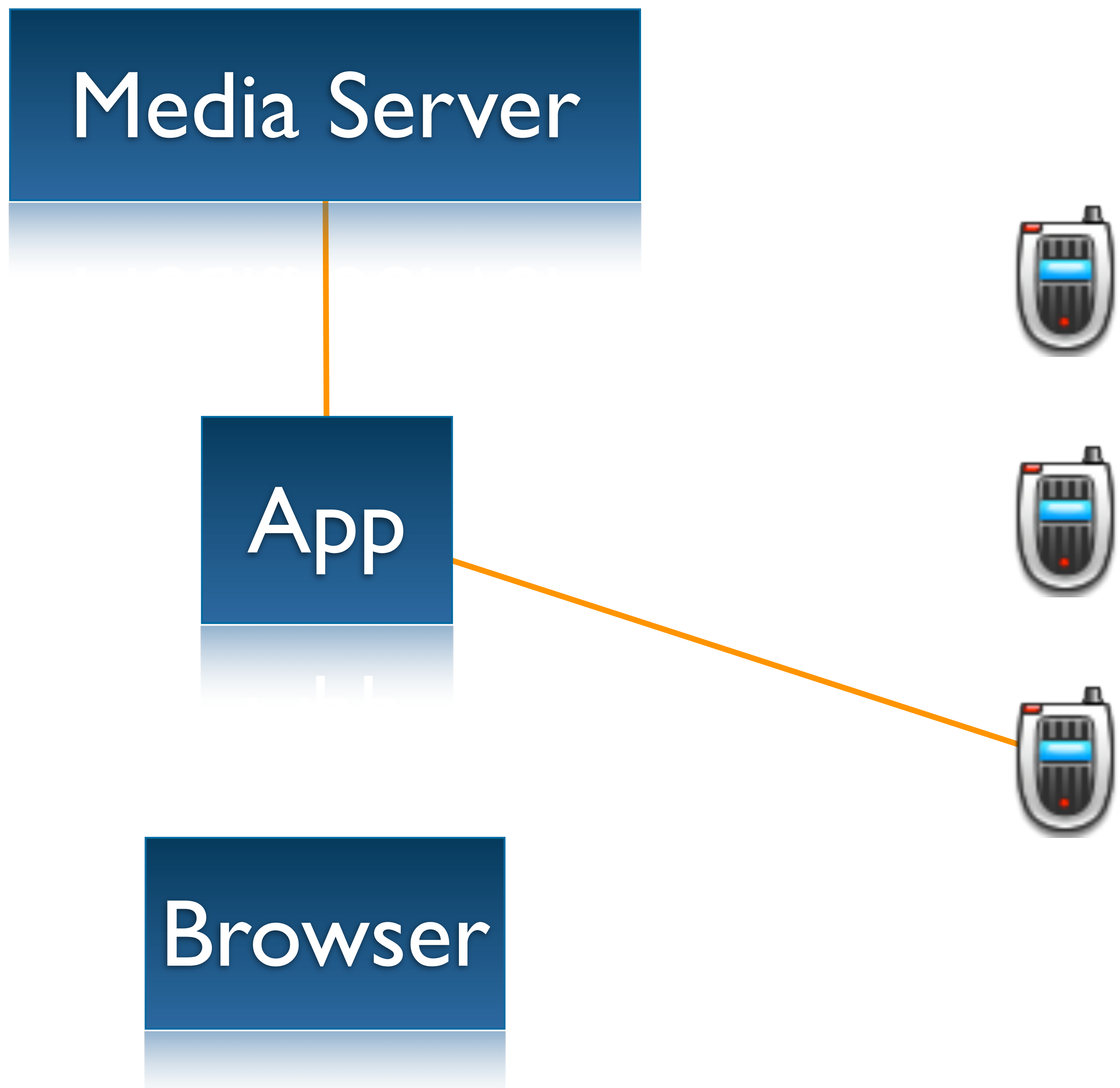

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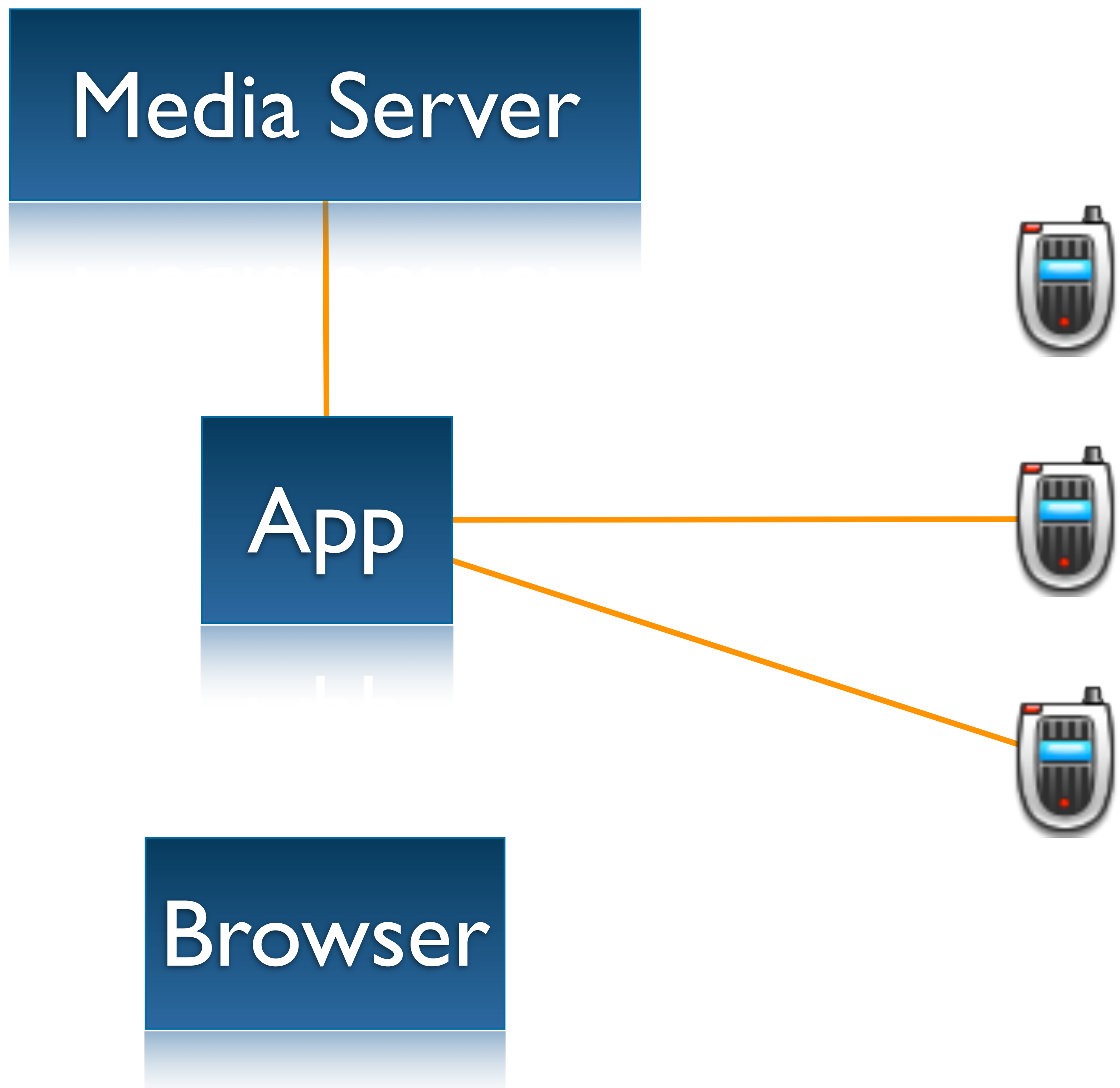

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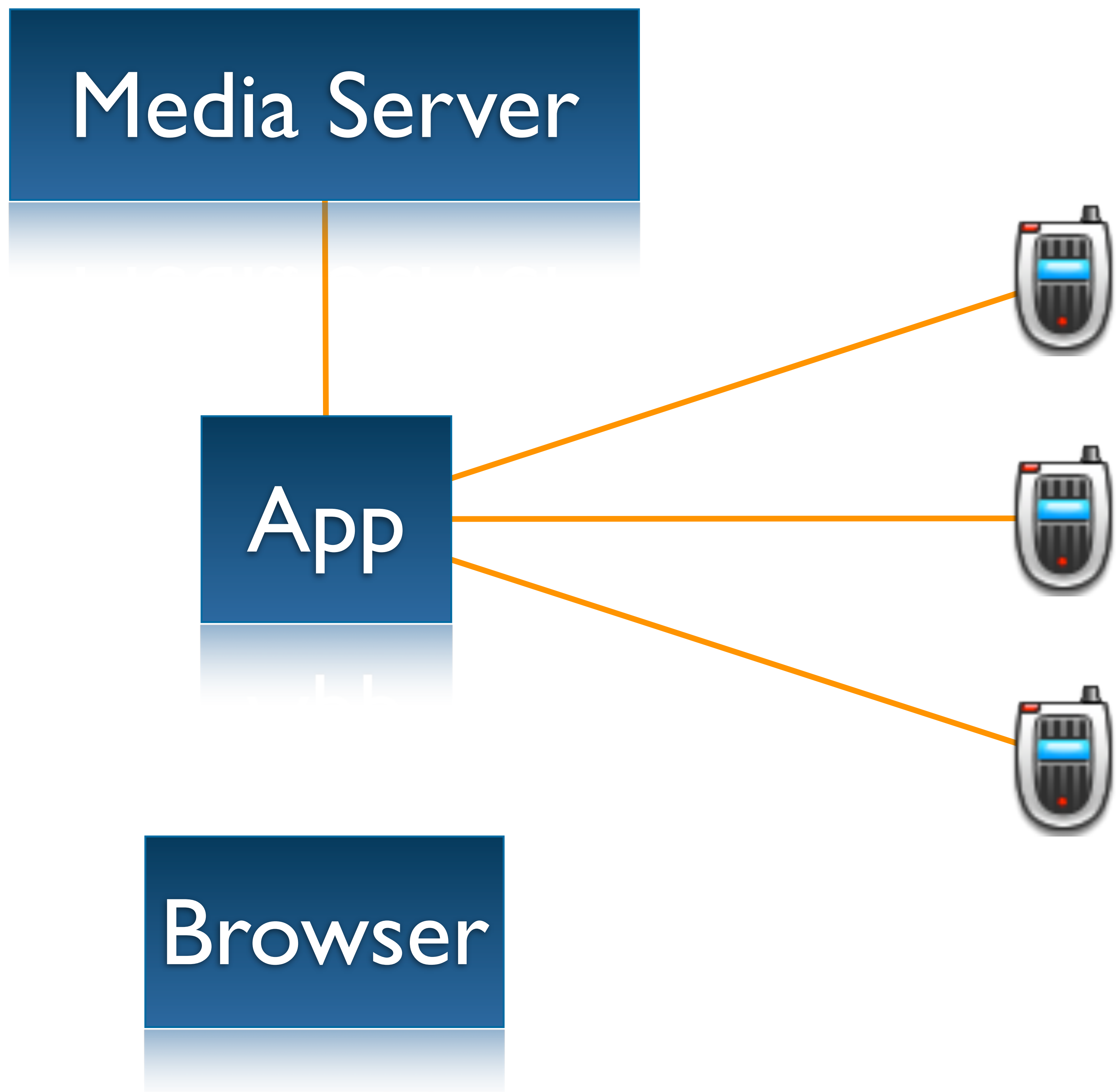


Open Source Tools for Rapid Development of Converged Apps

> 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)

ECharts
machine code

- 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

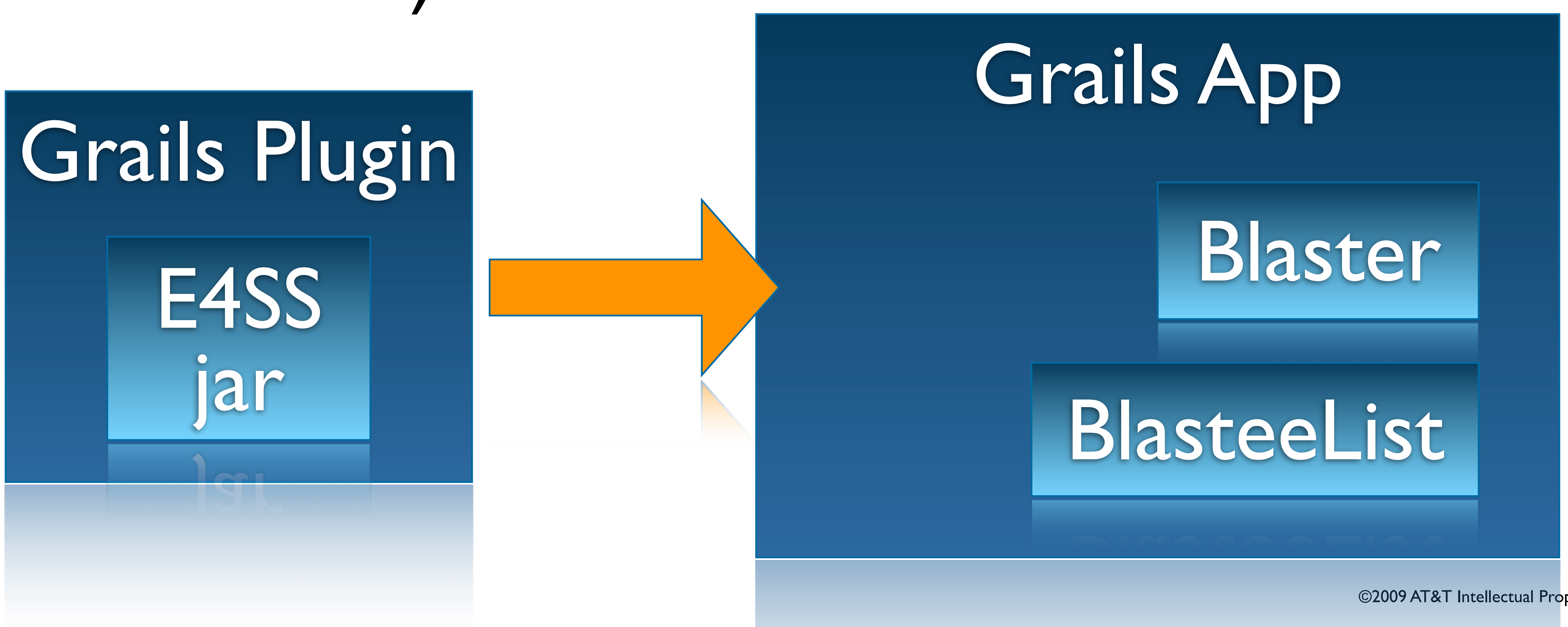
> Two components
  - Application development framework
  - Core runtime applications

> Framework combines E4SS and Grails
  - Grails used to develop web and DB side of app
  - E4SS used to develop telecom side of app

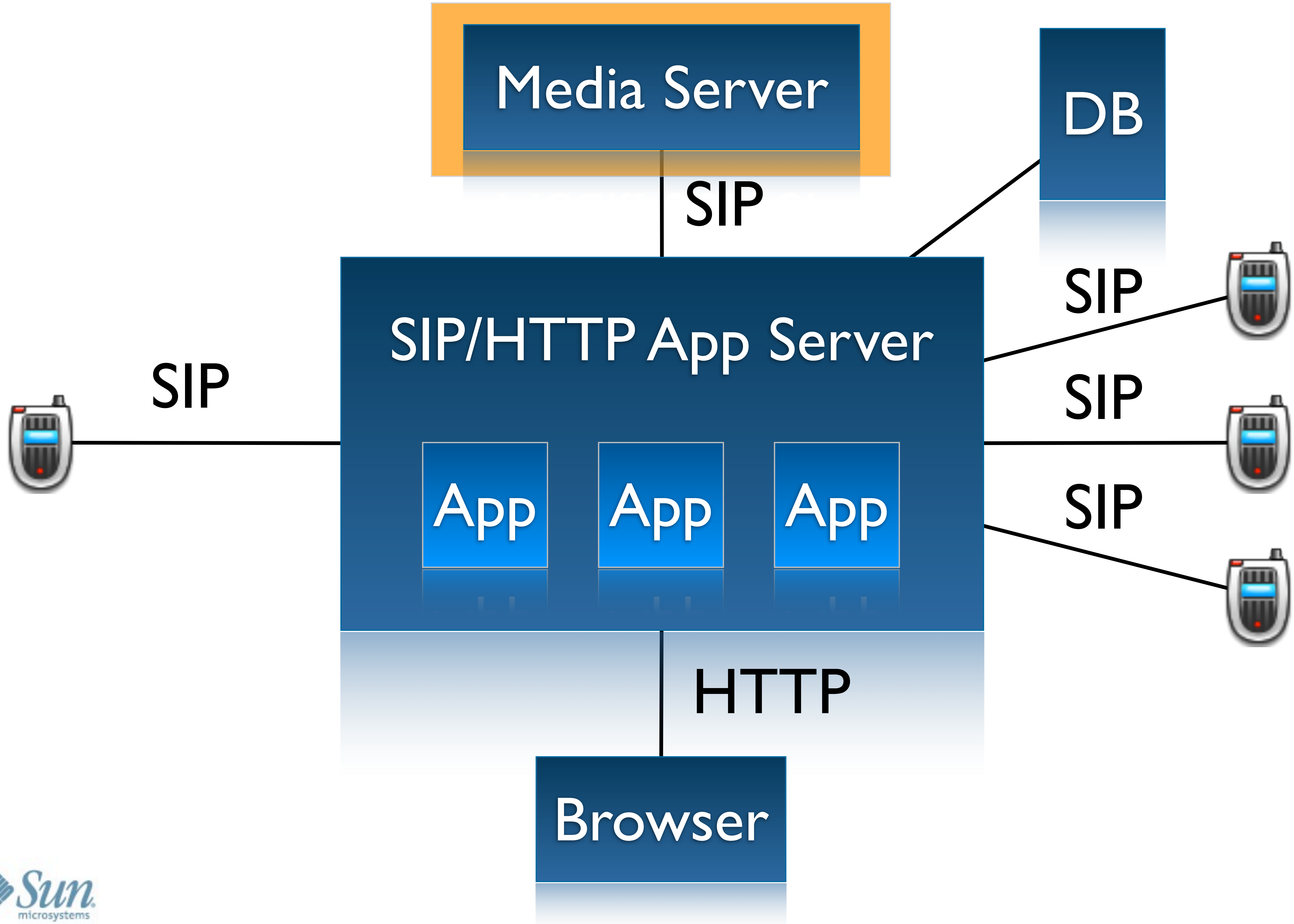
> Core applications include an application router, a SIP registrar and an administrative control panel

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



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

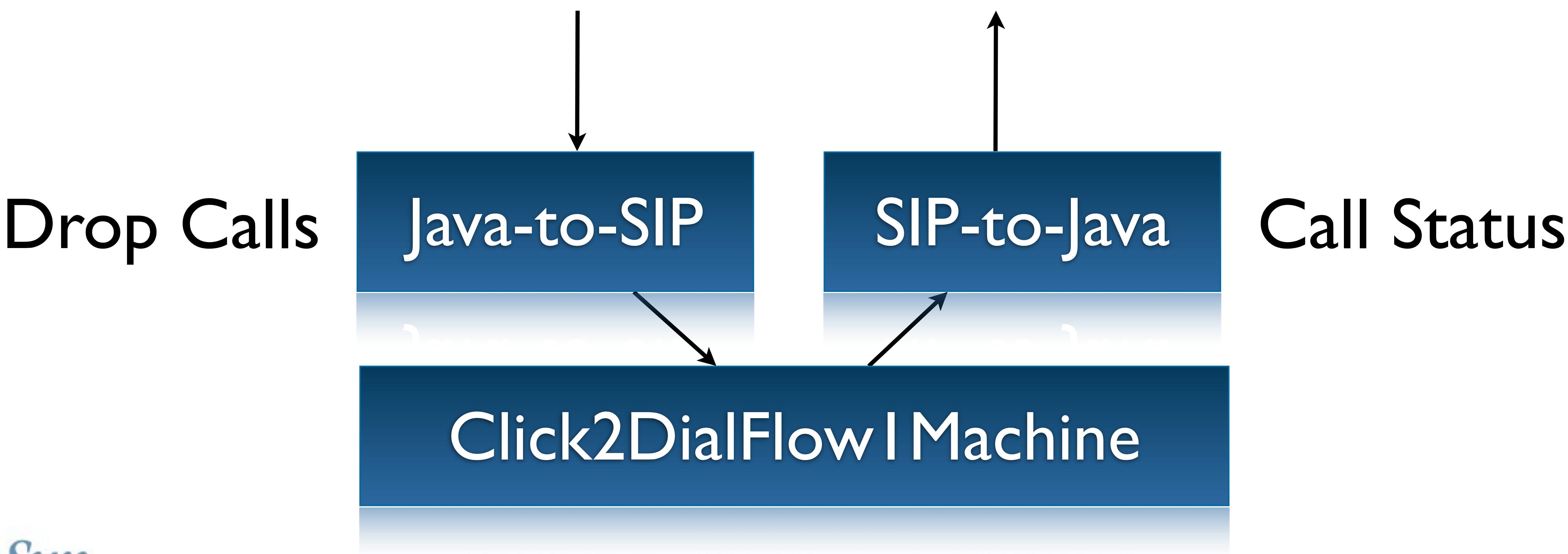
> Java-based testing framework supporting SIP, RTP, and HTTP *(via HtmlUnit)*
> Allows creation of JUnit-style test cases for converged applications
> High-level call control primitives to control test agents (e.g., call, answer, end) with options for low-level customization
> Send and receive RTP, send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

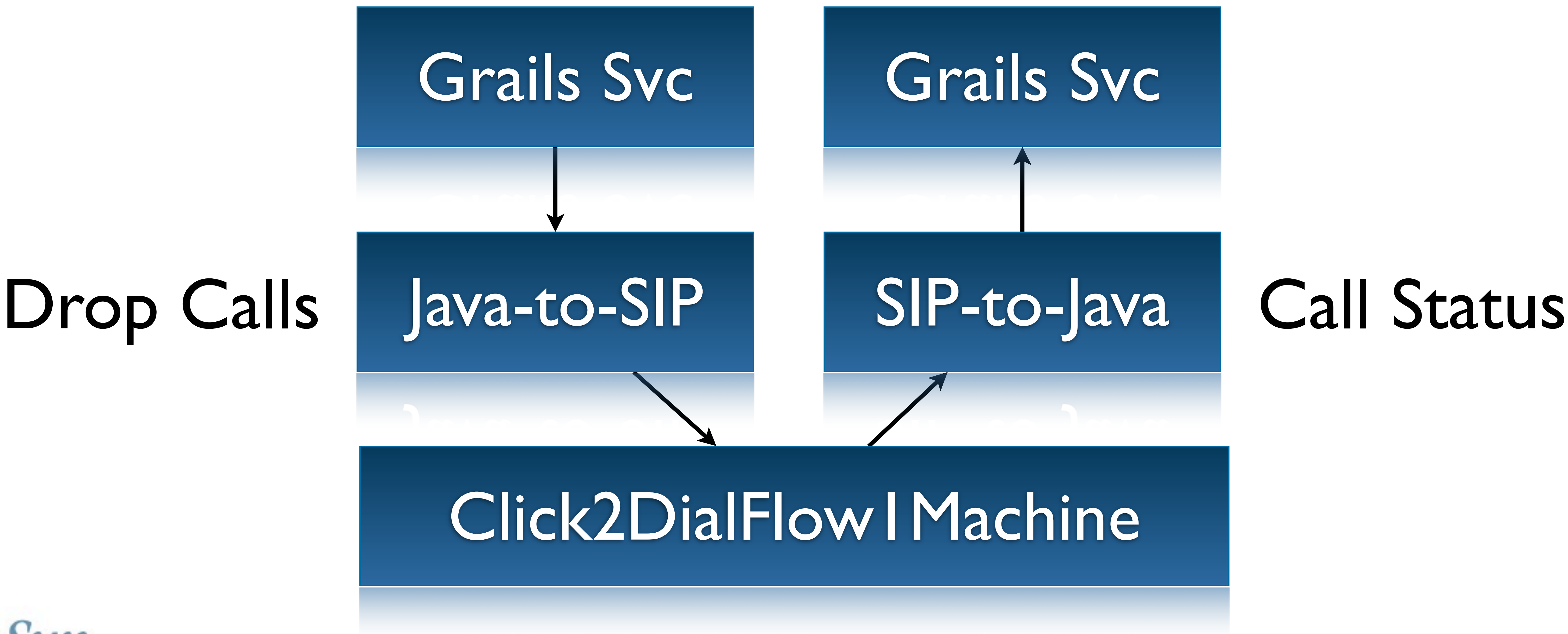
E4SS Convergence Framework

- 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

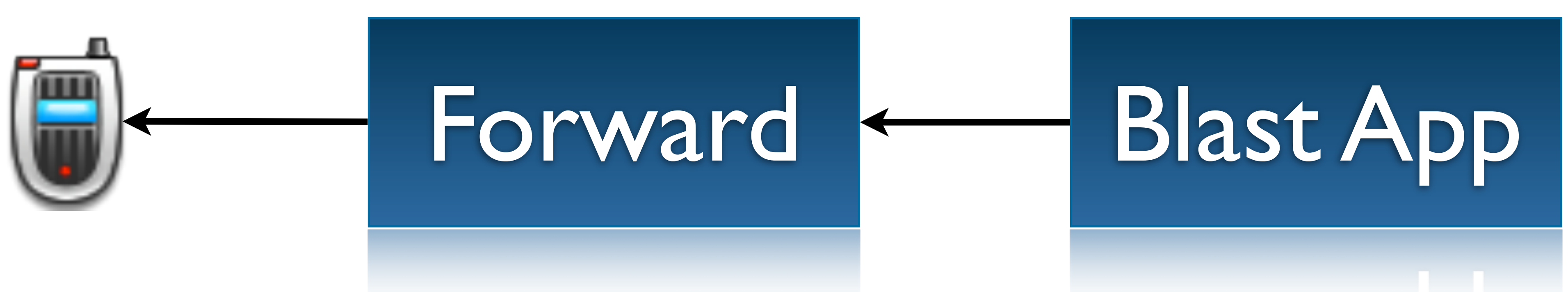


Blast Status

- 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

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Thank You

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Contact Info: {bond,tsmith}@research.att.com