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Building CLASS 5 CDRs with OpenSIPS and RabbitMQ

My experience

- Designing multi-tenant business VoIP platforms since 2009
- Lead developer for Ratetel's Virtual PBX and trunking platform
- First certified OpenSIPS professional
- Ratetel is the US sales partner for OpenSIPS Solutions

Advantages of using OpenSIPS

- Highly scalable
- Stable code base
- Can handle tens of thousands of registrations
- Central point for presence and billing
- Dynamic routing
- Packet mangling to alter packets for custom purposes
- Highly available

Advantages of using Freeswitch

- Supports more concurrent calls than most other open source PBXs (asterisk)
- Rich media handling capabilities
- Many different config methods (flat xml, lua, dynamic xml, many others)
- Stable code base and long time affinity with Opensips
- So many class 5 features, even ones you didn't think you needed

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What is RabbitMQ?

- RabbitMQ is an open source message broker software (sometimes called message-oriented middleware) that implements the Advanced Message Queuing Protocol (AMQP).
- The RabbitMQ server is written in the Erlang programming language and is built on the Open Telecom Platform framework for clustering and failover.

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Advantages of using RabbitMQ

- Robust messaging for applications
- Easy to use
- Runs on all major operating systems
- Supports a huge number of developer platforms
- Open source and commercially supported
- Reliable queuing
- Topic-based publish-and-subscribe messaging
- Flexible routing, transactions, and security.

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Traditional CDR problems

- Class 4 and Class 5 are usually separate systems from a CDR perspective.
- There are 2 sets of CDRs generated with different data, but most importantly different call-ids.
- Class 5 events during a call are not naturally logged to CDR, especially call transfers and dtmf input.
- Complex routing plans make it more problematic

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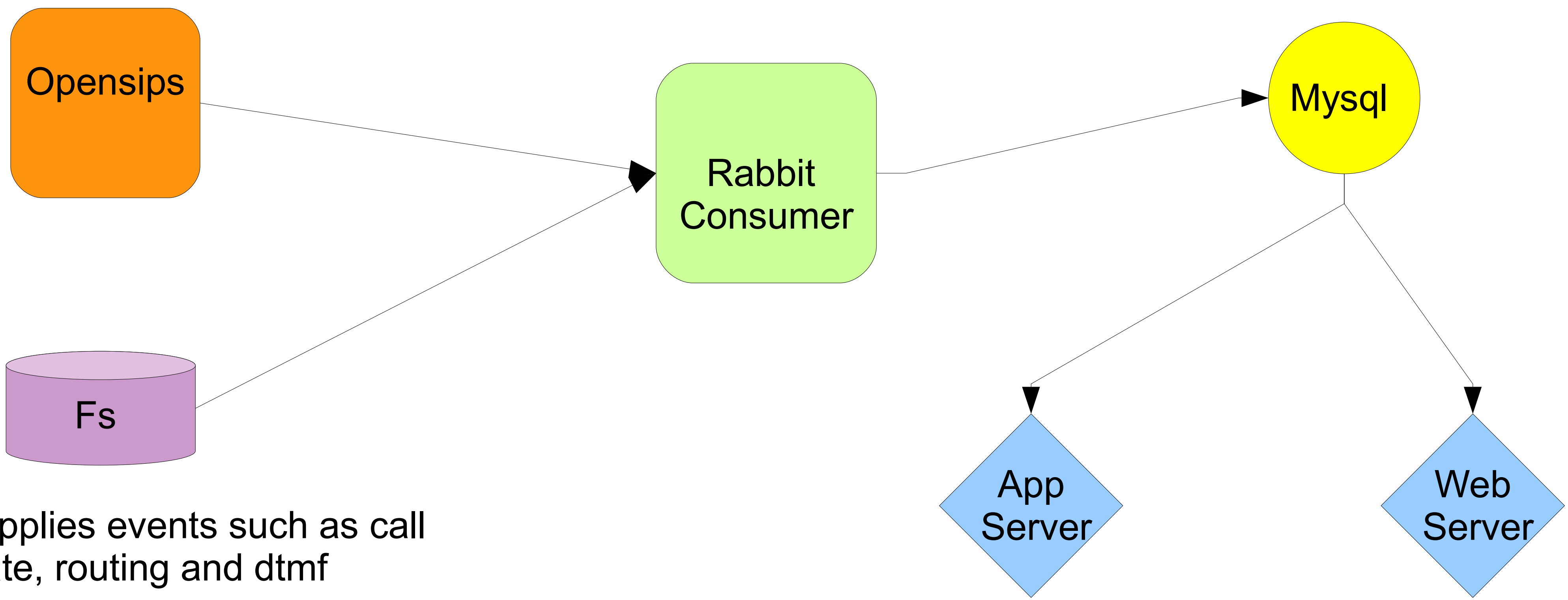
Network diagram

Supplies call start/answer/end times for billing

Store data in mysql



Supplies events such as call state, routing and dtmf

Make live call info available to web or app servers

Event Methodology

- Events can be consumed from the RabbitMQ server by any choice of clients available
- OpenSIPS is responsible for information related to call's start and end time, as well as marking billable time at the start of media
- Freeswitch will append call information based on events occurring in the CLASS 5 layer such as where the call is routed, when it's parked, put on hold, transferred, etc

Who's handling what...

OpenSIPS

- Registrations
- Ip Authentication
- Carrier facing
- nat

Freeswitch

- Call routing
- Voicemail
- Ivr
- Ring groups
- Queues
- conferences

Opensips configuration

- loadmodule "event_rabbitmq.so"
  - librabbitmq-dev required
- modparam("event_rabbitmq", "heartbeat", 3)
  - Enables heartbeat support for the AMQP communication. If no heartbeat from server is received within the specified interval, the socket is automatically closed.
  - Prevents OpenSIPS from blocking while waiting for a response from a dead rabbitmq-server. The value represents the heartbit interval in seconds

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Opensips Configuration

- modparam("event_rabbitmq", "sync_mode", 0)
  - 0 = default (async non-blocking)
  - 1 = synchronous (opensips waits for response)
- subscribe_event("E_RABBITMQ_EVENT", "rabbitmq:127.0.0.1/queue");
- raise_event("E_RABBITMQ_EVENT");
- The maximum length of a datagram event is 16384 bytes

Raising events in OpenSIPS

- Inject variables like CALLID, SRC, DST, starttime into avp variables
- raise_event("E_SIP_MESSAGE", $avp(attrs), $avp(vals))
- Calling this function on INVITE will send the first event to open a CDR record
- Calling this function on reply route will signal the start of media (billable time)
- Calling this function on BYE or CANCEL will signal the close of the CDR record

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Opensips Configuration

- Because new call-ids will be generated when calls are sent to CLASS 5, we must find a way to bind them to CLASS 4.
- append_hf("X-ORIGINAL-CALLID: $ci\r\n");
- All calls delivered to CLASS 5 will have this callid to reference as it's made available as a variable in all events sent from Freeswitch

Freeswitch Configuration

- autoload_configs/modules.conf.xml
  - Add <load module="mod_amqp"/>
- autoload_configs/amqp.conf.xml

```
<profile name="default">
  <connections>
    <connection name="primary">
     <param name="hostname" value="localhost"/>
     <param name="virtualhost" value="/"/>
     <param name="username" value="guest"/>
     <param name="password" value="guest"/>
     <param name="port" value="5672"/>
     <param name="heartbeat" value="3"/>
   </connection>
```

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Freeswitch Events

- Customize the Event Filter by editing the following lines. The default captures channel create and destroy, fs heartbeat, and dtmf.

```
<!-- <param name="eventFilter" value="SWITCH_EVENT_ALL"/> -->

<param name="event_filter"
value="SWITCH_EVENT_CHANNEL_CREATE,SWITCH_EVENT_CHANNEL_DESTROY,SWITCH_EVENT_HEA
RTBEAT,SWITCH_EVENT_DTMF,SWITCH_EVENT_CHANNEL_HOLD,SWITCH_EVENT_CHANNEL_UNHOLD,
SWITCH_EVENT_CHANNEL_PARK,SWITCH_EVENT_CHANNEL_UNPARK,SWITCH_EVENT_CHANNEL_STAT
E,SWITCH_EVENT_CHANNEL_ANSWER,SWITCH_EVENT_CHANNEL_CALL_STATE"/>
```

Freeswitch Events

- Bind the original call-id to new channels
- Use events to follow call activity in realtime
  - <action application="set" data="sip_h_X-ORIGINAL-CALLID=${sip_h_X-ORIGINAL-CALLID}"/>
- Track answers, hangups, transfers for basic CDR creation
- Enhance by injecting call data like DTMF, call parking/unparking, call hold/unhold, recording start/stop, CHANSPY events
-

Freeswitch Events

- CHANNEL_ANSWER
  - Will provide all channel variables including custom sip headers in the event
  - First bind on original callid
- CHANNEL_BRIDGE
  - Used to detect transfers as it provides all channel variables for both legs to be bridged
- CHANNEL_HANGUP_COMPLETE
  - Used to detect call hangup, all variables and sip headers available

Other Events in Opensips

- Can be used to track a multitude of other events in OpenSIPS as needed.
- Examples:
  - Alerts when counters are breached
  - Alerts when gateways become available/unavailable
  - Alerts when users register/unregister
  - Alerts when calls fail
  - Alerts on attacks such as floods, etc

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Other Events in Freeswitch

- Like in OpenSIPS, can be used to track many different kinds of events.
- Examples:
    - Conference rooms and user actions within
    - Voicemail box info after exiting mod_voicemail
    - Pin failures for call authorizations
    - Sending system status
    - Sending status of apps executed by dialplan

Thank You!

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Questions?