Janus, SignalWire and FreeSWITCH JanusCon 2019 - Luca Pradovera



A little introduction



Luca Pradovera

Sales Engineering Manager @ SignalWire



- + 20 years of RTC experience (from Flash to SIP to WebRTC, via XMPP!)
- + Pioneered the voice application approach at Mojo Lingo
- + Core contributor to the Adhearsion Ruby framework
- + Now bringing my experience to help disrupt the programmable telecoms industry

What is the presentation about?

David Duffett said this slide is important, so...

+ What is FreeSWITCH?
+ How can I use it with Janus?
+ What is SignalWire?
+ Tying it all together



FreeSWITCH 1.10, SignalWire, and much more



What is FreeSWITCH?

Helping people communicate since 2008



+ A free and open-source application server for real-time communication, WebRTC, telecommunications, video and VoIP

- + Built around a solid core, has a module-based approach to provide features
- + WebRTC(mod_verto), conferences (mod_conference), XML dialplans via HTTP (mod_xml_curl) and many others

+ Used as a PBX, as a switch, as an SBC, as a media server, as a WebRTC server...

State of the project

+ Version 1.10 was just released (August 2019), including:

- Raspberry PI support
- Debian Buster packages
- New unit test framework
- MariaDB module
- Ability to run Javascript code in the background
- Many, many more!

https://freeswitch.com/index.php/2019/08/19/freeswitch-1-10-0-release/

+ SignalWire is the official sponsor of the FreeSWITCH project, that remains free and open source



FS is moving to GitHub!

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- + Stash wasn't great, we know that!
- + Re-engage and re-energize the community
- + Required a lot of work on tooling, integrating double commits, internal process changes
- + Made possible by SignalWire
- + Contributions still drive the project, so come and help us!

GitHub

FreeSWITCH and WebRTC MCU vs. SFU



- + FreeSWITCH uses an MCU (**Multipoint Control Unit**) approach to conferencing
- + Video is mixed into a single stream, with layouts that can be defined according to many rules
- + Uses less resources on clients, more on server, suited for VoIP and mobile applications
- + Signaling is driven by Verto, a JSON-based protocol and related module

MCU video mode

Many streams in, one stream out





Example layout: 1_up_top_left+9

Fréeswitch + equals SUCCESS!



Complementary platforms

MCU vs. SFU - why not both?

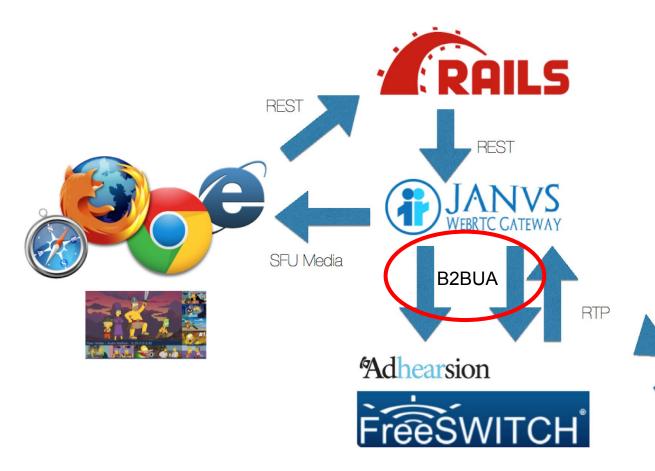


- + Janus provides an SFU
- + FreeSWITCH provides video mixing, but also SIP interop and general RTC features
- + Both platforms speak WebRTC and SIP
- + Signaling is different but it is always "just" JSON





- + Customer required SFU mode on desktop (for video quality) and MCU mode on mobile (for performance)
- + SIP bridging needed because of legacy conference room equipment
- + Recording the MCU with audio gives a ready-to-use rebroadcast format
- + Solution: A B2BUA-like agent operating on WebRTC







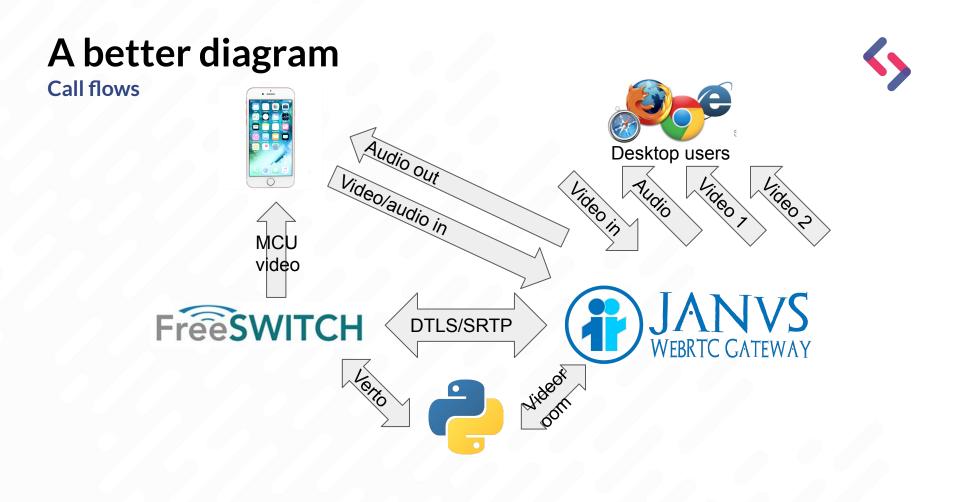
SIP

MCU Media

The implementation

Like Giacomo's, but in Python. There, done.

- + Python client connects to:
 - FreeSWITCH using Verto
 - Janus using the VideoRoom plugin as a subscriber only
 - + Everyone joins the Janus room
 - + B2BUA connects each stream to the FS conference
 - + FS conference is available with MCU mode video and mixed audio



Challenges



- + Managing participants is only easy in the current model
- + Still requires a lot of external state and orchestration
- + Solving complex NAT situations can be complicated
- + On the plus side, Python proved a great platform for this

What is next?



- + The architecture is very simple but there is no SIP access yet (subscriber only B2BUA)
- + As a result, that is much simpler and actually **works**
- + Next step is to add SIP via making the B2BUA a publisher
- + We are going to build that on...



What is SignalWire?

How we can help you succeed



What do we do?

Programmable communications



- + Cloud-based voice, messaging and video
- + SignalWire is a SaaS offering that brings the best of FreeSWITCH to the cloud, adding new features and at an unbeatable price
- + The company was founded because we believe in giving people innovative, disruptive features and not nick-and-diming them with pricing
- + One of the founders is Anthony Minessale, who also founded the FreeSWITCH project

Control APIs

Something new, something old

+ Relay

- JSON-based Websocket control channel
- Real time control with minimal latency
- Event streams and handlers
- +LāML Legacy Antiquated Markup Language
 - XML-based webhook interface
 - Paired with a REST API for outbound calls and other interactions



Relay vs. LaML

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Relay

```
"node id": "<UUID>",
"call id": "<UUID>",
"control id": "<ID>",
"record": {
    "audio": {
        "beep": true|false,
        "format": "mp3|wav...",
        "stereo": true|false,
        "direction":
"listen|speak|both",
"initial timeout": 5.0,
    "end silence timeout": 1.0,
    "terminators": "#*"
}
```

LaML

```
<?xml version="1.0" encoding="UTF-8"?>
<Response>
<Say>
```

</Say>

<Record

```
action="http://your-application.com/record.phy
method="GET"
maxLength="15"
finishOnKey="#"
/>
</Response>
```

SignalWire capabilities that Janus can leverage



+ SIP connectivity, PSTN numbers

+ WebRTC conferences and 1-1 calls

+ Ability to `tap` into an arbitrary RTP stream and send it to an endpoint

+ Recording, playback, events

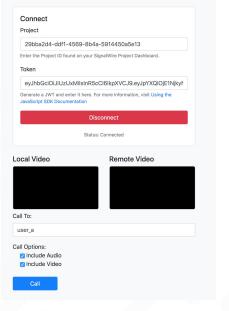
So, what's working now?

Testing integration pathways

Plugin Demo: SIP Gateway (Sofia) stop

6	sip:lpradovera-29bba2d4.sip.signalwire.com	
4	sip:janus@lpradovera-29bba2d4.sip.signalwire.com	
2+	janus	
a.		
99	Display name (e.g., Alice Smith)	
	Register	Register using HA1 secret
C	SIP URI to call (e.g., sip:1000@example.com)	
Call	⊂Use Video	

WebRTC to SIP to WebRTC





Coming soon

What's next for the Janus/SignalWire integration?

- + Making the Python connector release-able
- + Exploring the publisher connection possibilities
- + Making it as easy as possible for Janus to use SignalWire features
- + ...RIPP? (in the FAR future!)





Thank you!

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