

SignalWire

FreeSWITCH / RIPT





SignalWire - What Is FreeSWITCH?

An open-source Software-Defined Telecom Soft-Switch that can communicate via IP and Hardware and translates between various communications mediums including SIP/H.323/TDM and speaking common media formats allowing translation between SIP/WebRTC etc.

A communications platform that can terminate calls and host applications to manipulate or interact with the calls.




SignalWire - Why is RIPT Interesting

Countless organizations have struggled for years trying to scale IP Telephony. FreeSWITCH has evolved to scale with the existing challenges but many users still face problems.

Adding a RIPT endpoint to FreeSWITCH would allow for clusters of servers to exist behind existing HTTP load balancers and build media paths across challenging network NATing.

Many of the practices in VoIP thus far are antiquated and more modern techniques could prevail to make a more scalable future.



SignalWire - How Would a RIPT Module Look

The module would load into the core.

The module would get call requests either directly or forwarded by a proxy over a dedicated HTTP3 stack.

The module would extract the audio packets into its abstraction layer where calls could either be terminated or forwarded to other protocols.



SignalWire - Considerations / Challenges

Most popular protocols use RTP allowing passthrough (translation and media translation will be necessary)

A suitable HTTP3 stack must exist in C that is cross-platform and designed to support a highly threaded environment with high concurrency.

Some paradigms from traditional telephony will need to be preserved to carry signaling data from other protocols etc



SignalWire - Steps to an Implementation

Develop a RIPT STACK on top of an OSS HTTP3 implementation

Integrate the RIPT STACK into FreeSWITCH as mod_ript

Add support for receiving calls and do testing

Figure out the client-side so calls can be pushed b2b