22.0.0

New Feature

PS-12 Add remote wipe

PS-10 Add API to server for Asterisk 16 for conference and transfer call.

Improvement

PS-29 PrivateServer ISO build in Git environment and versioning

PS-23 Hide the push setting "Send caller ID"

PS-21 Enforce protection on malicious SIP INVITE request

PS-19 Adapt privateserver-release package to git versioning

PS-11 Call the transfer/conference API from Asterisk

Bug

PS-30 Call transfer/conference doesn't work when initiated from snom

PS-27 Server certificate update does not prepare certificate file for clients when SSL Pinning is enabled

PS-26 CallDetails Exporting Error CSV

PS-24 Trunk failover does not work with Asterisk 16

PS-22 Fix audio codec for snom (fixed line) account

PS-17 Routing rules match is not working properly

PS-15 Tomcat's stacktrace.log is not rotated

PS-18 Fix dialing rule configuration