REDCOM Sigma® 2.1.1 feature list

Call & session control software

APPLICATIONS

- Virtual PBX
- Hosted PBX
- Secure Conference Bridge
- Class 5 End Office expansion
- Local Session Controller

UNIFIED COMMUNICATIONS

- XMPP with Presence
  - Archive messages
  - Federation with 3rd party servers
  - Share rosters across federated networks
- Conference
- Voice Mail
  - Voice Mail to email
  - Sub-mailboxes
  - Message management
- Video (P2P)

END USER/SUBSCRIBER

- ANI Call Rejection
- Call Forwarding
  - Conditional based – by time of day
- Call History
- Call Hold
- Call Jump (Seamless Cell Phone Transfer)
- Call Waiting
- Call Transfer
- Caller ID Blocking
- Calling Name Delivery (CNAM)
- Do Not Disturb (DND)
- Find Me (Single Number Services)
- Incoming Caller ID
- Message Waiting Indicator
- MLPP support on non-AS-SIP Lines
- Selective Call Acceptance
- Selective Call Rejection
- Speed Dialing
- Time and date feature control
- Integrated support for REDCOM Secure Client

SYSTEM FEATURES

- Announcements
  - Pre-recorded
  - Customer recordable
  - Localization
  - Per user
- Asynchronous DNS Lookup
- Auto Attendant/Direct Inward System Access (DISA)
- Bandwidth Budgeting
- Broadcast Ringing Group
- Call Detail Records Configuration
  - SMDR/CDR/LAMA
  - Bellcore AMA Format
- Call Queuing
- Comfort Noise Generation
- Direct Inward Dial (DID)
- Direct Outward Dial (DOD)
- E.164 compliance
- Echo Cancellation
- In-band PTT functionality
- IPv4/IPv6 Dual-Stack
- Least Cost Routing
- Multitenancy
- Music on Hold
- Network Tandem operation
- Percentage Trunking
- Silence Suppression
- Simultaneous Ringing
- SIP/SDP message manipulation
- Toll Restriction
- Transcoding
  - RFC 2833 / DTMF
  - Codecs
- Transrating
- Trunk registration and authentication

CUSTOMIZABLE CALL ROUTING

- Alternate routing
- Dialed number parsing
- Full URI analyzer for routing
- Incoming number parsing
- International Numbering Plans
- North American Numbering Plan compliance
- Private network numbering plans
- Routing by time and day
- Virtual Directory Number

CONFERENCING

- Pre-set/Blast Dial
- Meet-me/Scheduled
- New Conference Manager app
  - Monitor and manage multiple conferences across several sites
  - Detailed attendee visibility & control
- Security level announcements and reverse caller ID
- Robust access control by user ID, ANI, PIN code, or clearance level

PROVISIONING

- New Device Management app
- Provisioning for TEO, Grandstream, Polycom, Cisco, Snom, and Yealink phones
- Custom phone provisioning
- Centralized phone software updates
- Remote device management
- Device firmware management

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END USER WEB INTERFACE

- End user access portal
- End user line management
- OA&M
- White Labeling
- Customizable Themes per user

OPERATION & ADMINISTRATION

- Web-based
  - User Time and date feature control
- CSV upload and download
- DHCP Server
- External LDAP database support for user management
- Hunt Groups
- LUA scripts for Media Server customization
- Mass editing of lines & trunks
- Network Diagnostics App
- Real-Time System Status Application
- REST API
- Security Logging
- SNMP support for alarms and system monitoring
- Syslog

REPORTING

- On-demand or scheduled
- Archived and/or distributed via email
- Reports can be exported to various file types, including PDF and Microsoft® Word
- Tracking of RTCP Voice Quality Data
- Available reports include:
  - Network Utilization Summary
  - SIP Registration Summary
  - System Summary
  - Failed Calls Summary
  - Media Engine Statistics
  - Precedence Calls Summary
  - Preemption Calls Summary
  - Subscriber Traffic Details
  - Trunk Traffic Details

CODEC SUPPORT

- G.711a 10/20/30/40ms
- G.711μ 10/20/30/40ms
- G.722 (HD) 10/20/30/40ms
- G.722.1 (HD) 24/32kbps 20/40ms
- G.723.1 6.3kbps send@30ms
  5.3/6.3 receive@30ms
- G.726 16/24/32/40kbps at 10/20/30/40ms
- G.729ab 10/20/30/40ms
- iLBC 20/30ms
- Opus
- Speex

CALL CONTROL PROTOCOLS

- SIP (Trunking & Lineside)
- AS-SIP (Trunking & Lineside)

SECURITY & ENCRYPTION

- ANI Lockdown
- HTTPS
- IPSec
- PKI (Public Key Infrastructure)
- SRTP
- SSL 3.0
- Suite B
- TLS 1.0–1.2
- Password protection
- Private Certificate Authority (CA)
  - Create local CA within Sigma
  - Issuance; renewal and rekey; revocation services
- Remote authentication of users (i.e. RADIUS & LDAP)
- Single sign-on with browser certificates
- Subscriber PIN Access
- Online Certificate Status Protocol (OCSP)

MINIMUM PLATFORM REQUIREMENTS

- Runs on Intel® 64 Architecture
- 1GB RAM
- 20GB Disk Space

INSTALLATION OPTIONS

- Bare Metal
- Virtualized
  (i.e. VMware®, Hyper-V®, KVM, etc.)

SUPPORTED SIP-RELATED RFCS

- RFC 2833 – RTP Payload for DTMF
- RFC 3261 – SIP
- RFC 2976 – SIP INFO method (i.e. for SIP-Q)
- RFC 3262 – PRACK method, 100rel
- RFC 3263 – Server location
- RFC 3264 – Offer-answer model for SDP
- RFC 3265 – SUBSCRIBE/NOTIFY method, Events
- RFC 3311 – UPDATE method
- RFC 3323 – Privacy mechanism for SIP
- RFC 3326 – Reason header field
- RFC 3515 – SIP REFER method
- RFC 3608 – SIP Extension header field
- RFC 3891 – Replaces header field
- RFC 3892 – Referred-by header field
- RFC 3903 – PUBLISH method
- RFC 4028 – SIP session timers
- RFC 4092 – ANAT in SIP
- RFC 5630 – SIP-SIPS
- RFC 5954 – Essential correction for IPv6 ABNF and URI comparison rules
- RFC 6140 – Registration for multiple phone numbers

SUPPORTED SDP-RELATED RFCS

- RFC 2327 – SDP
- RFC 3266 – Support for IPv6
- RFC 3605 – RTCP attribute in SDP
- RFC 4091 – Alternative Network Address Types (ANAT)
- RFC 4566 – SDP-new
- RFC 4568 – Security

SUPPORTED EVENT-PACKAGE RFCS

- RFC 3842 – Message waiting indication
- RFC 3856 – SIP extension for presence
- RFC 4235 – INVITE-initiated dialog