Splicecom

maximiser

Version 3.1 Features Summary

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The **max**imiser system features

PABX Features

Max. No. of Users	Up to 5,000
Max. No. of Virtual Users	Up to 5,000
Phones Supported	Analogue & IP (H.323 & SIP)
Max. No. of Trunks	Up to 3,800
Trunks Supported	BRI, PRI, IP, WAN & LAN, DPNSS
Centralised and/or Distributed Architecture	●
Redundant Call Server Module	Optional
Max. No. of Contacts	10,000
Automatic External Dial Detection	●
Transparent Multi-Site Operation	●
Centralised and/or Distributed VoiceMail	●
Alternate Call Routing (IP & LCR)	●
Analogue Caller Display Phone Support	●
Virtual Key System Emulation	●
Call Hold/Transfer/Forward/Park/Pick-up	●
Call Barring	●
Call Waiting	●
Distinctive Ringing	●
Simultaneous Ringing	●
External/Internal Follow-Me To (Users & Departments)	●
Forward On Busy/No Answer	●
Do Not Disturb with Exceptions	●
System & User Speed Dials	●
Contact Directory (LDAP)	●
Transparent Hot Desking across sites	●
Time Profiles	●
DDI/MSN with Time Profiles	●
System-Wide Dial Plans with Time Profiles	●
Least Cost Routing with Time Profiles	●
2-Stage LCR Set-Up (In-band DTMF)	●
Line Groups (Reserve Voice/ Data Capacity)	●
Reserve Outgoing Trunk Capacity	●
Call Logging Output	●
Call Management Application	Optional
Operators Console (Windows & Mac OS X)	Optional
External Door Relay Drivers	2 per Call Server
Inputs for External Alarms	2 per Call Server
Integral Digital Music-on-Hold	6 Channels per Call Server
External Music on Hold inputs	Via Analogue Port
Account Codes	●
Area Codes	●
Paging	Via PCS 400/100/50* or Analogue Paging Port
Paging Groups	●

*IP Softphone only

PABX Features (cont)

3-Party Conference Calls	●
Meet-Me-Conference	●
Ex-Directory Operation	●
Personal Call Recording	PCS 400/100/50
Automatic Call Recording	●
Call Recording Management Application	Optional
Absence Messages	Pre-Set & Customised
Dial Emergency Breakthrough	●
Dial Forced Call Waiting	●
Dial Direct Pickup	●
DDI to DTMF Conversion for Fax Servers	●
2-way transfer to mobile phone	●
Homeworking - IP, Analogue & GSM	●
Call Back when Free	Optional

Voicemail Features

Off-switch voicemail	●
Voicemail supplied as standard	30 VM Boxes per Call Server
Voicemail supplied as standard	10 VM Boxes per Remote Call Server
Max no. of voicemail boxes	1000 per Call Server
Max no. of voicemail boxes	300 per Remote Call Server
Max no. of simultaneous voicemail calls	8 per Call Server
Max no. of simultaneous voicemail calls	4 per Remote Call Server
Max no. of simultaneous off-switch voicemail calls	Platform Dependent
Recording Time	300 Hours per Call Server
Recording Time	100 Hours per Remote Call Server
Voicemail for Users, Departments & Virtual Extensions	●
Local & remote message retrieval	●
Personalised greetings	●
PIN code security	●
Automatic CLI recognition	●
One-touch dialback	●
Forwarding of voicemail to email	●
Voicemail alert to email and SMS	●
SMS Text Messaging	●
Simple Auto attendant	●
Multi-level Auto attendant	Optional
Wake-Up/Alarm/Scheduled Calls	Optional
Voice XML-based IVR	Optional
Text-to-Speech Conversion	Optional
Spoken email collection and reply	Optional

Contact Centre Features

Departments (Hunt Groups)	No Limit
Capabilities (Skills Based Routing)	●
Call Routing on CLI	●
Distinctive Ringing	●
Collective, Rotary, & Sequential Call Distribution	●
Oversquare Operation	●
Group Overflow	●
Call Queuing	●
Queue Entry & Update Messages	●
Wrap-Up Timer	●
Queue Limits for Departments	●
Agent Login/Out	●
Multiple Music-on-Hold Zones	●
Out of Service Short Codes	●
Silent Monitor	●
Supervisor Intrude	●
Informal Call Centre Desktop Client	Optional
Multi-media Inbound Contact Centre	Optional
Outbound Call Centre - Preview, Progressive & Predicitive Dialling	Optional
Queue Buster	Optional

IP Features

Static IP LAN-to-WAN Routing	●
IP WAN Header Compression	RFC2507/8/9
ISDN Dial-on-Demand & Dial Back-up	●
Integral WAN & LAN Security	●
PPP & MP	●
PAP & CHAP	●
Virtual Private Networking (VPN) - PPTP & Open VPN	●
Integral Apache Web Server	●
Integral DHCP Server	●
Integral LDAP Database	●
IP Multicasting	●
MP Fragmentation	●

H.323 VoIP & SIP VoIP (WAN & LAN Trunks)	●
H.323 IP & SIP VoIP Telephony Extensions	●
802.3af Power over Ethernet for IP Telephones	Call Server & Trunk Module (Optional)
Integral H.323 Gateway	●
Integral H.323 Gatekeeper	●
Support for Multiple External H.323 Gatekeepers	●

VoIP and IP Telephony Features (cont)

H.450.2/.4/.5/.7 Supplemental Services	●
DiffServ QoS	●
Voice Path Optimisation (Anti-Tromboning)	●
G.711, 64kbps Voice	●
G.729a, 8kbps Voice	Optional
Transparent Fax over IP	●
SIP Proxy Server	●
NAT Traversal	●

Integrated Management Features

Web Browser Management	●
Single Replicated Database	●
Access Security	●
Dynamic Configuration Changes	●
Web Based Help Text	●

Interfaces for 3rd Party Development

H.323/H.450	●
SIP	●
LDAP	●
HTML	●
PHP	●
XML	●
Voice XML	●
TAPI 2.2 (3.0)	●
Apple Script	Via PCS 50 for Mac OS X
SpliceCom PCS Partner Specification	●
SpliceCom Call Routing Protocol	●
SpliceCom Call Logging Interface	●
SpliceCom Busy Lamp Field Specification	●

Bundled Applications

Voicemail & Auto Attendant	●
Proactive Communication Station 50 - IP Softphone and Analogue Phone Partner (Windows, MAC OS X & Linux**)	●
Integrated Messaging (SMTP) Support	●
Unified Messaging (IMAP) Support	●
TAPI 2.2 (3.0) Service Provider Interface	●

**Phone Partner only

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