Splicecom maximiser

Free your Network

Version 3.2 Features Summary

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The **max**imiser system features

PABX Features

Max. No. of Users	Up to 10,000
Max. No. of Virtual Users	Up to 10,000
Phones Supported	SIP, H.323 & Analogue
Max. No. of Trunks	Up to 5,000
Trunks Supported	BRI, PRI, SIP, H.323 & 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	●
3-Party Conference Calls	●

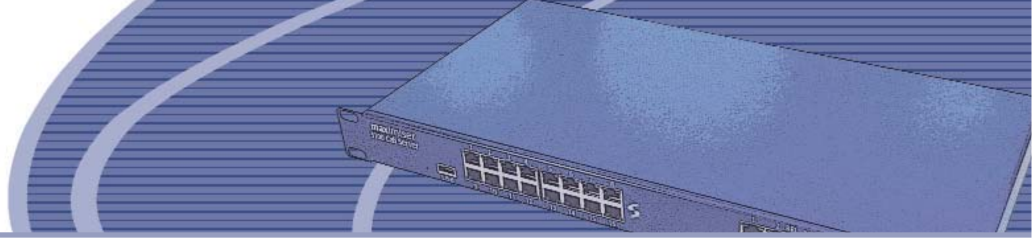
PABX Features (cont)

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
Micros Fidelio Opera Interface	●
ETSI Malicious Call Notification	●

Voicemail Features

Off-switch voicemail	●
Voicemail supplied as standard	10 VM Boxes per 5100
Voicemail supplied as standard	5 VM Boxes per 5108
Max no. of voicemail boxes	1000 per 5100
Max no. of voicemail boxes	50 per 5108
Max no. of simultaneous voicemail calls	4-16 per 5100
Max no. of simultaneous voicemail calls	2-8 per 5108
Max no. of simultaneous off-switch voicemail calls	Platform Dependent
Recording Time	1500 Hours per 5100
Recording Time	500 Hours per 5108
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

*IP Softphone only



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	●

VoIP and IP Telephony Features	
SIP & H.323 IP Telephony	●
SIP & H.323 IP Trunks	●
SIP Proxy Server	●
802.3af Power over Ethernet for IP Telephones	●
Integral H.323 Gateway	●
Integral H.323 Gatekeeper	●
Support for Multiple External H.323	●
Gatekeepers	●
H.450.2/.4/.5/.7 Supplemental Services	●

VoIP and IP Telephony Features (cont)	
DiffServ QoS	●
Voice Path Optimisation (Anti-Tromboning)	●
G.711, 64kbps Voice	●
G.729a, 8kbps Voice	Optional
Transparent Fax over IP	●
Media Relay for NAT Traversal	●

Integrated Management Features	
Web Browser Management	●
Single Replicated Database	●
Definable Administration Access Levels	●
Dynamic Configuration Changes	●
Web Based Help Text	●
Integrated VoiceXML Script Builder	●
Bulk Updates	●

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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