Splicecom

maximiser

Modules Datasheet

4100 Call Server
4140 Remote Call Server
4200 Trunk Module
4315/4330 Phone Modules
4400 Voice Compression Module

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Introduction

The **max**imiser business telephony system from SpliceCom provides a breakthrough in integrated voice communications. Developed from state of the art technology it delivers real life benefits associated with many traditionally separate components in one single, seamless system, currently supporting 8 to 5,000 extensions. When used in conjunction with SpliceCom's broad range of Proactive Communication Stations (PCS), the **max**imiser allows you to converge your telephone system with your core business applications, "pushing" the right information to the desktops of the right people at the right time.

Through the use of an innovative architecture, the **max**imiser eliminates the physical and geographical limitations of traditional telephone systems, allowing great savings to be made on administration, management and infrastructure costs, through the unification of networks. This approach allows all businesses, irrespective of size, to benefit from extended communications, and more importantly protects your initial investment by growing with you as your need for communication scales and becomes ever more demanding.

A single integrated business telephony solution

The **max**imiser system is completely modular and comprises of three different, 19" , 1U high, rack mountable components; Call Server, Trunk Module and Phone. Module.

4100 Call Server Module

At the very heart of the **max**imiser architecture lies the Call Server. PBX, voicemail/auto attendant, IP Router, Voice over IP (VoIP) Gateway and Gatekeeper, Dynamic Host Configuration Protocol (DHCP) Server, Apache Web Server and Lightweight Directory Application Protocol (LDAP) database are all integrated within the 4100 Call Server, which is supplied as a single, slim line unit. The 4100 Call Server also provides interfaces to support an extensive range of Trunk and Wide Area Network (WAN) services; a 30 channel Primary Rate ISDN (PRI) interface supporting Q.931 or DPNSS, Quad Basic Rate ISDN (BRI) (8 channels) and a single V.11 interface which supports point-to-point digital services at speeds up to and including 2.048Mbps. An eight port 10/100 Mbps Ethernet switch provides LAN connectivity, whilst a secondary power supply provides Power over Ethernet (PoE) to devices (typically IP Phones) supporting the 802.3af standard. Two 3.5mm sockets allow door relays to be connected and controlled, whilst two trigger inputs accepting signals from fire/intruder alarms, panic buttons, etc. are also provided in a mini-DIN format. All user-facing interfaces are terminated in RJ45 sockets and mounted on the front of the 4100 Call Server. This allows it to be installed in the same cabinet as your structured wiring and patched directly across. All trunk interfaces are mounted at the rear of the 4100 Call Server.

PBX

As a business telephone system, the **max**imiser architecture supports both IP and analogue phones. Analogue phones are connected to the system via the 4315/4330 Phone Modules, whilst IP terminals are attached to the 10/100 Mbps Ethernet at the front of the 4100 Call Server. As such, a single 4100 Call Server provides all the functionality required to provide a small, yet sophisticated telephony solution. From basic call handling capabilities through to applications until now only found on high-end, PBX's (such as One Number, 2-Way, Mobile Transfer, Simultaneous Ringing and Skills Based Routing), the **max**imiser provides easy-to-use and consistent operation independent of the type of phone deployed. The system capacity of the **max**imiser can be increased by adding LAN Switches to support a greater density of IP Phones, 4315/4330 Phone Modules to support more Analogue Phones and/or 4200 Trunk Modules to support further ISDN and WAN connections.

Voicemail/Auto Attendant

A fully integrated 8 port, 30 hour voicemail and auto attendant system with 30 active mail boxes enabled, is supplied as standard with each **max**imiser system and resides on the 4100 Call Server. A total of 1,000 voicemail boxes on each 4100 Call Server can be activated as required via license keys. If greater number of concurrent calls and/or mailbox capacity is required the **max**imiser voicemail/auto-attendant application can be run off-switch on a Linux PC or Server. **max**imiser has the ability to support multiple voicemail systems which can be independent of both platform and geography allowing them to be sited in the most appropriate location - or locations. The multi-port registration architecture allows voicemail to be spread across many servers, increasing both the overall capacity and resilience of the **max**imiser system. Voicemail facilities can be further enhanced through the addition of SpliceCom's Enhanced Speech Processing (ESP), which provides multi-layer auto-attendant, dial through and voice forms via a VoiceXML based Interactive Voice Response (IVR) application.

ISDN Trunks

The 4100 Call Server provides four Basic Rate ISDN interfaces and a single Primary Rate ISDN interface, these can all be used in TE (connecting to ISDN services) or NT (connecting 3rd party equipment - PBXs, Routers, Fax Servers, etc.) modes. When connected to ISDN services, these trunks can be used for incoming and outgoing voice (standard and IP) and IP data calls. These IP data calls can be triggered on-demand, or used as back-up for the IP WAN link. In situations where more ISDN trunks (or WAN links) are required, up to four additional 4200 Trunk Modules can be added to each 4100 Call Server.

Powered 10/100 Mbps LAN Switch

This eight port, Layer 2 switch provides the means of connecting other **max**imiser modules, and IP Phones to the 4100 Call Server - and/or the 4100 Call Server to the existing company LAN. If more than eight ports are required, this switch can be cascaded into other 3rd party switches. A maximiser system operates completely independent of LAN infrastructure. You can choose to keep voice calls totally separate from your data traffic by either running **max**imiser in a centralised, PBX replacement configuration, as in a traditional voice and data infrastructure, or over a distributed voice-only LAN. Alternatively you can choose to converge your voice and data network into one. If the decision is made to keep voice traffic separate the integral LAN switch will be utilised to link to 4315/4330 Phone Modules, 4200 Trunk Modules and IP Phones only, again further LAN switches can be deployed where greater connectivity is required. In a fully converged voice & data network, that is one where Quality of Service (QoS) is available throughout the network, the **max**imiser modules, and IP Phones can be overlaid as a "voice application" on top of your existing LAN infrastructure. 802.1p and DiffServ are supported to ensure QoS is maintained in converged networks. Where IP hardphones are utilised, the 4100 Call Server's support for Power over Ethernet (PoE), allows 802.3af compliant devices, such as SpliceCom's PCS 400/200/100 to be powered over the LAN. This eliminates the need for every phone to be separately powered, thereby saving costs and increasing overall system resilience in the case of mains power failure.

IP Router

The integrated IP Router embedded within the **max**imiser architecture, is distributed throughout the Call Server and Trunk Modules. The primary role of this Router is to forward internal Voice over IP (VoIP) calls over the Wide Area Network (WAN) between locations where no IP Router infrastructure already exists. Supporting static routes, the **max**imiser's IP router is standards based, allowing interoperability with similar devices from other vendors, and provides transport for both data and voice over ISDN and point-to-point WAN links. DiffServ is supported to ensure voice quality is preserved in a converged voice & data network.

IP WAN Interface

This connection operates at speeds up to and including 2.048Mpbs and routes IP voice and data traffic over fixed point-to-point digital services such as BT's Kilostream, Kilostream n or Megastream services - the latter being achieved through the use of a V.11-to-G.703 convertor. ISDN calls can be utilised in conjunction with this link to back it up in case of failure.

H.323 Gatekeeper

All **max**imiser trunks and phones register with the H.323 Gatekeeper integrated within the 4100 Call Server architecture, utilising standards based, secure H.323 methodology. Although IP Phones, or in the case of analogue phones, 4315/4330 Phone Modules, pass speech directly between themselves using the Real Time Protocol (RTP) once the call has been set-up, the H.323 Gatekeeper is responsible for all Phone Registrations, Call Routing and Call Logging. Where integration into large IP networks is required, the **max**imiser's internal Gatekeeper has the ability to register with multiple external H.323 Gatekeepers, on behalf of all system components.

H.323 Gateways

Trunk Gateways register with the H.323 Gatekeeper and act as a resource for the Gatekeeper to route calls to and from the public telephone (PSTN) network. Both Call Server and Trunk Modules are capable of operating as H.323 Gateways, allowing distribution across physically separate modules and geographic sites. Where required, an optional 4400 Voice Compression Module supporting 16 -32 channels, or internal Voice Compression Card supporting up to 8 channels can be deployed to allow a higher density of voice calls to be transported across a fixed speed link. 64 kbps, G.711 voice is supported as standard on **max**imiser whilst the Voice Compression Module/Card offers 8kbps, G.729a based encoding, providing an excellent balance between voice quality and bandwidth efficiency. The 4315/4330 Phone Modules also operate as H.323 Gateways.

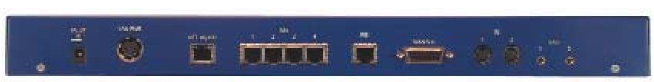
Distributed LDAP Database

The 4100 Call Server forms the heart of the **max**imiser as it runs the database that holds information relating to the rest of the system components and is therefore core to the overall system operation. In a multi-site network, or a high-availability single-site running standby/redundant Call Servers, this database information is replicated centrally and then distributed to all the other 4100 Call Server. This allows sites that reside on different continents to operate as if they were located in the same building.

Support for the Lightweight Directory Access Protocol (LDAP) allows external databases and applications to both read and write to the **max**imiser database in real-time. This enables the development and use of 3rd party applications to automate configuration tasks, relay live system information and change/update parameters and settings "on-the-fly."

Apache Web Server

maximiser provides unique functionality in its ability to converge voice with web-enabled IT applications to greatly improve desktop productivity - delivering the right information, to the right people, at the right time. This data is pushed to the desktop via the large touch screen displays of the PCS 400/200 IP Phones, or within the PCS 50 application running as either an IP Softphone or "partner" to an analogue phone on Windows, Mac OS X or Linux PCs. The information pushed to these displays can reside on the internet, intranet or be run on web-based application servers located anywhere on the LAN infrastructure. Web pages and applications can also be mounted on the Apache Web Server running within the 4100 Call Server, which is primarily used to control the images viewed on the PCS 400/200/50, and to provide the interface for system configuration, via a standard browser.



4200 Trunk Module

The 4200 Trunk Module extends the Trunk and WAN capabilities of the **max**imiser, in terms of both quantity of trunks supported and geographic location. The initial 4100 Call Server provides a single Primary Rate ISDN (PRI) (30 channels), Quad Basic Rate ISDN (BRI) (8 channels) and a single V.11 interface which supports point-to-point digital services at speeds up to and including 2.048 Mbps. If greater trunk density is required, the addition of a 4200 Trunk Module provisions a further PRI, Quad BRI and a single V.11 interface, again supporting digital services at speeds up to and including 2.048 Mbps. Connectivity between the 4100 Call Server and 4200 Trunk Modules is achieved over a private LAN network, for separate voice and data networking, or overlaid on the internal company LAN if full voice & data convergence is a requirement. The latter requires Quality of Service (QoS) to be supported throughout the LAN network. It is this LAN connectivity that enables the 4200 Trunk Modules to be located anywhere where they can be linked, via Ethernet, to the 4100 Call Server. An eight port, 10/100 Mbps LAN switch supporting QoS is provided at the front of the 4200 Trunk Module to meet this need. Where IP hardphones are utilised, there is an option to add Power over Ethernet support to the 4200 Trunk Module, allowing devices supporting 802.3af , such as SpliceCom's PCS 400/200/100, to be powered over the LAN. All ISDN & WAN interfaces are mounted at the rear of the 4200 Trunk Module.

4315/4330 Phone Module

The Phone Module is available in two variants, providing connectivity for up to fifteen (4315) or thirty (4330) analogue devices. These can be traditional analogue (POTS) telephones, fax machines or modems. Where standard analogue phones are already utilised, the deployment of 4315/4330 Phone Modules provides un-paralleled investment protection, whilst allowing the overall capital cost of a **max**imiser system to be minimised. These phones can then be further energised by using them in conjunction with the Proactive Communication Station (PCS) 50 application, which runs on a desktop or laptop PC as a partner to the analogue phone, independent of the operating system - Windows, Mac OS X and Linux are all supported. Connectivity between Phone Modules and their local Call Server, is achieved over a private LAN network, for separate voice and data networking, or overlaid on the existing company LAN with Quality of Service (QoS) support, if full voice & data convergence is a requirement. This connectivity enables Phone Modules to be located anywhere where they can be linked to a Call Server - either locally via the LAN or remotely across an IP WAN infrastructure. A 10/100 Mbps LAN port is provided at the front of the 4315/4330 Phone Module to meet this need. All analogue extension interfaces are also positioned at the front of the 4315/4330 Phone Module. This architecture allows installation costs to be drastically reduced as a single Category 5 cable (or better) running from a Call Server to a 4330 Phone Module replaces up to 30 cables in a traditional PBX installation.

Analogue Extensions

The Phone Module allows standard, low-cost 2-wire, analogue telephones to be used, reducing the overall system cost. Modems and analogue fax machines can also be connected to this module, whilst analogue DECT devices can be utilised where mobility is required - in a factory environment or by a school secretary for example. DECT solutions provide great flexibility, allowing your employees to always be in contact without needing to be desk or office bound. Where standard Caller Display phones are used the Phone Module forwards information allowing your employees to see who is calling them before they answer. As well as the caller's number (CLI), their name (if it is known to you) is also displayed. This information is held within the distributed **max**imiser database. Finally, the name associated with the number dialled by the caller is also shown, this could be an employee's direct number (DDI), or it may be that they are participating in a hunt-group; sales, admin, support, etc. The combination of this information allows your employees to know who's calling and to answer the call accordingly based on the number rung. The result is improved client handling and therefore happier customers, the very essence of Customer Relationship Management. The Caller Display is also used to notify users of the number of new or outstanding voicemail messages they have.

10/100Mbps LAN Port

A 10/100 Mbps Full Duplex Ethernet port provides the means of connecting the Phone Module to its local Call Server. This could be directly or via your existing LAN or WAN infrastructure. When installing a **max**imiser system, you have the choice to keep voice separate from your data traffic, as in a traditional voice and data infrastructure, or converge the voice and data network into one. If the decision is made to keep voice traffic separate Phone Modules will be interconnected to maximiser (Call Server, Phone & Trunk) Modules only. LAN switches can be deployed to provide extra interfaces where more connectivity is required. In a fully converged voice & data network the **max**imiser modules and data LAN network traffic will coexist on a single LAN network with QoS ensuring that voice calls have priority over data traffic, so ensuring that speech quality is preserved.

H.323 Client

To enable voice calls made from standard analogue phones connected to the 4315/4330 Phone Modules, to be converted into IP voice calls, the module acts as an H.323 Client. All **max**imiser trunks and phones register with the H.323 Gatekeeper integrated within the Call Server architecture, utilising standards based, secure H.323 methodology. Internal calls between analogue phones connected to Phone Modules, pass the voice traffic directly between themselves once the call is set-up. The integral H.323 Gatekeeper, which resides within the Call Server, is responsible for call set-up - Phone Registrations, Call Routing and Call Logging. Where integration into large IP networks is required, the **max**imiser's internal Gatekeeper has the ability to register with multiple external H.323 Gatekeepers, on behalf of all system components.

4400 Voice Compression Module & Internal Voice Compression Card

As an alternative to the 64kbps, G.711 voice supported as standard on **max**imiser, 8kbps, G.729a based encoding, provides an excellent balance between voice quality and bandwidth efficiency. The 4400 Voice Compression Module provides 16 channels, of 8kbps, G.729a based compression and can be deployed to provide a higher density of calls between 4100 Call Servers/4140 Remote Call Servers over low speed WAN links. A further 16 channels of voice compression can be enabled per 4400 Voice Compression Module through license keys - giving a maximum of 32 compression channels per module. By using a standard LAN for interconnectivity between the Voice Compression Module and its associated Call Server / Remote Call Server, there is no limit to the number of compression channels that can be practically deployed. This allows multiple 4400 Voice Compression Modules to be used with single or multiple Call Servers, with the compression channels being dynamically allocated as, or when, required. Alternatively, for lower density call requirements, an internal Voice Compression Card supporting up to 8 channels of 8kbps, G.729a based compression can be fitted inside the 4100 Call Sever & 4140 Remote Call Server.

4140 Remote Call Server

The 4140 Remote Call Server extends the cost-effective reach of **max**imiser down to the remote branch office of larger organisations and meets the standalone requirements of smaller businesses. Combining the core functionality of Call Server, Trunk and Phone Modules within a single small enclosure, the 4140 Remote Call Server provides cost-effective support for up to 40 analogue or IP Phone users, delivering the very same advantage and benefits as larger **max**imiser systems.

Interfaces

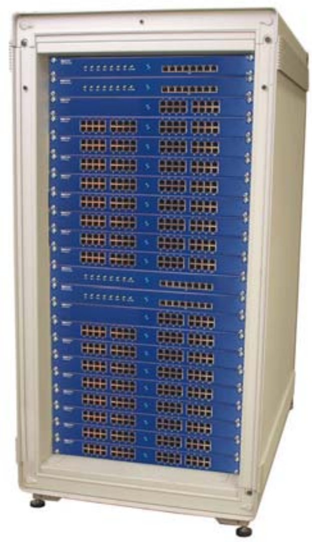
Supplied with 8 analogue extensions as standard, the 4140 Remote Call Server can be expanded by connecting its single 10/100 Mbps Ethernet port to an external LAN switch or network. This then allows further 4315/4330 Phone Modules, PCS 400/200/100s or 3rd Party IP Phones to be connected where higher densities of analogue or IP extensions are required. A total of 40 extensions, which can be any mix of analogue or IP, can be supported in this manner. Two BRI trunks (4 x 64 kbps B channels) are provided as standard on the 4140 Remote Call Server, however, a further two BRI trunks and a 15 channel PRI interface are also provisioned and can be enabled via license keys. All BRI and PRI interfaces can be individually configured for TE or NT operation, whilst the PRI interface can be used for ISDN or DPNSS connectivity. An X.21 IP WAN link running at speeds up to, and including, 2.048 Mbps, is also provided for scenarios where toll-saving, inter-site voice connectivity is required, but no IP WAN infrastructure (routers) currently exists. A pair of trigger inputs for external alarms and two outputs for door relays, as per the 4100 Call Server, completes the physical connections available on the 4140 Remote Call Server.

Small and Medium Sized Businesses

For smaller businesses that require a telephone system, the 4140 Remote Call Server provides all the familiar **max**imiser features and benefits within a single compact unit. From simple analogue PBX, through mobility, voice processing, IP Telephony and integration with web based IT applications - the 4140 Remote Call Server can support all these applications as it runs exactly the same maximiser software as its big brother the 4100 Call Server. An integrated, 8 port voicemail system with 30 active voicemail boxes is supplied as standard, as is forwarding/copying voicemail to email and the ability to generate SMS voicemail alerts (via an appropriate service provider).

The Branch Office

For larger organisations looking to utilise their existing IP infrastructure and bandwidth for voice communications, or roll out integrated desktop applications to even its smallest remote office, the 4140 Remote Call Server provides the most cost-effective means of doing so. Through the provision of a single unit that supports smaller number of extensions and trunks than the 4100 Call Server - but indentical features and facilities - the 4140 Remote Call Server provides the perfect "small of large" offering in extended maximiser networks.

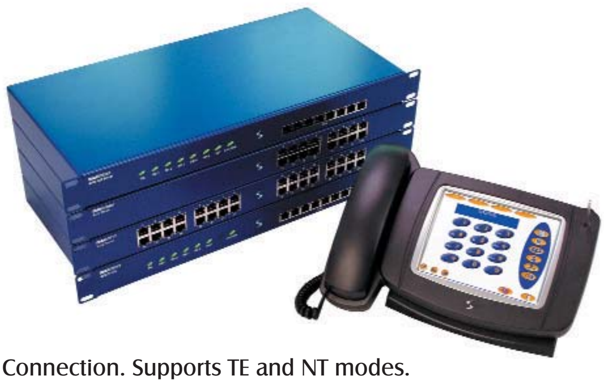


The **max**imiser - Start Small, Think Large

A single 4140 Remote Call Server provides all you need to deploy a simple, yet sophisticated business telephone system with eight extensions and four trunk lines, scaling up to 40 analogue or IP extensions and 15 trunks. The 4100 Call Server allows your system to scale up to 300 extensions, which can again be any mix of analogue or IP, and 190 trunk lines, by simply adding 4200 Trunk Modules, 4315/4330 Phone Modules and LAN Switches any where on your LAN network. Multiple 4100 Call Servers can be "clustered" to form a single system, meeting higher density telephony requirements and/or eliminate any single point of failure in business critical environments. In fact, the maximiser architecture supports up to 5,000 extensions, 3,800 trunks and 100 Call Servers/Remote Call Servers distributed across LAN and/or WAN networks within a single system. Yet from a customer, employee and network managers viewpoint, no matter how many Call Servers/Remote Call Servers are deployed, or where they may be located, the system appears, and just as importantly, is configured and managed, as one, from anywhere, via a simple, platform independent browser interface.

SpliceCom - Understanding Your Business

In the world of Business Telephony, SpliceCom are unique. Being a UK company ourselves, we fully understand your needs - because we share them with you. We have developed the **max**imiser and Proactive Communication Station, our sole products, accordingly. Between us, the SpliceCom team have enjoyed many years of major success in developing, launching, selling and supporting voice and data products for companies that include Case Communications, Racal Milgo, Bay Networks (now Nortel Networks), 3Com, Network Alchemy, Rocom, SDX Business Systems, Lucent Technologies and Avaya. SpliceCom's management team have proven track records in the leadership of several successful technology companies, developing marketing leading products and managing profitable growth having sold over 35,000 telephone systems which remain in everyday use. You can find out more about SpliceCom, the **max**imiser and the Proactive Communication Station by visiting www.splicecom.com



Technical Specifications

4100 Call Server

Interfaces

Quad Basic Rate ISDN S/T: 4 x RJ45 ETSI S/T interfaces CTR3 for Pan European Connection. Supports TE and NT modes.
Primary Rate ISDN S/T (30 Channels): 1 x RJ45 ETSI interface CTR4 for Pan European Connection. Supports DPNSS, TE and NT modes. Enabled by license.
WAN: 1 x 15 pin D-type (V.11, for point-to-point data connections). Operates at speeds up to and including 2.048 Mbps. Self Clocking mode supported.
Powered LAN Switch: Layer 2 Ethernet switch. 8 x RJ45 ports, dual speed, 10/100 Mbps FDX Ethernet interface with integral LEDs for Link and Data. Supports 802.3af for power over Ethernet. (All ports auto-sense for MDI/MDIX connectivity).
External Output: 2 x 3.5mm jack sockets to drive two external door release relays.
External Input: 2 x Mini-DIN sockets to accept inputs from external alarm systems.
Power: 1 x 48Vdc power jack. 1 x 48Vdc powered Ethernet input. See below.
Protocols PPP, Multi-Link PPP, MLCP, PAP, CHAP, DHCP, IP, TCP, IPCP, RTP, RTCP, LDAP, H.323, H.450, G.711, G.729*, HTML, PHP, DNS, TAPI 2.2 (3.0), 802.3af, DiffServ, 802.1p, DPNSS

* - Requires optional Voice Compression Card

4140 Remote Call Server

Interfaces

Analogue: 8 x RJ45 extensions. Supports standard analogue (POTS) telephones, fax machines and modems. Integral LEDs for Off-Hook and Ringing. BT SIN 227/242 support for Caller Display. 44V rms nominal Ringing Voltage. 48V nominal On-Hook voltage. 0-24V Off-Hook voltage. 25mA Off-Hook current.
Quad Basic Rate ISDN S/T: 4 x RJ45 ETSI S/T interfaces CTR3 for Pan European Connection. Supports TE and NT modes. 2 x Basic Rate ISDN interfaces enabled by license.
Fractional Primary Rate ISDN S/T (15 Channels): 1 x RJ45 ETSI interface CTR4 for Pan European Connection. Supports DPNSS, TE and NT modes. Enabled by license.
WAN: 1 x 15 pin D-type (V.11, for point-to-point data connections). Operates at speeds up to and including 2.048 Mbps. Self Clocking mode supported.
LAN: 1 x RJ45 port, dual speed, 10/100 Mbps FDX Ethernet interface
External Output: 2 x 3.5mm jack sockets to drive two external door release relays.
External Input: 2 x Mini-DIN sockets to accept inputs from external alarm systems.
Power: 1 x 48Vdc power jack. See below.
Protocols PPP, Multi-Link PPP, MLCP,PAP, CHAP, DHCP, IP, TCP, IPCP, RTP, RTCP, LDAP, H.323, H.450, G.711, G.729a*, HTML, PHP, DNS, TAPI 2.2 (3.0), 802.3af, DiffServ, 802.1p, DPNSS

* - Requires optional Voice Compression Card

4200 Trunk Module

Interfaces

Quad Basic Rate ISDN S/T: 4 x RJ45 ETSI S/T interfaces CTR3 for Pan European Connection. Supports TE and NT modes.
Primary Rate ISDN S/T (30 Channels): 1 x RJ45 ETSI interface CTR4 for Pan European Connection. Supports TE and NT modes.
WAN: 1 x 15 pin D-type (V.11, for point-to-point data connections). Operates at speeds up to and including 2.048 Mbps. Self Clocking mode supported.
Powered LAN Switch: Layer 2 Ethernet switch. 8 x RJ45 ports, dual speed, 10/100 Mbps FDX Ethernet interface with integral LEDs for Link and Data. Supports 802.3af for power over Ethernet. (All ports auto-sense for MDI/MDIX connectivity).
Power: 1 x 48Vdc power jack. See below.
Protocols PPP, Multi-Link PPP, MLCP, PAP, CHAP, IP, TCP, IPCP, RTP, RTCP, H.323 (Client), H.450, TAPI 2.2 (3.0), 802.3af, DiffServ, 802.1p, DPNSS

4315/4330 Phone Module

Interfaces

Analogue: 15 (4315) or 30 (4330) x RJ45 extensions. Supports standard analogue (POTS) telephones, fax machines and modems. Integral LEDs for Off-Hook and Ringing. BT SIN 227/242 support for Caller Display. 44V rms nominal Ringing Voltage. 48V nominal On-Hook voltage. 0-24V Off-Hook voltage. 25mA Off-Hook current.
LAN: 1 x RJ45 port, dual speed, 10/100 Mbps FDX Ethernet interface with integral LEDs for Link and Data.
Power: 1 x 48Vdc Power jack. See below.
Protocols: IP, IPCP, TCP, RTP, RTCP, H.323 (Client), H.450, TAPI 2.2 (3.0), DiffServ, 802.1p

4400 Voice Compression Module

Interfaces

LAN: 1 x RJ45 port, dual speed, 10/100 Mbps FDX Ethernet interface with integral LEDs for Link and Data.
Power: 1 x 48Vdc Power jack. See below.
Protocols: G.729a

Module Power

Primary: 48Vdc, 1A. Lump-in-Line PSU, CE Safety Approved. Pan-European variant.
Powered Ethernet (supplied as standard on 4100 Call Server, optional on 4200 Trunk Module): 48Vdc, 2.7A. Lump-in-Line PSU, CE Safety Approved. Pan-European variant.

Module Dimensions (mm)

486.80 (w) x 224.00 (d) x 42.23 (h)

Module Weight

4100 Call Server: 3.55 Kg
4140 Remote Call Server: 3.54 Kg
4200 Trunk Module: 3.41 Kg
4315 Phone Module: 3.42 Kg
4330 Phone Module: 3.48 Kg
4400 Voice Compression Module: 3.09 Kg

SpliceCom Limited The Hall Business Centre, Berry Lane, Chorleywood, Hertfordshire WD3 5EX
Tel: 01923 287700 Fax: 01923 287722 Email: info@splicecom.com Website: www.splicecom.com