Protocol & Port Information for the deployment of

maximiser

and

PCS proactive communication station

within IP Networks

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Introduction

SpliceCom have specifically developed **max**imiser and the Proactive Communication Station desktop devices to allow voice to be deployed as an “overlay’ application on IP LAN & WAN infrastructures. The **max**imiser architecture allows for three different implementation scenarios; traditional PBX replacement, separate voice & data LANs and fully converged voice & data networking. All three implementation methods can utilise standard Ethernet/IP LAN infrastructure equipment and components – Category 5/5e/6 Cabling & Structured Cabling Systems, Layer 2/3 Switches and Routers – to provide connectivity between Call Servers, Trunk Modules, Phone Modules, Voice Compression Modules & IP Phones (Proactive Communication Stations and 3rd Party H.323 & SIP IP Phones). This guide provides the pre-requisite background information required when deploying these **max**imiser components across an IP infrastructure.

maximiser

SpliceCom’s PBX currently supports up to 5,000 extensions which can be any mix of analogue or IP terminals. Constructed around a scalable and modular architecture all systems are constructed from just four module types; Call Server, Trunk, Analogue Phone and Voice Compression. These modules are linked via an IP LAN and WAN infrastructure allowing both the modules, and the users of the telephone system, to be geographically independent, i.e. located anywhere throughout that companies’ premises where their IP network exists.

The **max**imiser system is constructed around and utilises the following open, industry standard, protocols to communicate;

TCP, IP & IPCP

All **max**imiser communications, inter-module and module to end-point, utilise the Transmission Control Protocol, Internet Protocol & Internet Protocol Control Protocol.

LDAP

The operation, configuration and active management of **max**imiser systems is centered around a single LDAP database. Each Call Server holds a replicated copy of this database with any active changes being passed between all units, ensuring integrity of the information, whilst also allowing remote-site survivability in the event of IP network failure. This database can be read and written to by external applications allowing changes of operation to be made on the fly.

H.323

maximiser utilises H.323 V4 for its core IP Telephony/VoIP operation. The Call Server supports Gateway and Gatekeeper functionality, whilst the Phone and Trunk Modules both function as Gateways. In addition to SpliceCom’s PCS 400 and PCS 50 IP Softphone, any 3rd party H.323 IP Terminal can be utilised with the system. Voicemail is also terminated as a H.323 call.

H.323 uses TCP ports 1718 (Gatekeeper Discovery), 1719 (Gatekeeper RAS) and 1720 (Call Setup).

H.450

Supplementary telephony services are supported via H.450. Specifically these are;

- Call Transfer (H.450.2)
- Call Hold (H.450.4)

- Call Park & Pickup (H.450.5)
- Message Waiting Indication (H.450.7)

SIP
Version 3.1 for **max**imiser introduced support for an integral Session Initiation Protocol (SIP) Proxy Server, allowing SIP end-points and trunks to be supported concurrently with H.323 devices and WAN services. The following RFC's are currently supported by **max**imiser;

- SIP Version 2 (RFC 3261)
- Simple Call Transfer – Refer Method (RFC 3515)
- Message Waiting Indication (RFC 3842)
- Session Timers (RFC 4028)
- RTP Payload for DTMF Digits (RFC 2833)
- HTTP Authentication: Basic and Digest Access Authentication (RFC 2617)
- URL's for Telephone Calls (RFC 2806)

SIP uses TCP port 5060.

HTML
maximiser is managed via a platform independent, standards based browser. The active pages that are presented on the PCS 400 and PCS 50 applications are also HTML based, allowing information from WWW, Intranet or Web-enabled applications to be "pushed' to the screen of the phone based on a wide range of events. For example presented on the calling number or number called, time of day/day of week or from an external alarm or trigger source.

PHP
Every **max**imiser Call Server features an integrated Apache Web Server to run the telephony application interface for the PCS 400 and PCS 50. This allows customised web pages offering embedded functionality to be added using PHP scripting, enabling a tailored and bespoke solution to be delivered to meet specific customer needs. **max**imiser utilises PHP for all database configuration tasks and LDAP to interact.

VoiceXML
SpliceCom's Enhanced Speech Processing (ESP) application can be deployed as a valued added supplement to the standard voicemail service and has been developed using VoiceXML. Running on **max**imiser Call Servers or a standalone Linux platform, the use of VoiceXML within ESP allows for the development of Interactive Voice Response (IVR) applications that can integrate with and utilise existing business applications and data.

TAPI
Although we view HTML as our primary method of integration with IT applications, support for TAPI allows CTI support for those applications that have yet to be web enabled, or where an existing investment in TAPI has already been made. A TAPI Service Provider Interface (TSPI) is supplied with **max**imiser as standard and supports TAPI 2.2, allowing applications adhering to 2.2 and 3.0 to be utilised on Windows 98 platforms and above.

IP Multicasting
maximiser utilises the following multicast ports to pass information between Call Servers.

- Gatekeeper Registration Requests 224.0.1.41 Port 1718
- BLF Info (Park Slots, Active Phones, etc.) 224.0.1.41 Port 1717

• Paging Announcements	224.0.1.41	Ports Various
• Hold Music	224.0.1.51	Port 16640 - Default (User Configurable)

Quality of Service (QoS)
DiffServ provides the primary QoS mechanism for **max**imiser based systems to preserve voice quality in converged IP networks. **max**imiser marks the voice IP datagrams it generates so that routers and switches can give these packets priority over data datagrams. This is important when using slower or congested links to avoid voice degradation when data is very active. This is achieved by;

- Setting the SERVICE TYPE of every voice IP datagram to the value of 0xA0 (10100000)
- When using the DIFFERENTIATED SERVICES interpretation of the SERVICE TYPE, the CODEPOINT (DSCP) is set to 40
- When using the older TYPE OF SERVICE (TOS) interpretation, the PRECEDENCE is 5, D is 0, T is 0, and R is 0

Please note that the voice path is direct from IP endpoint to IP endpoint (phone to phone) and does not need to be routed via the Call Server. Thus PCS400/100/50/Phone Module/Trunk Module/Call Server may be the source/destination of the media stream (voice path). This is important if you are deciding routing priorities by IP address.

RTP/RTCP
The Real Time Protocol & Real Time Control Protocol are used to transport the media stream (voice path) directly between endpoints, once H.323 and H.450 have been used to establish the call. Wherever possible we will place our outgoing RTP and RTCP ports in the range 6000 to 6999.

G.711/G.729a
maximiser supports G.711 encoding with echo suppression as standard for transporting voice. This is a Pulse Code Modulation of voice frequencies (PCM), where 3.1 kHz analogue audio is encoded into a 64 kbps stream. Where compressed voice is a requirement, G.729a between Call Servers is supported through optional internal Voice Compression Cards, or rack mounted, Voice Compression Modules. This is based around a speech codec that toll quality audio encoded into an 8 kbps stream using the AS-CELP method. Annex A is a reduced complexity codec.

802.3af
maximiser supports Power over Ethernet (PoE) utilising the 802.3af standard. The 4100 Call Server is supplied with a PoE PSU as standard providing 8, powered, LAN switch ports and delivers a total of 130 Watts The same PoE PSU is available as a cost option for the 4200 Trunk Module. The PCS 400 supports 802.3af PoE and draws 15 Watts of power.

DHCP
The Dynamic Host Configuration Protocol is used to issue IP addresses to all **max**imiser components.

On power up each 4100 Call Server/4140 Remote Call Server will act as a DHCP Server using an address range of 192.168.0.1 (which will be taken as its own address) to 192.168.0.250 **unless** another DHCP server has been detected. This is the default mode of operation.

Configurable alternatives are;

- DHCP Enable (Default)
    - The Call Server will act as the DHCP server using the IP Address range specified in the DHCP Start Address and DHCP End Address fields.
- DHCP Disable
    - DHCP is disabled; the Call Server will use the IP address entered in the IP Address.
    - **(This is the recommended option if DHCP is being provided from another vendors DHCP Server)**
- DHCP Client
    - The Call Server will act as the DHCP client obtaining an IP address from a DHCP Server on the network.

When using a Call Server as a DHCP Server in its default configuration, 192.168.0.250 will be the first IP address to be given out, followed by 192.168.0.249 etc.

If this Call Server is the only one on the network it will become the Primary Call Server. However, if a Call Server already exists on the network and subsequent Call Servers are then introduced, the new Call Servers will attempt to join the original system. A Call Server must be disconnected from the network during power-up if this is not required as **max**imiser modules will always accept the first address they are given on power-up - they do not check for "best /acceptable options or server.

maximiser provides the option of setting static IP addresses for Phone, Trunk and Voice Compression Modules. The PCS 400 and PCS 100 still need to obtain their addresses from a DHCP Server.

When using 3rd party DHCP servers, we would generally recommend the use of a static address for the Call Server/Remote Call Server, using the local NT/2000/XP Windows server to provide reserved addressed for the Trunk Modules, Phone Modules and PCS 400s. The only DHCP options that must be provided are IP address, net mask and default router. However, when upgrading the software in the Phone/Trunk Modules the Next Server field (SIADDR) must also be completed. On a Microsoft DHCP server setting options 66 (Call Server's IP address) and 67 (File Name e.g. dummy.txt [**max**imiser is given but ignores this file name]) will force it to fill in the SIADDR correctly. Please note that BOTH these fields must be entered for the SIADDR to be set correctly. This approach allows the IP address of **max**imiser components to remain constant. This is especially important for Phone Modules where PCS 50s are deployed in Partner Mode.

DNS
maximiser supports Domain Name Services and can operate as Domain Name Server if required. Call Server IP Address Port 53.

PPP, Multi-Link PPP, MLCP
The IP Router integrated within the **max**imiser architecture, utilises the Point-to-Point Protocol, Multilink Point-to-Point Protocol, and Multilink Control Protocol to establish communications via the X.21 IP WAN and ISDN BRI/PRI interfaces. This is for IP

communications between Call Servers, Trunk Modules and/or 3rd Party Routers. Static routes are utilised.

PAP, CHAP
For security across WAN links the Password Authentication Protocol and Challenge Handshake Authentication Protocol are both supported.

PPTP
maximiser supports the Point to Point Protocol (PPTP) to allow secure Virtual Private Network (VPN) tunnels to be established between PCS 400s/PCS 100s and Call Servers/Remote Call Servers, or other networking equipment that can terminate PPTP. PPTP uses TCP Port 1723 and as it also uses the Generic Routing Encapsulation (GRE) protocol, IP protocol 47 must also be allowed through a firewall for a PPTP tunnel to be set-up.

maximiser **Specific Protocols**
In addition to the standard protocols documented above, there are areas of telephony where no standards exist. To address these areas we have implemented our own protocols, utilising the standard TCP/IP framework. These are as follows;

- Call Logging Spec. Call Server IP Address TCP Port 4001
  - Provides information on call completion for Call Logging, Call Billing and Call Management applications
- Call Routing Control Spec. Call Server IP Address TCP Port 4005
  - Allows the routing of Departmental calls to be controlled by an external device or application
- PCS Partnering Spec. PCS 400/100/50 IP Address TCP Port 5001
- PCS Partnering Spec. Phone Module IP Address TCP Port 50xx
  (Where xx is the physical port number of the Phone Module)
  - Allows PCS telephony control functionality to be directly embedded within 3rd party software applications

These interfaces are documented and can be provided to software developers and interested parties on request.

maximiser **TCP Port Summary**

TCP Port No.	IP Address	Description
53	Call Server	DNS
1640	224.0.1.51	Hold Music (Default)
1717	224.0.1.41	BLF Info
1718	224.0.1.41	Gatekeeper Registration Requests
1719		Gatekeeper RAS
1720		H.323 Call Set Up
1723		PPTP
Various	220.0.1.41	Paging Announcements
4001	Call Server	Call Logging Output
4005	Call Server	Call Routing Control
5001	PCS 400/100/50	PCS Partner Protocol (PCS 400/100/50)
50xx	Phone Module	PCS Partner Protocol (Phone Module)
5060		SIP

6000-6999	PCS 400/100/50/Phone Module	H.245
6000-6999	PCS 400/100/50/Phone Module	RTCP
6000-6999	PCS 400/100/50/Phone Module	RTP Media Stream

We strongly recommend that the Call Server or Remote Call Server is fronted by a firewall whenever it is connected to the public internet. Only the port for the specific function that you wish to expose should be allowed to pass through.

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