

**Protocol & Port Information  
for the Deployment of  
Splice.com SelectVoice  
Systems within IP Networks**

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## Introduction

Splicecom have specifically developed their comprehensive range of soft, hard and virtual PBXs, SelectVoice, Proactive Communication Station desktop devices and the Vision business management application suite to allow voice to be deployed as an “overlay” application on IP LAN & WAN infrastructures.

The SelectVoice 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 Soft PBX/Call Servers, Network Service Gateways, Trunk Modules, Phone Modules, IP Phones (Proactive Communication Stations and 3rd Party SIP Phones), Vision server and SSL Gateway. This guide provides the pre-requisite background information required when deploying these Splicecom system components across an IP infrastructure.

## SelectVoice

Splicecom's soft, hard and virtual PBXs currently support up to 100,000 extensions, which can be any mix of mobile, IP or analogue terminals. Constructed around a scalable and modular architecture all systems are constructed from a maximum of four component types; Soft PBX/Call Server, Network Service Gateway, Trunk and Analogue Phone. These modules are linked via an IP LAN and WAN infrastructure allowing both the components, and users of the telephone system, to be geographically independent, i.e. located anywhere throughout that companies' premises where their IP network exists or can be reached.

The SelectVoice system is constructed around and utilise's the following open, industry standard, protocols to communicate:

## TCP, IP & IPCP

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

## LDAP/LDAPS

The operation, configuration and active management of SelectVoice systems is centered around a single LDAP database. Each Soft PBX/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. The database is of Splicecom's own design. It runs entirely in memory, is optimised for speed and is accessed via TCP Port 4000 for LDAP and TCP Port 4100 for Secure LDAP<sup>1</sup>

## H. 323

SelectVoice utilises H.323 V4 for its core IP Telephony/VoIP operation. The S716/S8000 Soft PBXs and 5 and 4 Series Call Servers supports Gateway and Gatekeeper functionality, whilst the 5 and 4 Series Phone Modules and Trunk Modules all function as Gateways – as does the Network Service Gateway. In addition to Splicecom's wide range of PCS IP Phones and Softphones, any 3rdparty H.323 IP Terminal can be utilised with the system. Voicemail is also terminated as a H.323 call.

H.323 uses UDP ports 1718 (Gatekeeper Discovery), 1719 (Gatekeeper RAS) and 1720 (Call Setup) H.323 also uses dynamic port ranges from 32768 to 61000.

## H. 450

Supplementary telephony services are supported via H.450. Specifically, these are:

H.450.2	Call Transfer
H.450.4	Call Hold
H.450.5	Call Park & Pickup
H.450.7	Message Waiting Indication

NOTE:

<sup>1</sup>Secure LDAP is only used over the Public Internet when using STUN with SIP devices.

## SIP

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, is also available with SelectVoice.

The following RFC's are currently supported by SelectVoice:

RFC 3261	SIP Version 2
RFC 3515	Simple Call Transfer – Refer Method
RFC 3842	Message Waiting Indication
RFC 4028	Session Timers
RFC 2833	RTP Payload for DTMF Digits
RFC 2617	HTTP Authentication: Basic and Digest Access Authentication
RFC 2806	URL's for Telephone Calls , SIP uses TCP port 5060.

## HTML/ HTTP

SelectVoice is managed via a platform independent, standards based browser utilising HTML & HTTP. The active pages that are presented on the PCS 58xG/410/400 and PCS 60/50 applications are also HTML based, allowing information from WWW, Intranet or Web-enabled applications to be "pushed" to the screen of the phone or app 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. The default port for HTTP is TCP Port 80.

## PHP

Every SelectVoice based Soft PBX and Call Server utilises an integrated Apache Web Server to run the telephony application interface for the PCS 58xG/410/400 and PCS 60/50. 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. SelectVoice utilises PHP for all database configuration tasks and LDAP to interact.

## VoiceXML

Splice.com'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 the embedded SelectVoice Voice Processing app or a standalone Linux or Mac OS X 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/PHP 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 SelectVoice 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 Multicast

SelectVoice utilises the following multicast ports to pass information between Soft PBXs/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 (Default)	224.0.1.51	Port 16640
Hold Music (User Configurable)	224.0.1.51	Ports Various

## Quality of Service (QoS)

DiffServ provides the primary QoS mechanism for SelectVoice based systems to preserve voice quality in converged IP networks. SelectVoice 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;

- 1) Setting the SERVICE TYPE of every voice IP datagram to the value of 0xA0 (10100000)
- 2) When using the DIFFERENTIATED SERVICES interpretation of the SERVICE TYPE, the CODEPOINT (DSCP) is set to 46 (SelectVoice OS v3.2.1562 and later)
- 3) When using the DIFFERENTIATED SERVICES interpretation of the SERVICE TYPE, the CODEPOINT (DSCP) is set to 40 (SelectVoice OS v3.2.1561 or earlier)
- 4) 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 PCS IP Phones & Softphones/Phone Module/Call Server/Trunk Module 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

SelectVoice 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 Soft PBX/Call Servers is supported through the addition of a licence Key (S716/S800 and 5 Series) or optional internal Voice Compression Cards, or rack mounted, Voice Compression Modules (4 Series). 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

Splicecom's 5/4 Series hardware and PCS IP Phones support Power over Ethernet (PoE) utilising the 802.3af standard. The 5100 & 5108 Call Servers each support four PoE switched LAN ports as standard and deliver 20 Watts – as do the 5215 & 5330 Phone Modules. The 4100 Call Server is supplied with a separate PoE PSU as standard providing 8, powered, LAN switch ports and delivers a total of 120 Watts. The same PoE PSU is available as a cost option for the 4200 Trunk Module. All Splicecom PCS IP Phones support 802.3af as standard, power ratings are shown in the table below;

### Splicecom IP Phone Power over Ethernet Power Classifications

Splicecom IP Phone	PoE Class	Power Requirement
PCS 542	Class 1	Up to 4 Watts
PCS 552	Class 1	Up to 4 Watts
PCS 553	Class 1	Up to 4 Watts
PCS 552G	Class 1	Up to 4 Watts
PCS 553G	Class 1	Up to 4 Watts
PCS 561/562	Class 1	Up to 4 Watts
PCS 563	Class 1	Up to 4 Watts
PCS 571G/572G	Class 2	Up to 7 Watts
PCS 573G	Class 2	Up to 7 Watts
PCS 582G	Class 3	Up to 15 Watts
PCS 582GX	Class 3	Up to 15 Watts

### Yealink IP Phone Power over Ethernet Power Classifications

Yealink IP Phone	PoE Class	Power Requirement
T19 E2	Class 2	Up to 7 Watts
T40G	Class 2	Up to 7 Watts
T41S	Class 2	Up to 7 Watts
T42S	Class 2	Up to 7 Watts
T46S	Class 3	Up to 15 Watts
T48S	Class 3	Up to 15 Watts
T49G	N/A - External PSU required	N/A
T52S	Class 3	Up to 15 Watts
T54S	Class 3	Up to 15 Watts
T56A	Class 3	Up to 15 Watts
T58A/T58V	Class 3	Up to 15 Watts
W52P	Class 3	Up to 15 Watts
W56P	Class 3	Up to 15 Watts



Yealink IP Phone	PoE Class	Power Requirement
W60P	Class 1	Up to 4 Watts
CP860	Class 1	Up to 4 Watts
CP960	Class 1	Up to 4 Watts

## Splicecom Intelligent Gateway Module (IGW) Power over Ethernet Power Classifications

Splicecom IGW	PoE Class	Power Requirement
I208 – Basic Rate ISDN	Class 1	Up to 4 Watts
I230 – Primary Rate ISDN	Class 1	Up to 4 Watts
P308 – Analogue Phone	Class 3	Up to 15 Watts

## DHCP

The Dynamic Host Configuration Protocol is used to issue IP addresses to all SelectVoice components.

On power up each Soft PBX/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 Soft PBX/Call Server will act as the DHCP client obtaining an IP address from a DHCP Server on the network.

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

SelectVoice provides the option of setting static IP addresses for Phone, Trunk and Voice Compression Modules. The PCS IP Phones 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 Soft PBX/Call Server, using the local Windows server to provide reserved addresses for the Trunk Modules, Phone Modules, Compression Modules and PCS IP Phones. The only DHCP options that must be provided are IP address, net mask and default router. However, when upgrading the software in the

Phone and/or 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 [SelectVoice 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 Splicecom system components to remain constant. This is especially important for Phone Modules where Navigate or PCS 60/50s are deployed in Partner Mode.

## DNS

SelectVoice 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 SelectVoice 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 (4 Series only) and ISDN BRI/PRI interfaces. This is for IP communications between Call Servers, Trunk Modules (4 Series only) 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

SelectVoice supports the Point-to-Point Protocol (PPTP) to allow secure Virtual Private Network (VPN) tunnels to be established between PCS IP Phones and Soft PBX/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.

## SelectVoice 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 Call Server IP Address - TCP Port 4001

Provides information on call completion for Call Logging, Call Billing and Call Management applications.

Database Replication Call Server IP Address - TCP Port 4002

- Used by the SelectVoice to maintain the LDAP databases between systems

Busy Lamp Field Call Server IP Address - TCP Port 4004 224.0.1.41 UDP Port 1717

- Passes Ringing/Busy Extensions & Park Slot info between Call Servers in a multicast format.

Call Routing Control Call Server IP Address - TCP Port 4005

- Allows the routing of Departmental calls to be controlled by an external device or application

Conference Control Call Server IP Address - TCP Port 4011

- Allows Meet-Me-Conference features to be controlled by an external device or application

PCS Partner PCS IP Phone Address - TCP Port 5001

PCS Partner 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 external interfaces are all documented and can be provided to software developers and interested parties upon request.

## SelectVoice Port Summary

Port Number	IP Address	UDP/TCP	Description
20	Soft PBX/Call Server	TCP	FTP
21	Soft PBX/Call Server	TCP	FTP
22	Soft PBX/Call Server	TCP	SSH
23	Call Server	TCP	Telnet
47		TCP	PPTP Tunnel Setup
53	Soft PBX/Call Server	Both	DNS
80	Soft PBX/Call Server	TCP	Web Management (HTTP)
80	SSL Update Server	TCP	SSL Phone Software Update
123	All Devices	Both	NTP
443	Soft PBX/Call Server	TCP	Web Management (HTTPS)
443	S8000 Licence Server	TCP	Licence Validation (HTTPS) <a href="https://validate.Splicecom.com">https://validate.Splicecom.com</a>
873		TCP	Rsync
1717	224.0.1.41	UDP	Busy Lamp Field Information
1718	224.0.1.41	UDP	Gatekeeper Discovery
1719		UDP	Gatekeeper Registration
1720	Soft PBX/Call Server	TCP	H.323 Call Setup
1723		TCP	PPTP
3478	STUN	Both	STUN
4000	Soft PBX/Call Server	TCP	LDAP
4001	Soft PBX/Call Server	TCP	Call Logging
4002	Soft PBX/Call Server	TCP	Database Replication
4003	Soft PBX/Call Server	UDP	Splicecom inter-module use
4004	Soft PBX/Call Server	Both	Busy Lamp Field Information
4005	Soft PBX/Call Server	TCP	Call Routing Control
4006	Soft PBX/Call Server	TCP	Splicecom inter-module use
4007	Soft PBX/Call Server	TCP	Splicecom inter-module use
4008	Soft PBX/Call Server	TCP	Splicecom inter-module use
4009	Soft PBX/Call Server	TCP	Splicecom inter-module use
4010	Soft PBX/Call Server	TCP	Splicecom inter-module use
4011	Soft PBX/Call Server	TCP	Conference Control Interface
4018	Soft PBX/Call Server	TCP	Centralised Partner

4100	Soft PBX/Call Server	TCP	Secure LDAP (LDAPS)
5000	SSL Gateway	TCP	Default SSL Port for PCS devices
5001 - 5xxx	SSL Gateway	TCP	Custom SSL Port for PCS devices
5001	PCS 5xx Phones	TCP	PCS Partner Protocol
5001	PCS60/Navigate	TCP	PCS Partner Protocol
5001 - 5030	Phone Module	TCP	PCS Partner Protocol ports 1 to 30
5001 - 5030	PCS60/Navigate	TCP	Local PC needs to be able to use these ports to communicate with PCS devices
5060	Call Setup	Both	SIP
5061	Call Setup	TCP	Secure SIP (SIPS)
5901	MAP/S8000/S716	Both	VNC Remote Control
6600 - 6899	Call Server	UDP	RTP Media Stream
6000 - 6299	Phone Module	UDP	RTP Media Stream
6000 - 6299	PCS 5xx Phones	UDP	RTP Media Stream
6600 - 6899	Navigate Softphone	UDP	RTP Media Stream
6600-6899	Voicemail	UDP	RTP Media Stream (4.0, SV1.0 and SV1.1 variant's only)
10000-14000	Voicemail	UDP	RTP Media Stream (4.1.2, SV1.2 variant's and above)
6600 - 7859	Soft PBX	UDP	RTP Media Stream (4.0, SV1.0 and SV1.1 variant's only)
6900 - 8899	Soft PBX	UDP	RTP Media Stream (4.1.2, SV1.2 variants and above)
8081	license.splicecom.com	TCP	Download Licences
16384 - 16403	224.0.1.41	UDP	Paging Announcements
16640	224.0.1.51	UDP	Hold Music (Default)
16642 - 16702	224.0.1.51	UDP	Hold Music via an Analogue Phone module 16640 + (2 x port number)
21050	Navigate UC/CRM/Lync	TCP	Client Port to Server Port
21059	Navigate UC/CRM/Lync	UDP	Client Detect Port
49152	224.0.0.71	UDP	TDM Sync
32768 - 61000	H323	TCP	H323 Dynamic Port Range

**Note:** Splicecom strongly recommend that their Soft or Virtual PBX must be fronted by a firewall whenever it is connected to the public internet. Only the port for the specific function(s) that you wish to expose should be allowed to pass through.

## Vision

Vision is Splicecom's modular Business Management app and comprises of modules providing historical reports, call recording, live wallboards/dashboards, call centre and personal mobility applications. The Vision server uses a sub-set of the protocols and ports utilised by SelectVoice as previously documented. These are as Port 80 (HTTP), Port 4000 (LDAP) and Port 4005 (Call Routing Control). In addition, FTP is used to synchronise the SelectVoice and Vision databases, requiring the use of Ports 20/21 (FTP) and Port 4002 (Database sync).

## SSL Gateway

The SSL Gateway (formally iPCS Gateway) manages secure SSL connectivity between soft, hard and virtual PBXs running SelectVoice and PCS 5x1/5x2/5x3 IP Phones, Apple iOS iPhones/iPads and Android Smart Phones and Tablets utilising Splicecom's iPCS Softphone app. If an internet address needs to be assigned to the SSL Gateway machine ensure that the router has the relevant port forwarding enabled. The port that needs forwarding will be the one that has been configured in the iPCS 5x1/5x2/5x3 config and/or PCS config files.

The default port is currently 5000. Internally, the SSL Gateway uses UDP ports in the range 6600-6899, to pass audio to other parts of the SelectVoice system.

https port 443 to [fcm.googleapis.com](https://fcm.googleapis.com) is also used for Android iPCS push notifications.

## Quick Reference Tables

### ISIP Ports

Port	Transport	Description
5060	TCP/UDP	SIP Signaling Port
6000-6299	UDP	RTP Media Stream
443	TCP	Updates and Provisioning

### Yealink Phone Ports (STUN)

Port	Transport	Description
80	TCP	Yealink Configuration uploads
443	TCP	HTTPS
4100	TCP/UDP	Secure LDAP (LDAPS)
3478	TCP/UDP	STUN
5061	TCP/UDP	Secure SIP (SIPS)

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Ports & Protocols



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