Release Notes
SIP Application
SoundPoint® / SoundStation® IP

Version 1.5.2
19 April 2005

Part Number 3804-11530-152

Table of Contents

1. General

These release notes apply to version 1.5.2 of the SoundPoint / SoundStation IP SIP application. For more information, refer to the documents listed in § 4 Reference Documents.

1.1 System Requirements

Platform	BootROM version
SoundPoint IP 300	2.6.1 or greater
SoundPoint IP 301	2.6.1 or greater
SoundPoint IP 500	2.6.1 or greater
SoundPoint IP 501	2.6.1 or greater
SoundPoint IP 600	2.6.1 or greater
SoundPoint IP 601	3.1.0 or greater
SoundStation IP 4000	2.6.1 or greater

2. Changes

2.1 Version 1.5.2

2.1.1 Added or Changed Features

- 11356: **Changed configuration of presence and instant messaging features to be disabled by default**
- 11552: **Added phone UI and web interface configuration support for lineKeys and callsPerLineKey**

2.1.2 Removed Features

- 11816: **Pressing a line key will no longer terminate a call**

2.1.3 Corrections

The following issues have been resolved with this release:

- 9491: **Empty "to" header may be sent in some cases**
- 9776: **Parsing errors when dealing with the override file**
- 9817: **Configuration override file gets unnecessary extra parameters**
- 11189: **User can corrupt the directory by editing it when “presence” feature is disabled**

- 11343: **Pressing handsfree or headset button activates handset if handset is off hook**
- 11409: **Provisioning may not work reliably with the proftpd FTP server on Linux**
- 11417: **Phone may not be able to boot from a remote subnet**
- 11426: **Secondary dial tone plays incorrectly on certain digit maps**
- 11466, 11558: **Provisioning may fail using HTTPS if a custom certificate is used**
- 11556: **Stored authentication key from a SAS-VP server is deleted when the phone is reset to factory defaults**
- 11558: **Provisioning may fail using HTTPS if a custom certificate is used**
- 11575: **SoundPoint IP300/301 doesn't give warning message if duplicate IP is detected by DHCP client**
- 11584: **Automatic key repeats do not work**
- 11595: **Phone displays URL encoded digits when dialing**
- 11599: **Check-sync and polled configuration change features do not work**
- 11600: **Phone ignores maximum password length parameters**
- 11608: **Disabling "presence" feature does not remove it from phone's menu**
- 11609: **Disabling “messaging” feature on SoundStation IP 4000 and SoundPoint IP30x disables voice message feature as well**
- 11612: **When Do Not Disturb per-registration is enabled, the Do Not Disturb “clear all” soft key is missing**
- 11616: **CANCEL requests include tag when they shouldn't**
- 11633: **Phone should use flash credentials when boot server URL lacks them**
- 11641: **Phone shows an error message on the display when Hold is invoked on the last available call appearance**
- 11644: **Join does not work from the last available call appearance**
- 11665: **Pressing the headset button in ringing state does not answer call when headset memory is enabled**
- 11685: **Line configuration cannot be changed using web server**
- 11739: **A call can be lost when Split is used under certain circumstances**
- 11760: **Custom certificate gets corrupted if SAS-VP is used**
- 11788: **Pressing "New Call" soft key auto dials the previous number entered using on-hook dialing if the previous call failed**
- 11789: **The "more" soft key for establishing a conference can disappear, hiding the “Join” soft key**
- 11798: **There is an incompatability when using EPSV with proftpd**

2.1.4 Configuration File Parameter Changes

.cfg File	Action	Parameter	Description
sip	changed	feature.1.enabled, feature.2.enabled changed from 1 to 0	Presence and Instant Messaging are disabled by default.
sip	changed	voIpProt.server.x.transport	Explicitly set default to DNSnaptr
phone1	changed	reg.x.server.y.transport	Explicitly set default to DNSnaptr

2.2 Version 1.5.1

2.2.1 Added or Changed Features

- 966: **A single call will always show up in the first call appearance position**
- 1509: **Improved menu hierarchy**
- 1842: **Added visual "status" to contacts assigned to speed dial bins**
- 3924: **Added conference feature enhancement to "join" calls in progress**
- 7204: **Added flashing time/date until successful SNTP response**
- 7663: **Added ability to specify boot server address as URL per RFC 1738**
  This requires bootROM 3.0 or greater.
- 7894: **Added support for having more than one line key associated with the same SIP identity**
  This includes a new feature – pressing and holding down the line key provides call information about a call which is on hold on that line key.
- 7899: **Added support for the application to provision its own configuration files**
- 7900: **Added application support for HTTP and HTTPS boot server transport**
  This requires bootROM 3.0 or greater.
  For HTTPS, if the time on the phone is wrong the SSL certificate may be rejected. Configure SNTP to obtain an accurate time.
- 8055: **Added support for SAS-VP v2 management**
  This requires bootROM 3.0 or greater.
- 8521: **Added a menu entry to format the file system**
- 8786: **Added display of name and number on incoming caller ID**
- 9053: **Added support for displaying a useful CID when display name is uninformative**
- 9096: **Added customization options for SSL certificates**
- 9108, 10480: **Added support for SoundPoint IP 601 hardware platform**
- 9299: **Added allowing all files in <MAC>.cfg to be full URL's**
- 9323: **Removed requirement for at least two audio codecs to be configured**
- 9496: **Merged sip.cfg and ipmid.cfg configuration files into new sip.cfg file**

- 9548: **Added allowing user to disable time and date display**
- 9579: **Added allowing specific master configuration file to be specified in boot server URL**
- 9588: **Changed offering LED animation to continuous 2 Hz flash, rather than intermittent**
- 9659: **Added feature to split conferences and consultation calls into separate calls**
- 9675: **Added feature to allow conference initiation from call hold context**
- 9694: **Changed example directory file to no longer use silent ring type for contacts**
- 9710: **Changed default hold signaling to be the RFC 3261 style**
- 10806: **Added build ID to software revision stamps in User-Agent header**
- 11235: **Added support for arrow-key call-list shortcuts when phone is playing dial tone**

2.2.2 Removed Features

- 11973: **Removed support for port mode FTP server configurations**
  Use an FTP server/firewall that supports passive mode connections.

2.2.3 Corrections

The following issues have been resolved with this release:

- 737: **Phone will not accept IP packets bigger than 38,000 bytes**
- 2311: **Line labels do not line up with line keys on SoundPoint IP 600**
- 3707: **Can't use speed dial when one call already on Hold**
- 7952: **FTP transfers should remove partially written files in a failure scenario**
- 8050: **Parameters which were not changed are saved in configuration override file**
- 8333: **Improve source data for random device**
- 8416: **Bridged Line second call appearance is incorrect in specific scenario**
- 8616: **Incorrect message on display for incoming call on shared line on SoundPoint IP 4000**
- 8674: **Missing remote hold call appearance in specific Bridged Line scenario**
- 8755: **For TCP, the response to a request should try the remote port that sends the request first**
- 8771: **IP 4000 cannot download large directory file**
- 8801: **Phone ignores X-Syl-Line-ID and mixes call appearances**
- 8873: **When a new DHCP lease is obtained, the updated DNS information is not used**

- 8962: **Active Bridged Line cannot switch to incoming call**
- 9090: **Clock date menu choice ending in 'YYYY' not displayed properly**
- 9135: **Random string for CNONCE value for digest authentication should be limited to the base64 character set**
- 9187: **GMT offset and SNTP address set in flash are ignored if parameters exist in configuration file but have no associated value (i.e. are empty)**
- 9243: **Web server buttons not labeled, and some labels are incorrect**
- 9326: **DST not working for Southern Hemisphere**
- 9452: **DTMF tones not recognized by specific IVR after shared line remote resume**
- 9481: **Phone will attempt to download files indefinitely if connection to FTP server lost**
- 9482: **Phone waits for an error response from the FTP server when none is forthcoming**
- 9584: **Call duration missing from placed call list items on IP 4000**
- 9601: **DNS resolution fails when downloading [mac]-phone.cfg**
- 9735: **Web interface of SoundStation IP 4000 phone edits some non-IP 4000 parameters**
- 10825: **Phone should not collect digits after dial tone has timed out**
- 11285: **SIP authentication password stored in [mac]-phone.cfg file**
- 11303: **SoundPoint IP 300 phone loses contrast settings during reboot**
- 11348: **Large DHCP messages get truncated**
- 11350: **SoundStation IP 4000 phone can lock up when a key is pressed**
- 11458: **Audio loss on one leg of conference after second conference automatically put on Hold (first conference is Resumed)**
- 11516: **Off-by-one error when ringTypes are saved**
- 11548: **Cannot change administrator password or user password on SoundPoint IP 300**
- 11559: **ACD login does not work on SoundStation IP 4000**
- 11563: **ACD available/unavailable functions work differently on Bridged and Private lines**
- 11573: **Pressing Handsfree button does not put you back to handset when handset is off hook**

2.2.4 Configuration File Parameter Changes

.cfg File	Action	Parameter	Description
ipmid	removed	All parameters	The contents of this file have been added to sip.cfg and this file is no longer used.
sip	added	All “ipmid.cfg” parameters	The contents of the old ipmid.cfg file have been added to sip.cfg.
sip	added	call.callsPerLineKey	The number of calls or conferences which may be active or on hold per line key on the phone. For the IP 600, range is 1 to 24 and default is 24. For all other phones, range is 1 to 8 and default is 8.
sip	changed	voIpProt.SIP.useRFC2543hold	Changed the default value to "0" (it used to be “1”) which means that RFC 3261-style hold signalling is the default.
sip	changed	voice.gain.rx.analog.chassis.IP300 voice.gain.rx.analog.ringer.IP300	Changed name to voice.gain.rx.analog.chassis.IP_300 voice.gain.rx.analog.ringer.IP_300
sip	changed	call.shared.oneTouchResume	This applies to SoundStation IP 4000 phones only in this build. For all other phones, one-touch resume is the default. In order to view call information about a call on hold on another phone with a shared line – press and hold down the line key for a few seconds.
sip	changed	ind.gi.IP_600.x…	Changed values used for locating line key labels. Update this whole section.
sip	removed	ind.pattern.8.step.3 to 6	An incoming call causes the LED to flash continuously at 2Hz rather than flash intermittently.
sip	removed	all .obs parameters from logging section	These parameters are no longer used.
phone1	added	reg.x.ringType	The ring type for each registration can be configured. Range is 1 to 22. **Note**: ring type number 1 is “silent ring”.
phone1	added	reg.x.lineKeys	The number of line keys on the phone to be associated with registration ‘x’. Range is 1 to the maximum number of line keys on the phone (IP 300 = 2, IP 500 = 3, IP 600 = 6, IP 4000 = 1). Default is 1.
phone1	added	reg.x.callsPerLineKey	The number of calls or conferences which may be active or on hold per line key for a specific registration on the phone. This will override the global call.callsPerLineKey parameter in sip.cfg. Same range and defaults as call.callsPerLineKey above.
000000000000	removed	ipmid.cfg from list of CONFIG_FILES	The ipmid.cfg file is no longer used.

2.3 Version 1.4.1

2.3.1 Added or Changed Features

- 9017: **Added audio processing diagnostic information**
- 7967: **Added configurable '+' global prefix character to E.164 user parts in sip: URIs**

2.3.2 Removed Features

None.

2.3.3 Corrections

The following issues have been resolved with this release:

- 8282: **Application can get into reboot loop when polling configuration**
- 9146: **Speakerphone indicator turns off on second call to the same line**
- 9162, 9226, 9525: **Speakerphone indicator turns off when audio active in specific scenarios**
- 9169, 9398, 9399: **Audio performance improvements on IP 4000**
- 9240: **The phone responds with 482 Loop Detected when server responds to a challenge**
- 9313: **Bridged Line shows as active after call abandoned in some scenarios**
- 9374: **IP 4000 reboots if “Edit” softkey is pressed in Contact Directory when it is empty**
- 9385: **2833 payload type change during local conference establishment can result in no audio on two legs of conference**
- 9459: **Line LED stays flashing red toggling hold on a held call**
- 9503: **Volume adjustments from Audio Diagnostics display can put volume setting out of range until next reboot**
- 9599: **SIP User Agent header has incorrect text on IP 4000**

2.3.4 Configuration File Parameter Changes

None.

2.4 Version 1.4.0

2.4.1 Added or Changed Features

- 1259: **Added automatic periodic boot server poll for upgrade**
- 2402: **Added phone restart menu command**
- 5761: **Added support for IP 4000 hardware platform**

- 7175: **Made provisioning file cache more intelligent**
  When TFTP is used for provisioning, the names of the ".ld" and ".cfg" files on the server no longer have to change for the phone to recognize that the files have changed. This requires bootROM version 2.6.0 or greater.
- 7970: **Integrated transfer and conference with automated dialing / speed dial feature**
- 8232: **Added DHCP VLAN Discovery**
  This depends on having BootROM version 2.6.0 or greater
- 8304: **Local conference on IP 4000 only available if G.729 codec not enabled**
- 9118: **Added support for IP 301 hardware platform**

2.4.2 Removed Features

- 7055: **Removed bootROM date stamp from System Status / General menu**
- 8896: **Removed idle display microbrowser configuration**

2.4.3 Corrections

The following issues have been resolved with this release:

- 7003: **<MAC>-phone.cfg file can be saved as non-well-formed XML - excluded Unicode in attribute name**
- 7424: **Phone ignores short DHCP packets**
- 7838: **Incoming call to shared line which is remotely active cannot be answered**
- 7839: **While in a call, handsfree indicator turns off upon receiving Subscribe**
- 7850: **A Held Bridged Line Appearance indicates Idle in some scenarios**
- 7911: **Phone adds extra tabs into <MAC>-phone.cfg files**
- 7940: **DHCP Client does not send DHCPDECLINE**
- 7946: **Remotely active Bridged Line means line cannot be used to initiate a call**
- 7947: **A remote call cannot be retrieved if a second line is active**
- 8096: **Incorrect termination LED and icon display in some shared line scenarios**
- 8119: **Unnecessary NOTIFY's sent in some situations with Bridged Line Appearances**
- 8151: **The call index in signaling is wrong when resuming a Bridged Line**
- 8189: **Rapid hold, remote resume sequence on shared lines can put the phone into the wrong state**
- 8194: **When using the "offline" state, Presence can stop working for 30 minutes**

- 8426: **When in number dialing, the star key no longer toggles between * and ., and the pound key no longer toggles between # and @**
- 8447: **In some Bridged Line scenarios, IP 300 clears display while in a call**
- 8533: **Default configuration files should not contain an SNTP server**
- 8550: **Phone sends epid in the To header when it should not**
- 8569: **The CANCEL request should be sent to the destination of INVITE**
- 8682: **Phone uses some English soft key labels after a language change**
- 8718: **Phone does not handle forked INVITE properly**
- 8746: **P-Asserted-Identity in re-INVITE is not displayed**
- 8747: **Phone caches orbit number after pickup in some scenarios**
- 8772: **Web config DNS setting is wrong**
- 8999: **No audio after re-INVITE. (SDP problem).**
- 9000: **TLS in via has been changed to TCP.**
- 9011: **Phone should not reboot if polling fails due to FTP server unavailability**
- 9112: **Phone should send 415 and 488 indicate different case in DSP negotiation.**
- 9322: **IP 300 does not give warning when Network Link is down**

2.4.4 Configuration File Parameter Changes

.cfg File	Action	Parameter	Description
ipmid	added	voice.codecPref.IP_4000.G711Mu voice.codecPref.IP_4000.G711A voice.codecPref.IP_4000.G729AB	Codec preferences for IP 4000 platform. Note that the phone will host local conferences only if the G.729 codec is not included in the list of preferences i.e. voice.codecPref.IP_4000.G729AB = blank
ipmid	added	voice.gain.rx.analog.chassis.IP_4000 voice.gain.rx.analog.ringer.IP_4000 voice.gain.rx.digital.chassis.IP_4000 voice.gain.rx.digital.ringer.IP_4000 voice.gain.tx.analog.chassis.IP_4000 voice.gain.tx.digital.chassis.IP_4000 voice.aes.hf.duplexBalance.IP_4000.x voice.ns.hf.IP_4000.enable voice.ns.hf.IP_4000.signalAttn voice.ns.hf.IP_4000.silenceAttn voice.rxEq.hf.IP_4000.enable voice.rxEq.hf.IP_4000.type voice.rxEq.hf.IP_4000.size voice.rxEq.hf.coef.IP_4000.x voice.txEq.hf.IP_4000.enable voice.txEq.hf.coef.IP_4000.x	Parameters to support IP 4000 platform.

.cfg File	Action	Parameter	Description
ipmid	removed	tcpIpApp.sntp.resyncTime	The SNTP resynchronization now has an automatic retry mechanism which replaces this parameter.
ipmid	added	ind.class.2.state.34.index to ind.class.2.state.37.index	
ipmid	added	ind.gi.IP_4000.x.y	Indicators for IP 4000 platform.
ipmid	added	log.level.change.dns	Logging level control for DNS module.
ipmid	added	prov.polling.enabled prov.polling.mode, prov.polling.period, prov.polling.time	Controls for automatic periodic boot server poll for upgrade. prov.polling.enabled turns this on or off, default = 0. prov.polling.mode can be `rel` (relative) or `abs` (absolute), default = `abs`. prov.polling.period is the polling period in seconds, default = 86400, min. = 3600. Rounded up to the nearest number of days in `abs` mode. In `rel` mode measured relative to boot time. prov.polling.time is used only in `abs` mode. It is specified as hh:mm, default = 03:00.
ipmid	changed	attributes under the “bitmaps” tag	New IP 4000 bitmaps have been added. Update this whole section.
ipmid	changed	attributes under the “Animations” tag	New IP 4000 animations have been added. Update this whole section.
ipmid	removed	all .obs parameters in voice section	IP 400 platform no longer supported.
sip	added	voIpProt.SIP.WM50	0 means phone will support Windows Messenger 4.7, this is the default. 1 means phone will support Windows Messenger 5.0
sip	added	voIpProt.SIP.keepalive.sessionTimers	1 means enable session timer 0 means disable session timer, this is the default. When session timer is disabled, the phone will not declare "timer" in "Support" header in INVITE. The phone will still respond to the re-INVITE or UPDATE. The phone will not try to re-INVITE or do UPDATE even if remote endpoint asks for it.

3. Notes

3.1 Distribution Files

The following files constitute the 1.5.2 distribution of the SoundPoint / SoundStation IP SIP application. For centrally provisioned systems, copy these files to the boot server, maintaining the folder hierarchy present in the zip file.

Some of the configuration files must be modified. Refer to the *Administrator Guide* for details.

Files	Description
sip.ld	SIP application executable, App Version 1.5.2.0054
	IP 300 2345-11300-001: 1.5.2 23-Mar-05 17:06 IP 301 2345-11300-010: 1.5.2 23-Mar-05 16:45
	IP 500 2345-11500-001: 1.5.2 23-Mar-05 16:59 2345-11500-010: 1.5.2 23-Mar-05 16:59 2345-11500-030: 1.5.2 23-Mar-05 16:55 2345-11500-020: 1.5.2 23-Mar-05 17:02 2345-11500-040: 1.5.2 23-Mar-05 16:55
	IP 600 2345-11600-001: 1.5.2 23-Mar-05 16:42 IP 601 2345-11605-001: 1.5.2 23-Mar-05 16:42
	IP 4000 2201-06642-001: 1.5.2 23-Mar-05 16:51
sip.cfg	main core and SIP configuration file
phone1.cfg	example per-phone SIP configuration
000000000000.cfg	example master configuration file
000000000000-directory~.xml	example per-phone local contact directory XML file (edit and then remove '~' from name to seed phones which have no directory)
SoundPointIP-dictionary.xml	dictionary files for multilingual support included (no IP 300 support): Chinese, China (for IP 600, 601 and IP 4000 only) Danish, Denmark Dutch, Netherlands English, Canada English, United Kingdom English, United States French, France German, Germany Italian, Italy Japanese, Japan (for IP 600, 601 and IP 4000 only) Korean, Korea (for IP 600, 601 and IP 4000 only) Norwegian, Norway Portuguese, Portugal Russian, Russia Spanish, Spain Swedish, Sweden
SoundPointIPWelcome.wav	start up welcome sound effect

3.2 Upgrading

3.2.1 From Version 1.5.1 to 1.5.2

This section lists the changes that should be made to configuration files when using the centralized (boot server) provisioning model. For general guidelines, see the Updating and Rebooting information in Section 4.3 of the Administrator Guide.

3.2.1.1 Mandatory Changes

- **Presence and Instant Messaging Disabled by Default**
  These features have been disabled in sip.cfg by setting feature.1.enabled and feature.2.enabled to 0. If these features are required they must be enabled in sip.cfg.

3.2.2 From Version 1.4.1 or 1.4.2 to 1.5.1

This section lists the changes that should be made to configuration files when using the centralized (boot server) provisioning model. For general guidelines, see the Updating and Rebooting information in Section 4.3 of the Administrator Guide.

3.2.2.1 Mandatory Changes

- **ipmid.cfg File No Longer Used**
  All parameters which used to exist in the ipmid.cfg configuration file have been added into the sip.cfg file.
  Edit the master configuration file which lists the application file name and all configuration file names for each phone. Remove ipmid.cfg from the list of CONFIG_FILES. See 000000000000.cfg revision 1.13 for an example.
  Any customization which was done to parameters in ipmid.cfg should be done to parameters in the new combined sip.cfg revision 1.71.4.1.
  Make a backup copy of ipmid.cfg then delete it from the boot server.
- **Default Hold Signalling Changed**
  The default value of voIpProt.SIP.useRFC2543hold has changed from 1 to 0. If RFC 2543 hold signalling is still required, change this parameter value to 1.
- **Changed SoundPoint IP 300 and 301 Gain Parameter Names**
  The parameter names of voice.gain.rx.analog.chassis.IP300 and voice.gain.rx.analog.ringer.IP300 have been changed to voice.gain.rx.analog.chassis.IP_300 and voice.gain.rx.analog.ringer.IP_300. These changes will be automatically included when the latest sip.cfg file is used.
- **Line Label Locations Shifted for SoundPoint IP 600**
  The location of line key labels on the SoundPoint IP 600 have shifted slightly. These changes will be automatically included when the latest sip.cfg file is used.

3.2.2.2 Optional Changes

- **Setting the Number of Calls Per Line Key**
  The default configuration allows a maximum of 24 calls per line key on the SoundPoint IP 600/601, and a maximum of 8 calls per line key on all other phones. If this number must be changed, it can be set for all registrations (lines) at once by setting the value of call.callsPerLineKey in sip.cfg. If this number needs to be

different for different registrations, the reg.x.callsPerLineKey values in phone1.cfg can be changed.

- **Setting Individual Ringer Sound Effects Per Registration (Line)**
  The ringer sound effect can be customized for each registration on the phone. The new parameter reg.x.ringType is included in phone1.cfg. **Note that ring type number 1 is the "silent ring".**
- **Multiple Line Keys Per Registration (Line)**
  Each registration on the phone can have more than one line key assigned to it (except on the SoundStation IP 4000). The new parameter reg.x.lineKeys is included in phone1.cfg.

3.2.3 From Version 1.4.0 to 1.4.1

This section lists the changes that should be made to configuration files when using the centralized (boot server) provisioning model. For general guidelines, see the Updating and Rebooting information in Section 4.3 of the Administrator Guide.

3.2.3.1 Mandatory Changes

None

3.2.3.2 Optional Changes

- **Global '+' Prefix Support**
  Adding a global '+' prefix character to E.164 user parts in sip: URIs can be turned on by setting voIpProt.SIP.requestURI.E164.addGlobalPrefix to 1 in sip.cfg.

3.3 Outstanding Issues

The following issues will be fixed in a subsequent release.

- 4310: **No QoS support for signaling protocol (TCP)**
  Workaround: The default QOS parameters will still be used for TCP signaling packets, and these may be specified in the ipmid.cfg configuration file.
- 5085: **Cannot answer an incoming call while directory is being saved**
  Workaround: None.
- 7521: **Transfer from a shared line can be interrupted (other end of shared line can pick up call before Transfer completes)**
  Workaround: None.
- 8921: **Centralized conference fails due to RTP port slow to open in some cases**
  Workaround: None.
- 9292: **IP 4000 reboots upon downloading a wave file with a path containing '\' instead of '/'**
  Workaround: Wave file paths must be specified using '/' e.g. "wavs/ring1.wav"
- 11190: **Incorrect time zone is used for one to two minutes after a reboot**
  Workaround: Change the "GMT Offset" value configured in flash settings to match the GMT offset which will be sent by the SNTP server.
- 11510: **CDP changes after startup on IP 4000 not used**
  Workaround: None.
- 11916: **Logo and File Path corrupted during "Loading"**
  Workaround: The application filename is correctly displayed if it is 20 or fewer characters.

4. Reference Documents

- *Administrator Guide – SoundPoint IP SIP* – Version 1.5