PHOENIX Audio Technologies

Provisioning and configuring the SIP Spider

Administrator Guide

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1. Introduction

This document details the steps available to prepare the Spider SIP unit for usage. It explains how to prepare the unit through the unit's GUI (manual provisioning), or alternatively automatically using configuration files which are prepared ahead of time. The document includes description of the parameters available for the user in setting up the unit.

In order for the Spider to connect to the network it needs to be assigned with a number. The service provider, when assigning a number, provides you with a 'User ID', 'Authentication Name', which in many cases is similar to the user ID, and an 'Authentication Password'. In the case of hosted services you should also get the address of the SIP Proxy/Registrar.

Provisioning the Spider is the process in which this information is embedded in the unit, and the unit uses it to register with the service provider and operate like a phone.

The providing can be done by manually inserting the information through the unit's GUI (using web browser), or by uploading a configuration file that contains this information onto the unit.

All of the system's parameters, including dialing protocols, nickname, address book, and many others, can be inserted and controlled manually through the web GUI, or by editing this information in a form of a configuration file and uploading it to the system.

The "uploading" process can be initiated manually (through the web GUI) or automatically by scheduling a periodic uploading process that will be initiated by the unit on the scheduled times. The same concept applies to the unit's firmware – whenever a new software version is available it can be uploaded onto the unit manually, through the GUI, or scheduled to be done automatically and initiated by the unit itself.

Optionally - the units can be set up to search for the configuration file automatically upon the first power up using DHCP (dynamic host configuration protocol). The server, to which the units connect to, has to support this protocol, and the configuration file for this particular unit has to be present on the server carrying the proper address.

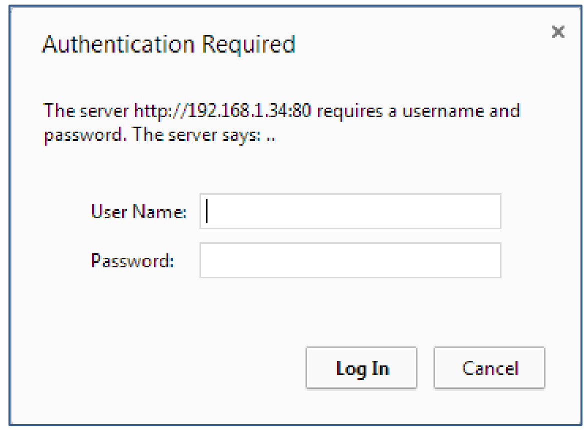
In this user guide we will describe the provisioning process, both manual and automatic, and discuss briefly the other parameters that can be controlled using the unit's GUI.

2. Manual provisioning

As a default - the Spider comes pre-loaded with a default configuration file. The user has to enter the registration information using the unit's Web GUI:

- Connect the unit to your network; the unit searches for an IP address and will notify you (on its display) when an IP is obtained. If you missed the address and it disappeared from the screen too quickly, simply click on the menu button and then 0.

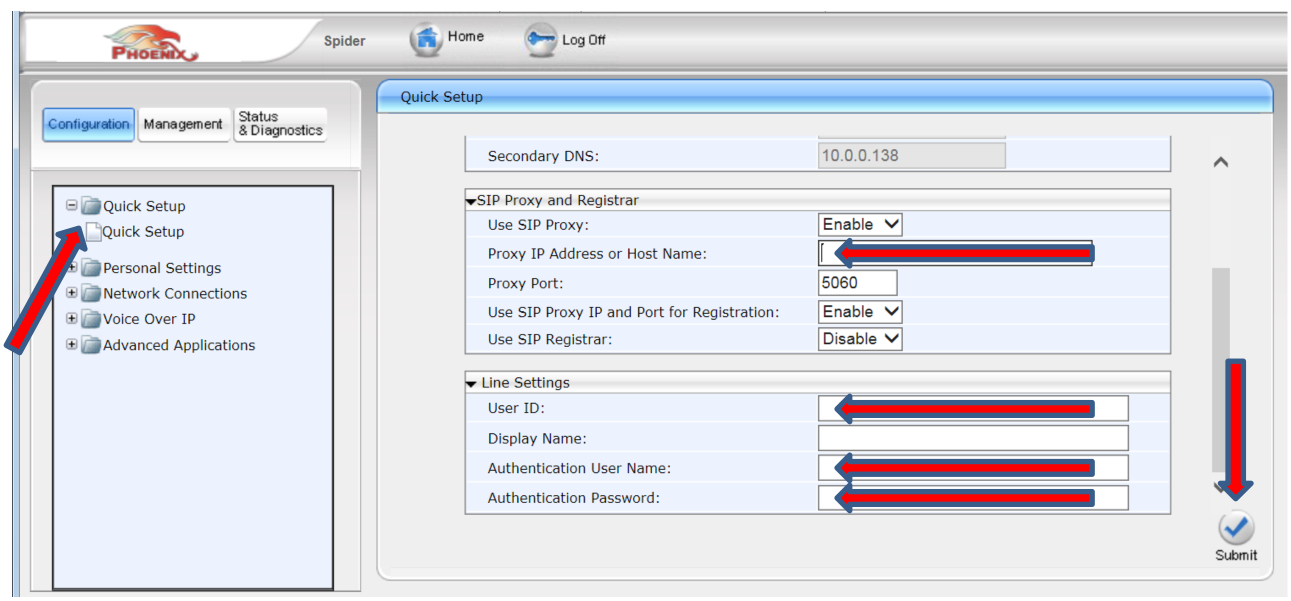
- From a computer which is connected to the same network as the phone, type the IP address on your browser's address window.
- When prompt with the opening screen of the GUI, type ***admin*** as the user name and ***1234*** as the password.



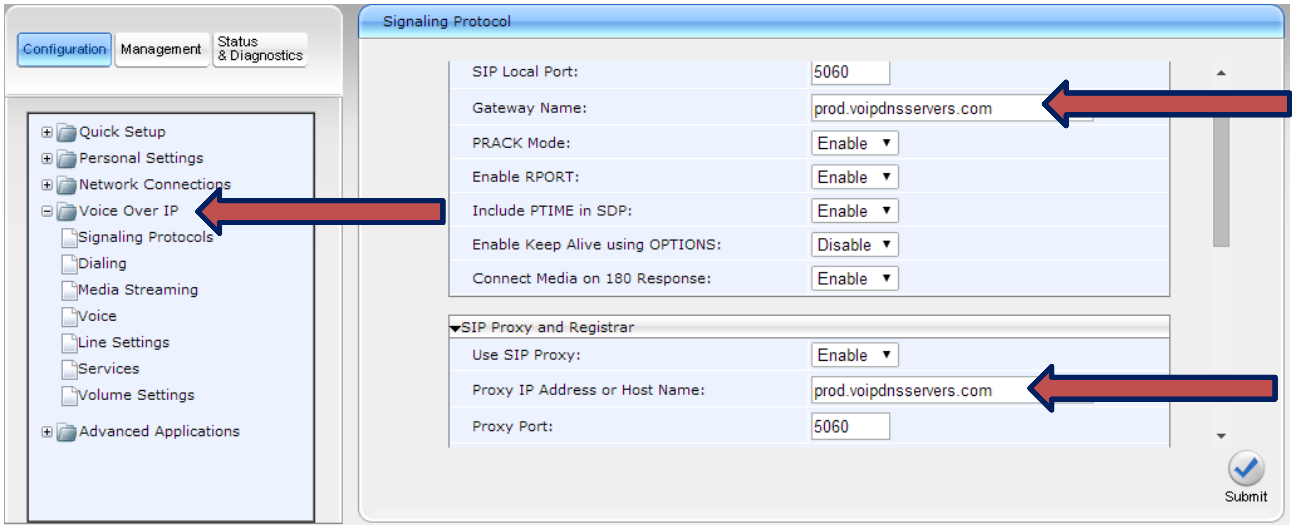
- Go to Quick Setup, enter the user ID, user name, password, and the SIP proxy address. This information is provided by the service provider when you sign up for the telephone line. You can set the text that will be displayed on the Spider's display while nit in a call.

- Click Submit.

- Wait until the unit is registered and start using it



In some cases (with some service providers) you may have to go to the Voice Over IP screen and insert the Proxy address in the ***Gateway Name*** and ***Proxy IP Address or Host Name*** fields.



3. Automatic provisioning

3.1 Concept

The Spider, as a default, is shipped set for manual provisioning. Upon request the unit can be shipped set for automatic provisioning. With automatic provisioning, the registration information (user name, authentication password, and other required information) is edited into the unit's configuration file. During the automatic provisioning the unit will search for its configuration file and upload it automatically.

The Spider's automatic provisioning supports two provisioning methods: Static and Dynamic.
With static provisioning, the configuration file is given an arbitrary name and placed on a server, the url of the file is defined using the GUI in the 'Configuration URL' parameter of the Management/Automatic Update menu.
With dynamic provisioning, the Spider use DHCP options to get the configuration file name. In this case, the configuration file name should be <Spider's MAC Address>.cfg and it should be defined on the DHCP server. Upon the initial power up, the unit will obtain an IP address then search for a file with a name that carries its MAC address and has an extension .cfg. This file should contain the registration information; it will be uploaded to the unit, it will register and be ready to function as a telephone.

3.2 Preparing the configuration file

The configuration file is a text file that contains many of the system's parameters. The Spider's current configuration can be saved to a configuration file via Management/Manual Update menu of the GUI. Using the GUI to define the parameters and saving them to a file is an efficient way to prepare a configuration file without knowing all the syntax rules. Alternatively, a default file with the Spider's default parameters can be downloaded from our website and then edited and adapted to the environment.

For multiple Spiders provisioning support, an ‘Include’ mechanism is added, enabling to split the configuration parameters to several files, one specific file for a specific Spider, which holds the registration information, and has an “include” directive to link to additional configuration files which hold parameters common to other Spiders .

Following is an example for such a specific configuration file:

```
[HW_TYPE]
type=MT505
[VOIP]
voip/line/0/id=<enter id here>
voip/line/0/description=<enter unit’s nickname here>
voip/line/0/auth_name=<enter authentication name here>
voip/line/0/auth_password=<enter authentication password here>
Include <url of configuration file with common parameters>
Include <url of phone book file>
```

The url’s protocol part of the configuration files can be one of FTP, TFTP, HTTP, or HTTPS.

3.3 The VoiProvision Tool

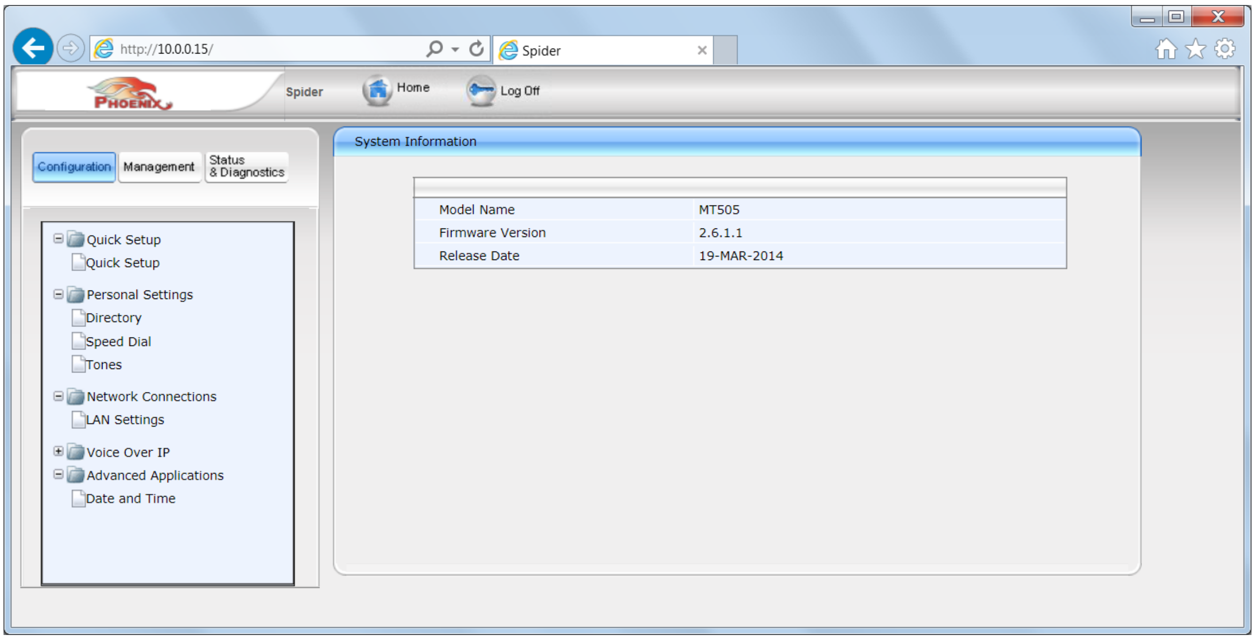
The VoiProvision tool is an application, that will be provided per request, that will take registration information (MAC address, user ID, user name, password, nickname) from a list stored in a single CSV file and prepare a configuration file for each of the MAC addresses in the list.

Please contact Phoenix for more information.

4. Setting system's parameters

Setting system's parameters is achieved using the web browser GUI. It can also be done through the configuration file. When many units need to be set with the same parameters, the user can set them up once using the GUI, saving the configuration file to a known location, and then uploading the same file to the other units changing the name of the file to contain the other units' MAC addresses.

The home screen of the GUI is shown here; it shows the firmware configuration version, the date it was released, it has three tabs on the upper left side – Configuration, Management, and Status and Diagnostics.

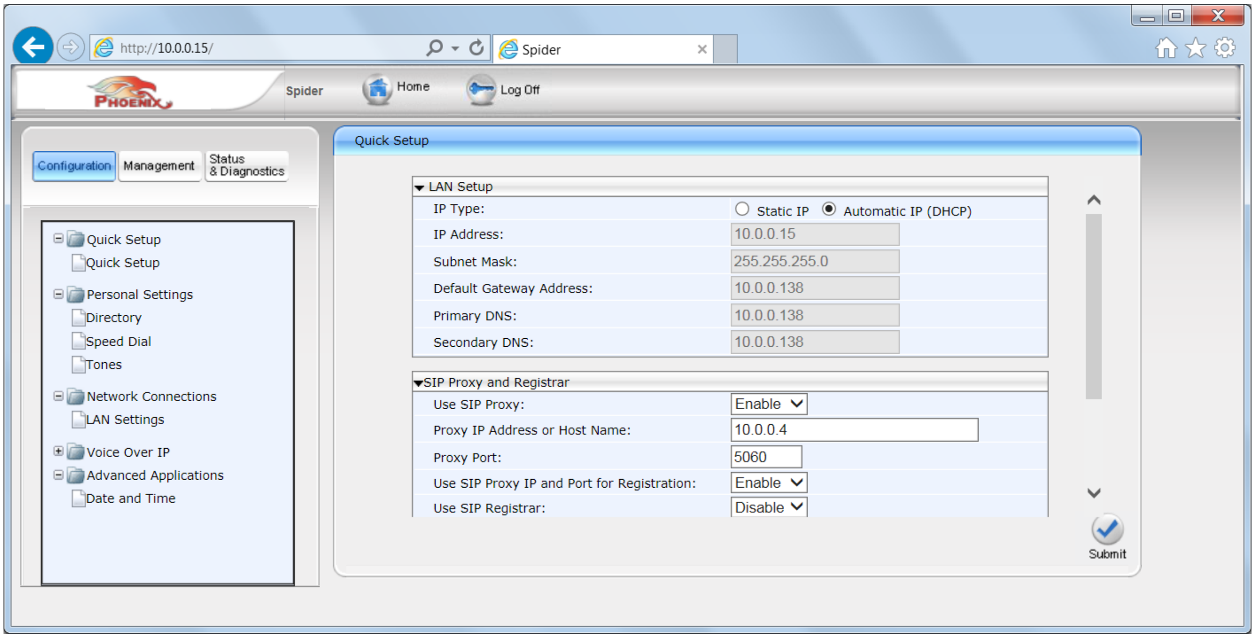


4.1 Configuration Parameters

The GUI will open with the Configuration tab selected. On the left side of the screen there are the menu items which include: Quick Setup, Personal Settings, Network Connections, Voice Over IP, and Advanced Applications.

4.1.1 Quick Setup

The Quick Setup screen is used for the provisioning process (as mentioned before).



The user can select a Static IP option or a DHCP Automatic IP address. If a unit is set with a static IP address, and the user, for some reason, does not know what it is, he can always reset the system to default back to the DHCP IP option through the unit's display.

The Quick Setup screen contains the following fields:

LAN Setup

IP Type {Static of dynamic – DHCP}
IP Address
Subnet Mask
Default Gateway Address
Primary DNS
Secondary DNS

Note: if automatic IP (DHCP) option is selected these data is determined by the DHCP server and cannot be changed by the user.

SIP Proxy and Registrar

Use SIP Proxy {default: Enables}
Proxy IP Address or Host Name {to be entered for proper provisioning}
Proxy Port {default 5060}
Use SIP Proxy IP and Port for Registration
Use SIP Registrar

Line Settings

Line Activate {default: enabled}
User ID {to be entered for proper provisioning}
Authentication User Name {to be entered for proper provisioning}

Authentication Password {to be entered for proper provisioning}

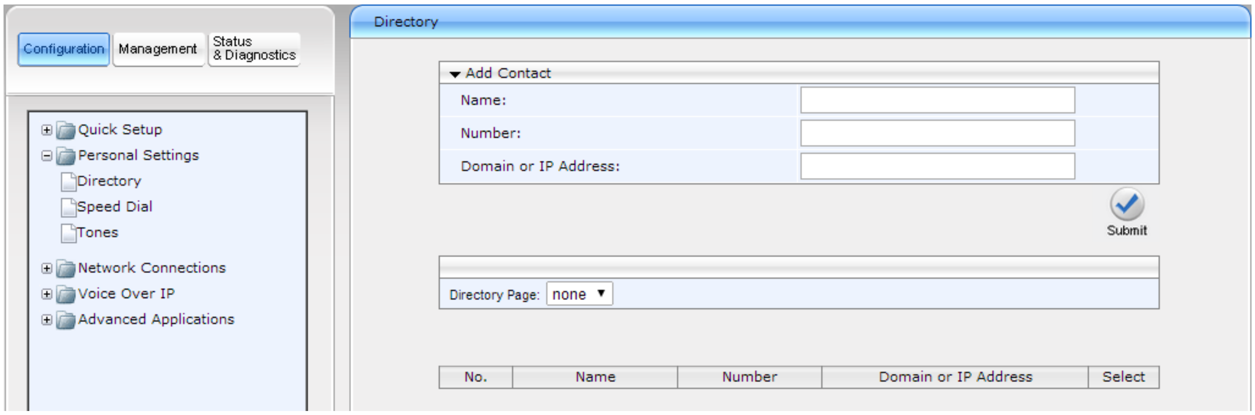
4.1.2 Personal Settings

The Personal Settings menu contains three items:

Directory – the user can enter up to 100 telephone numbers

Speed Dial – up to 10 numbers can be assigned a Speed Dial number (from 0 to 9)

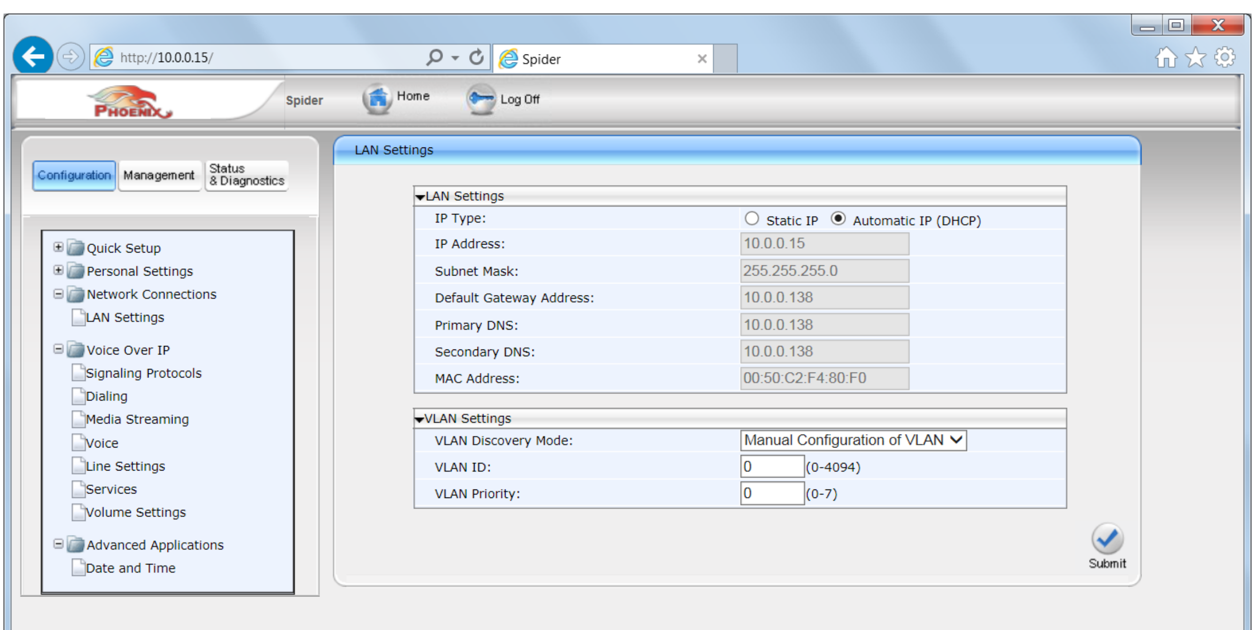
Tones – the user can select his current location so that the telephone will use the standard tones required for this particular location. The units has 11 optional locations; the default is the USA



4.1.3 Network Connections

Network Connections contains the IP information part (in the case of automatic IP the fields are greyed out and cannot be changed. In addition in contains the VLAN Setting, if the phone is supposed to work under VLAN environment (default is disabled). Note that if the phone is set for VLAN the user will not be able to log into the GUI (or get and IP for that matter) outside of the VLAN. The user can reset the unit to work with standard network and Dynamic IP by resetting it to default through the unit's built in menu.

4.1.4 Voice Over IP



The Voice Over IP menu controls most the SIP protocol parameters. It includes the following sub-menus: Signaling Protocols, Dialing, Media Streaming, Voice, Line Settings, Services, and Volume Settings. The table below details the parameters available under the VOIP submenu and their meaning. In most cases the user can use the default configuration file as provided without changing any of the parameters in this menu. In some cases these parameters should be set differently. Please consult your service carrier or our tech support team.

Parameter	Description
4.1.4.1 Voice Over IP -> Signaling Protocols	
Voice Over IP -> Signaling Protocols -> SIP General	
SIP Transport Protocol	Determines the transport layer for outgoing SIP calls initiated by the phone. ▪ UDP (default) ▪ TCP
SIP Local Port	Defines the local SIP port (UDP or TCP) for SIP messages. The valid range is 1024 to 65535. The default value is 5060.
Gateway Name	Assigns a name to the phone. The name is used as the host part of the SIP URI in the From header. **Notes:** ▪ Ensure that the name you choose is the one with which the Proxy is configured to identify the phone. ▪ If not specified, the phone's IP address is used (default).
PRACK Mode	Determines whether the phone sends PRACK (Provisional Acknowledgment) messages upon receipt of 1xx SIP reliable responses. ▪ Disable ▪ Enable (default)
Enable RPORT	Determines whether the phone adds the ‘rport’ parameter to the relevant SIP message (in the SIP Via header). ▪ Disable (default) ▪ Enable
Include PTIME in SDP	Determines whether the phone adds the PTIME parameter to the SDP message body. ▪ Disable (default) ▪ Enable
Enable Keep Alive using OPTIONS	Determines whether keep-alive is performed using SIP OPTIONS messages sent to the Proxy. ▪ Disable (default) ▪ Enable
Keep Alive Period	Defines the Proxy keep-alive time interval (in seconds) between Keep-Alive messages. The valid range is 0 to 86400. The default value is 300.

Connect Media on 180 Response	Determines whether the media is connected upon receipt of SIP 180, 183, or 200 messages. When the parameter is disabled, media is connected upon receipt of 183 and 200 messages only. ▪ Disable (default) ▪ Enable
Voice Over IP -> Signaling Protocols -> SIP Proxy and Registrar	
Use SIP Proxy	Determines whether to use a SIP Proxy server. ▪ Disable ▪ Enable (default)
Proxy IP Address or Host Name	The IP address or host name of the SIP proxy server.
Proxy Port	The UDP or TCP port of the SIP proxy server. The valid range is 1024 to 65535. The default value is 5060.
Maximum Number of Authentications Retries	The SIP proxy server registration timeout (in seconds). The valid range is 0 to 86400. The default value is 3600.
Use Proxy IP and Port for Registration	Determines whether to use the SIP proxy's IP address and port for registration. When enabled, there is no need to configure the address of the registrar separately. ▪ Disable (default) ▪ Enable
Use SIP Registrar	Determines whether the phone registers to a separate SIP Registrar server. ▪ Disable (default) ▪ Enable
Registration Expires	
Use SIP Outbound Proxy	Determines whether an outbound SIP proxy server is used (all SIP messages are sent to this server as the first hop). ▪ Disable (default) ▪ Enable
Outbound Proxy IP Address or Host Name	The IP address of the outbound proxy. If this parameter is set, all outgoing messages (including Registration messages) are sent to this Proxy according to the Stack behavior.
Outbound Proxy Port	The port on which the outbound proxy listens. The valid range is 1024 to 65535. The default value is 5060.
Use Redundant Proxy	Enables the redundant proxy mechanism. ▪ Disable (default) - Phone doesn't use redundant proxy (default). ▪ Enable - Phone registered to redundant proxy if the primary proxy does not respond to SIP signaling messages.

Redundant Proxy Address	The IP address or host name of the redundant proxy. The default value is 0.0.0.0. **Note:** This parameter is applicable only if the parameter **use_redundant_proxy** is set to 1.
Redundant Proxy Port	The UDP or TCP port of the redundant proxy server. The valid range is 1024 to 65535. The default value is 5060. **Note:** This parameter is applicable only if the parameter **use_redundant_proxy** is set to 1.
Redundant Proxy Keep Alive Period	Defines the interval in seconds for sending keep-alive messages to the proxy. The valid range is 0 to 300. The default value is 60. **Note**: This parameter is applicable only if the parameter **use_redundant_proxy** is set to 1.
Switch back to Primary SIP proxy when available	Determines the proxy redundancy mode. ▪ Asymmetric mode. (default) ▪ Symmetric mode. The Redundant Proxy feature allows the configuration of a backup SIP proxy server to increase QoS stability. Once this feature is enabled, the phone identifies cases where the primary proxy does not respond to SIP signaling messages. In these scenarios, the phone registers to the redundant proxy and the phone seamlessly continues normal functionality, without the user noticing any connectivity failure or malfunction with the primary proxy. The Redundant Proxy feature includes two operational modes: ▪ Asymmetric mode: the primary proxy is assigned a higher priority for registration than the redundant proxy. In the Asymmetric mode, once the phone is registered to the primary proxy, it sends keep-alive messages (using SIP OPTIONS messages) to the primary proxy. If the primary proxy does not respond, the phone registers to the redundant proxy, but continues sending keep-alive messages to the primary proxy**.** If the primary proxy responds to these keep-alive messages, the phone re-registers to the primary proxy. Therefore, the phone assigns the primary proxy a higher priority for registration. ▪ Symmetric mode: both proxies are assigned the same priority for registration. In the Symmetric mode, once the phone is registered to a proxy, it sends keep-alive messages to this proxy. The phone switches proxies only once the proxy to which it has registered does not respond. Therefore, the phone assigns both proxies the same priority for registration In both modes, the following applies: If the phone is not registered (i.e., if the proxy server–redundant or primary–to which the phone currently tries to register does not respond), the phone attempts to register to an alternative proxy. These attempts continue until the phone

	successfully registers. If this feature is enabled and the user reboots the phone, the phone registers to the last proxy to which it was trying to register, and not necessarily to the primary proxy. **Note:** This parameter is applicable only if the parameter use_redundant_proxy is set to 1.
Voice Over IP -> Signaling Protocols -> SIP Timers	
Retransmission Timer T1	The time interval (in msec) between the first transmission of a SIP message and the first retransmission of the same message (according to RFC 3261). The valid range is 100 to 60000. The default value is 500. **Note:** The time interval between subsequent retransmissions of the same SIP message starts with SipT1Rtx and is multiplied by two until SipT2Rtx. For example (assuming that SipT1Rtx = 500 and SipT2Rtx = 4000): ▪ The first retransmission is sent after 500 msec. ▪ The second retransmission is sent after 1000 (2*500) msec. ▪ The third retransmission is sent after 2000 (2*1000) msec. ▪ The fourth retransmission and subsequent retransmissions until SIPMaxRtx are sent after 4000 (2*2000) msec.
Retransmission Timer T2	The maximum interval (in msec) between retransmissions of SIP messages (according to RFC 3261). The valid range is 4000 to 60000. The default value is 4000. **Note:** The time interval between subsequent retransmissions of the same SIP message starts with SipT1Rtx and is multiplied by two until SipT2Rtx.
Retransmission Timer T4	The SIP T4 retransmission timer according to RFC 3261. The valid range is 5000 to 60000. The default value is 5000.
Invite Timer	The SIP INVITE timer according to RFC 3261. The valid range is 0 to 65535. The default value is 32000.
4.1.4.2 Voice Over IP -> Dialing	
Dialing Timeout	The duration (in seconds) of allowed inactivity between dialed digits. When you work with a proxy, the number you have dialed before the dialing process has timed out is sent to the proxy as the user ID to be called. This is useful for calling a remote party without creating a speed dial entry (assuming the remote party is registered with the proxy). The valid range is 0 to 10. The default value is 5.
Phone Number Length	The maximum number of digits that you can dial The valid range is 0 to 32. The default value is 15.
Dialing Complete Key	Defines the Dialing Complete key. Pressing the Dialing Complete key forces the phone to make a call to the dialed digits even if there is no match in the dial plan, digit map or phone book. ▪ Disable ▪ Enable (default) The pound (#) key and the SEND key are pre-selected as

	dialing complete keys.
Dial Tone Timeout	Defines the maximum duration of the dial tone (in seconds) after which the dial tone stops and a reorder tone is played. The valid range is 0 to 300. The default value is 30.
Reorder Tone Timeout	Defines the maximum duration of the reorder tone (in seconds) after which the reorder tone stops and a howler tone is played. The valid range is 0 to 300. The default value is 40.
No Answer Call Timeout	Defines the maximum duration that an outgoing call wait for answer before disconnecting. The default value is 40.
Howler Tone Timeout	Defines the duration (in seconds) of the howler tone. If the limit is exceeded, the howler tone stops. The howler tone indicates that the phone has been left in an off-hook state. The valid range is 0 to 300. The default value is 120.
DTMF Transport Mode	DTMF transport mode. ▪ Inband ▪ RFC 2833 (default) ▪ Via SIP
Digit Map	Enables the administrator to predefine possible formats (or patterns) for the dialed number. A match to one of the defined patterns terminates the dialed number. The valid value can be up to 256 characters. There are two main formats for the digit map configuration. The formats are distinguished by the separator ‘;’ or ’|’. ▪ **Using ‘|’ separator:** The following constructs can be used in each numbering scheme: ✓ **Digit:** A digit from "0" to "9". ✓ **DTMF:** A digit, or one of the symbols "A", "B", "C", "D", "#", or "*". Extensions may be defined. ✓ **Wildcard:** The symbol "x" which matches any digit ("0" to "9"). ✓ *** Range:** One or more DTMF symbols enclosed between square brackets ("[" and "]"). ✓ **Sub range:** Two digits separated by hyphen ("-") which matches any digit between and including the two. The subrange construct can only be used inside a range construct, i.e., between "[" and "]". ✓ **Position:** A period (".") which matches an arbitrary number, including zero, of occurrences of the preceding construct. **For example:** [2-9]11|0|100|101|011xxx.T|9011xxx.T|1[2-9]xxxxxxxxx|91[2-9]xxxxxxxxx|9[2-9]xxxxxx|*xx|[8]xxxx|[2-7]xxx This example includes the following rules: ✓ **[2-9]11:** 911 rule: 211, 311, 411, 511, 611, 711, 811, 911 are dialed immediately ✓ **0:** Local operator rule: After dialing "0" the phone waits T seconds and then completes the call automatically ✓ **100:** Auto-attendant default extension ✓ **101:** Voicemail default extension ✓ **011xxx.T:** International rule without prefix ✓ **9011xxx.T:** International rule with prefix

	✓ **1[2-9]xxxxxxxxx:** LD rule without prefix ✓ **91[2-9]xxxxxxxxx:** LD rule with prefix ✓ **9[2-9]xxxxxx:** Local call with prefix ✓ ***xx:** 2-digit star codes ✓ **[1-7]xx:** A regular 3 digit extension that does not start with 9 or 8 is dialed immediately ✓ **[2-7]xx:** A regular 3 digit extension that does not start with 9 or 8 or 1 is dialed immediately ✓ **[2-7]xxx:** A regular 4 digit extension that does not start with 9 or 8 or 1 is dialed immediately ✓ **[8]xxx:** A 3 digit extension prefixed with an 8 (routes calls directly to voicemail of extension xxx) ✓ **[8]xxxx:** A 4 digit extension prefixed with an 8 (routes calls directly to voicemail of extension xxxx) ✓ **T**: Refers to the **Dialing Timeout.** ▪ **Using ';' separator:** An 'x' in the pattern indicates any digit. ';' separates between patterns. For example: '10x;05xxxxxxxx;4xxx'. In this example, three patterns are defined. A number that starts with 10 is terminated after the third digit, and so on. If the user dials a number that does not match any pattern, the number is terminated using the timeout or when the user presses the pound ('#') key.
Dial Plan	This parameter works in conjunction with the **digit_map** parameter and enables translation of specific patterns to specific SIP destination addresses. An 'x' represents any dialed digit. Each backslash at the right side of the '=' represents one of the dialed digits. Rules are separated by the character ';'. The valid value can be up to 256 characters. For example: '4xxx=Line_\\\@10.1.2.3' This rule issues a call to 10.1.2.3 with the SIP ID of Line_ followed by the last three digits of the dialed number.
4.1.4.3 Voice Over IP -> Media Streaming	
RTP Port Range – contiguous Series of 4 Ports Starting From:	Defines the starting port range for Real Time Protocol (RTP) voice transport. The valid range is 1024 to 65535. The default value is 4000.
DTMF Relay RFC 2833 Payload Type:	Defines the RTP payload type used for RFC 2833 DTMF relay packets. The valid range is 96 to 127. The default value is 101.
Type of Service (ToS):	QoS in hexadecimal format. This is a part of the IP header that defines the type of routing service to tag outgoing voice packets originated from the phone. It informs routers that this packet must receive a specific QoS. The default value is **0xb8**. Values can be set in decimal (e.g. 184) or hexadecimal (e.g. 0xb8).
Codecs	Determines the codecs that are in use and their priority. Up to five codecs can be configured, where the first codec has the highest priority. To make a call, at least one codec must be configured. In addition, for best performance it is recommended to select as many codecs as possible. When a call to a remote party is started, the available codecs are compared with the remote party's to determine the codec to use. If there is no codec that both parties have made

	available, the call attempt fails. If more than one codec is common to both parties, the side that originate the call cannot force which of the common codecs are used by the remote party's client. If the side originating the call wishes to force the use of a specific codec, he should configure the list with only that specific codec.
Codec Type	Name of the codec. The first codec has the highest priority. The valid codec parameters are: ▪ G.722 ▪ G.711 A-Law ▪ G.711 Mu-Law ▪ G.729 ▪ G.723
Packetization Time	The length of the digital voice segment that each packet holds. The default is 20 millisecond packets, excluding G.723 which is 30 millisecond packets.
G.723 Bitrate	Low or high bit rate for G.723. ▪ Low ▪ High (default)
4.1.4.4 Voice Over IP -> Voice	
Jitter Buffer Minimum Delay	The initial and minimal delay of the adaptive jitter buffer mechanism, which compensates for network problems. The value should be set according to the expected average jitter in the network (in milliseconds). The valid range is 0 to 300. The default value is 35.
Jitter Buffer Optimization Factor	The adaptation rate of the jitter buffer mechanism. Higher values cause the jitter buffer to respond faster to increased network jitter. The valid range is 0 to 13. The default value is 7.
Enable Silence Compression	Enables silence compression for reducing network bandwidth consumption. ▪ Disable (default) ▪ Enable
4.1.4.5 Voice Over IP -> Line Settings	
User ID	Lines VoIP user's ID for identification to initiate and accept calls. The user's ID can be up to 30 characters.
Display Name	Arbitrary name to intuitively identify the line. This name is displayed on the LCD display when the device is in Idle, and is displayed to remote parties as your caller ID.
Authentication User Name	User name provided to you from the VoIP service provider. This is used when sending a response to Unauthorized or Proxy Authentication Requested (401/407). The authentication name can be up to 35 characters.
Authentication Password	Password provided to you from the VoIP Service Provider. This is used when sending a response to Unauthorized or Proxy Authentication Requested (401/407). The authentication password can be up to 35 characters.
4.1.4.6 Voice Over IP -> Services	

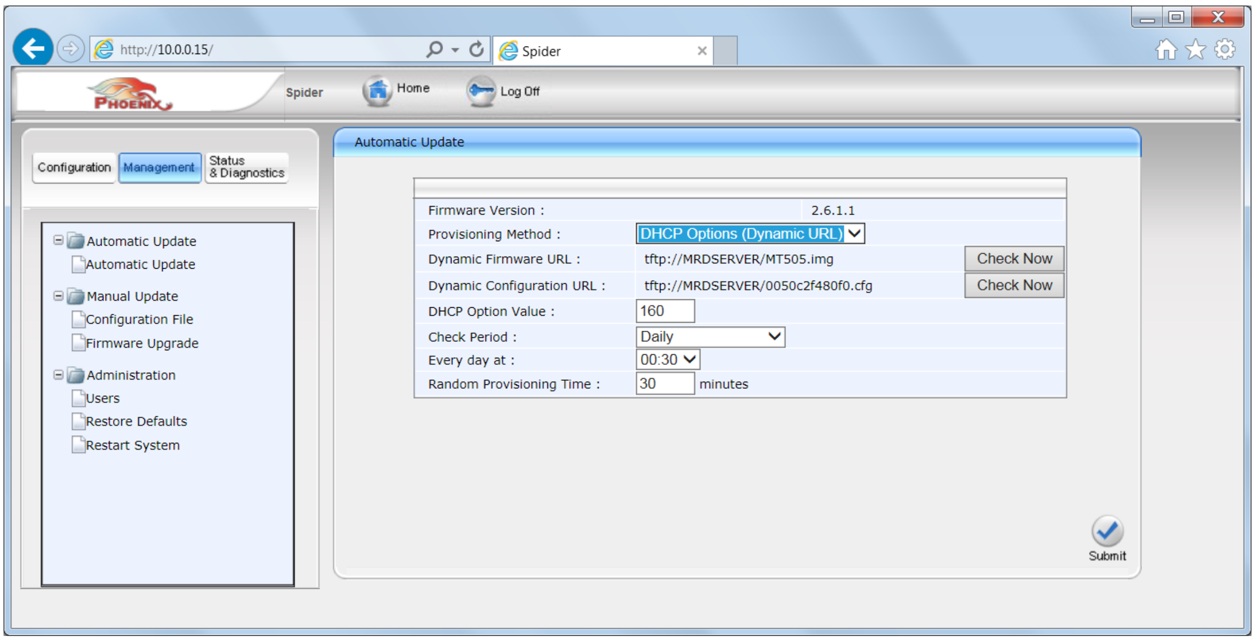
Call Waiting Activate	Enables the Call Waiting feature. ▪ Disable ▪ Enable (default)
Call Waiting SIP Reply	Determines the SIP response that is sent when another call arrives while a call is in progress: ▪ Ringing - 180 Ringing ▪ Queued (default) - 182 Queued
Call Forward	Not implemented
Voice Over IP -> Services->Message Waiting Indication (MWI)	
Activate	Enables the MWI feature. ▪ Disable ▪ Enable (default)
Voice Mail Number	Defines the extension number for accessing your voice mail messages. The valid value is up to 64 characters.
Subscribe to MWI	Determines whether the phone registers to an MWI server. ▪ Disable (default) ▪ Enable
MWI Server IP Address or Host Name	The IP address or host name of the MWI server. The default value is 0.0.0.0.
MWI Server Port	The port number of the MWI server. The valid range is 1024 to 65535. The default value is 5060.
MWI Subscribe Expiry Time	The interval between the MWI Subscribe messages. The valid range is 0 to 86400. The default value is 3600.
Out of Service Behavior	Determines whether a reorder tone is played instead of a dial tone if you configured a Registrar IP address and the registration failed. ▪ No Tone ▪ Reorder Tone (default)
4.1.4.7 Voice Over IP -> Volume Settings	
Ringer Volume	Ringing tone volume. This volume can be modified on-the-fly by pressing the phone’s VOLUME UP or VOLUME DOWN keys when the phone is in idle state, or when it rings for incoming call. The valid range is [0] to [-31] dB in steps of 1 dB, or Mute.
Speaker Volume	Speaker’s volume. This volume can be modified on-the-fly by pressing the phone’s VOLUME UP or VOLUME DOWN keys during a call, or while ringback indication is played. The valid range is [0] to [-46] dB in steps of 2 dB, or Mute.
Microphone Volume	The volume of the voice signal sent from the device to the other side during a call. The valid range is [0] to [-46] dB in steps of 2 dB, or Mute. The microphone can be Muted or unmuted on-the-fly by pressing the Spider’s MUTE button during a call. On call termination, the microphone’s mute state is reset to unmuted.

4.1.5 Advanced Applications

This menu items includes the parameters required to set the Spider's clock. If NTP is selected the time will be taken from the server, otherwise the clock can be set manually. The NTP parameters include selecting the primary server (from which the time will be taken), time zone, enabling the Daylight Saving Time, and the update time interval.

4.2 Management

The management tab includes parameters for the Automatic Update, Manual Update, and Administration



4.2.1 Automatic Update

Firmware Version: the SIP processor software version

Provisioning Method: can be set to Static (the user specifies the static address of the image file under the Firmware URL field), DHCP (the unit will get the image file using the DHCP protocol), or disable the automatic update altogether.

Firmware URL: the user will enter the static address of the image file

Configuration URL: the user defines the static address for the configuration file

Check Period: defined the intervals in which the unit will check for software update

Every day at: defines the time of the day in which the update will occur

Random Provisioning Time: the unit will check for update at a random time around the selected time; this prevents a situation in which many units try to download the updates files from the server at the same time.

The menu also includes “Check Now” buttons that allows the user to initiate an update check and download at any time

4.2.2 Manual Update

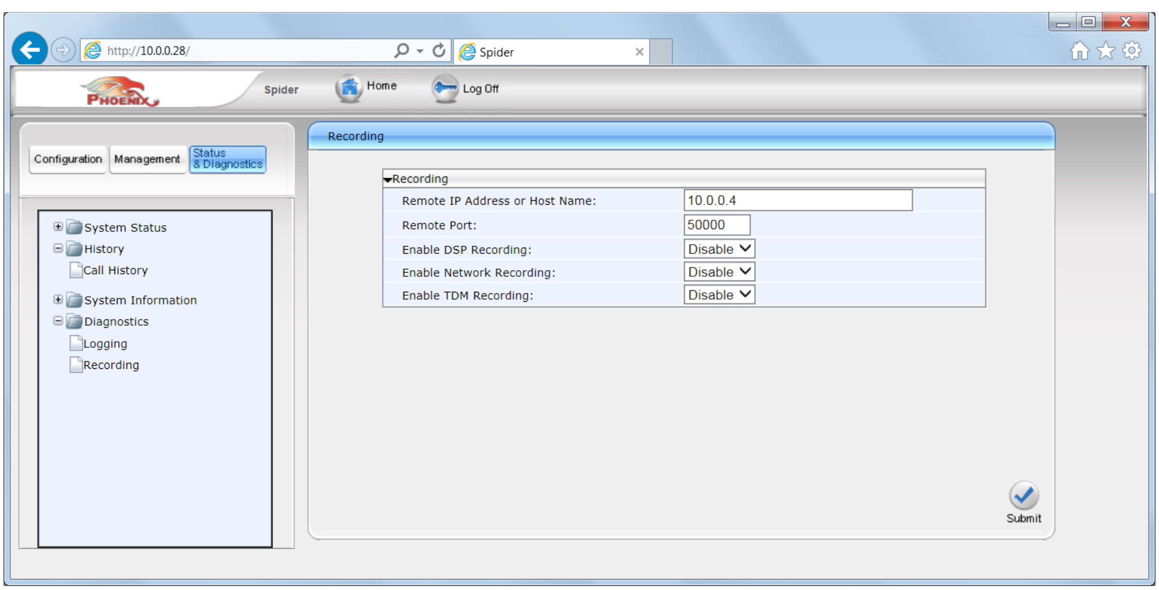
Under the Configuration File items the configuration file is displayed. It can be saved to a file on the host, or new configuration file can be loaded from the host.

Under the Firmware upgrade the user can point to an image file on his PC, which will be uploaded and update the firmware software. The ‘Initialize Configuration’ is disabled by default, keeping your current configuration. If enabled, the configuration will be initialized, the phone book, call logs and preferred ringtone will be erased.

4.2.3 Administration

Under Administration, the username and password can be changed, the initial ‘factory’ configuration can be restored, and the device can be restarted.

4.3 Status & Diagnostics



4.3.1 System Status

4.3.1.1 Network Status

Displays LAN information: Type, IP Address, Subnet Mask, Default Gateway Address, DNS address, and MAC address.

4.3.1.2 VOIP Status

Not implemented.

4.3.2 History

Includes the call logs (Missed, Received, and Dialed).
Each list is displayed by pages of up to 10 item.
Each item in the lists displays the phone number, time of the call and its duration.
Each item can be removed from the list, and the whole list can be deleted.

4.3.3 System Information

Display the Model number, Firmware Version, and the release date of the Firmware.

4.3.4 Diagnostics

4.3.4.1 Logging

System logging can be configured using the Web to help debugging.
It is not implemented in the released version.

4.3.4.2 Recording

Recording parameters allows debugging the voice activity of the phone using the Web.

Parameter	Description
Recording	
Remote IP Address or Host Name	The IP address (in dotted-decimal notation) of the remote computer to which the recorded packets are sent. The recorded packets should be captured by a network sniffer (such as Wireshark). The default value is 0.0.0.0.
Remote Port	Defines the UDP port of the remote computer to which the recorded packets are sent. The valid range is 1024 to 65535. The default value is 50000.
Enable DSP Recording	Activates the packet recording mechanism. ▪ Disable (default) ▪ Enable

Enable RTP Recording	Activates the DSP RTP recording. ▪ Disable (default) ▪ Enable
Enable Network Recording	Activates the DSP network (TDM Out) recording. ▪ Disable (default) ▪ Enable
Enable TDM Recording	Activates the DSP TDM (TDM In) recording. ▪ Disable (default) ▪ Enable