

VoIP ATA User's Manual

Version 4.x

SignalSys, Inc.
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1.0 INTRODUCTION

This document is the User's Guide for Voice-over-IP (VoIP) ATA and IAD. This document covers SP100, SP100X, SP200X, SP200X1S1O, and SP400. Please ignore the router function in the SP100 and SP400 models. The examples and pictures used here are SP200X.

Safety Precaution

Read these Safety Precautions carefully before you install the SP200X

- Read the User Manual before using the SP200X
- Keep your User Manual in a safe place
- Pay attention to all remarks with warning marks
- Check the power voltage in your area
- Make sure the power plug is not over burdened, which may cause equipment damage and fire
- Keep power cord standard wiring. Do not put anything on it
- Keep the equipment in a cool place
- Do not use other accessories that are not provided in your package
- Prevent thunder damage to the device. During thunderstorms, please UN-PLUG the power cord
- Do not disassemble the device
- Be careful of the adapter when un-plugging the power cord
- Make sure the power switch is turned OFF before you PLUG-IN/UNPLUG the cord
- Do not connect the phone port of SP200X to wall jacket
- Do not connect SP200X with PBX without help of technical support engineer

2.0 CONFIGURING THE PLATFORM

The web interface has two privilege levels: **Administrator** and **User**. Both privilege levels are password protected by default. The user names for the Administrator and User are "**admin**" and "**user**" respectively. And the default password for both privilege levels is "**signalsys**". The major difference between an Administrator and a User is Administrator has full control over the VoIP device while User only has limited control. Here is a screen shot of the login prompt.



Enter Network Password

Please type your user name and password.

Site: 192.168.123.1

Realm: /

User Name: admin

Password:

Save this password in your password list

OK Cancel

2.1 Home



SP200X VoIP ATA

[Home](#) [WAN](#) [LAN](#) [SIP](#) [CODECS](#) [System](#) [Download](#) [Configuration](#) [Reset](#) [Logout](#)

Home

Welcome to the SP200X download and configuration utility.
Select from the configuration options in the menu above.

System Information

System Uptime:	0 days, 0h 10m 36s
NTP time:	NTP Time Not Available
LAN IP Address:	192.168.123.1 (Static)
MAC Address:	00:09:d5:00:11:99
Security:	Password installed
Application Code Version:	SP200X_R4.2.2
Downloader Code Version:	5.1.0NOCRC
Serial Number:	S2AFI2635102X

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SP200X VoIP ATA

[Home](#) [WAN](#) [LAN](#) [System](#) [Reset](#) [Logout](#)

Home

Welcome to the SP200X download and configuration utility.
Select from the configuration options in the menu above.

System Information

System Uptime:	0 days, 0h 13m 55s
NTP time:	NTP Time Not Available
LAN IP Address:	192.168.123.1 (Static)
MAC Address:	00:09:d5:00:11:99
Security:	Password installed
Application Code Version:	SP200X_R4.2.2
Downloader Code Version:	5.1.0NOCRC
Serial Number:	S2AFI2635102X

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This page is the first page displayed when the device is logged in by an Administrator or a User. As the screen shots shown above, user level privilege doesn't have access to the configuration page of SIP, CODECS, Download and Configuration. It shows how long the device has been running since its last reboot, the IP address the device is currently using, whether or not the device is password protected, and also displays the main application and downloader application

firmware versions. In addition, MAC address of the WAN port, and serial number of the device, if it has one, are also displayed in this page.

2.2 WAN

This page and its sub-pages are available on device that supports routing/bridging and allows viewing and configuration of the WAN interface status/settings. Please note that any actions /modifications which alter the topology of the Ethernet Bridge will result in the spanning tree protocol to relearn. During this relearn period (may take up to one minute), HTTP access will be unavailable on all bridged interfaces (including the LAN interface). Note that user level privilege doesn't have access to MAC Spoofing sub-page.

WAN Status

SignalSys[®] **SP200X VoIP ATA**

Home **WAN** LAN SIP CODECS System Download Configuration Reset Logout

WAN Status WAN Settings PPPoE MAC Spoofing

WAN Status

Interface Status

Enabled:	Yes
Service:	Routed
Protocol:	Ethernet
Interface Status:	Acquiring IP...

Network Settings

Dynamic IP Assignment:	YES (via DHCP)
IP Address:	0.0.0.0
MAC Address:	00:09:d5:00:11:99
Subnet Mask:	255.255.255.0
Default Gateway:	0.0.0.0
DNS Address:	80.0.0.1
DNS Address 2:	
Host Name:	
Domain Name:	
Priority Tag:	Not set
Broadcast limit:	100% (of downstream bit rate)
Multicast limit:	100% (of downstream bit rate)

This page displays the current status of the WAN interface, including the IP address and other network settings the interface is currently using. If the interface is configured for PPPoE, a button is available to allow the administrator to DISCONNECT an active WAN PPPoE session or to initiate (CONNECT) a PPPoE session (if currently disconnected).

WAN Settings

SignalSys[®] SP200X VoIP ATA

Home **WAN** LAN SIP CODECS System Download Configuration Reset Logout

WAN Status **WAN Settings** PPPoE MAC Spoofing

WAN Configuration

Device Operating Mode: Router

Obtain WAN configuration dynamically

Specify fixed WAN configuration

IP Address: 192.168.0.10

IP Netmask: 255.255.255.0

IP Gateway: 192.168.0.1

IP DNS Server:

IP DNS Server2:

Host Name:

Domain Name:

Uplink Configuration

Uplink Bandwidth (kbts/sec):

Fragment low-priority packets when bandwidth is low

Multicast Limits

This sub-page allows the user to configure the Ethernet WAN interface. First, select whether the device is to act as a router or a bridge between the WAN and LAN interfaces. Select whether you wish for the WAN interface to be configured dynamically (via a DHCP server on the network if Ethernet, or via PPP if using PPPoE), or statically configured. If you wish to statically assign the WAN interface settings, enter the IP address, subnet mask, default gateway IP address and DNS server IP address. It is also recommended that the network domain name be provided as well, to ensure correct DNS operation. Press "Save WAN Settings" to save and apply the new values. Any new settings will take effect immediately.

PPPoE



SP200X VoIP ATA

Home	WAN	LAN	SIP	CODECS	System	Download	Configuration	Reset	Logout
WAN Status	WAN Settings	PPPoE	MAC Spoofing						

WAN PPPoE Configuration

Enable PPPoE:

Authentication

Username:

Password:

Settings

Echo Timeout: seconds

Echo Count:

On this sub-page you can specify whether the WAN interface is to use PPPoE, and configure the PPPoE client. First, select whether PPPoE is enabled or disabled. If enabled, enter the username and password required for the login (authentication) process. Setting the idle-timeout will result in the PPP connection being torn down if the client detects inactivity on the connection over the specified timeout period. Leaving this field blank will result in the connection being permanently up (i.e. without timeout). Setting the echo-timeout will result in the PPP connection to send the echo request to server in specific time and the echo-count is used for disconnecting this PPP connection if the device does not receive the response of echo request. If the PPPoE server (service provider) requires any special Service Name or AC Name to be set, you can specify these tags here. Press "Save PPPoE Settings" to save and apply the WAN interface PPPoE client settings. Any new settings will take effect immediately.

MAC Spoofing



SP200X VoIP ATA

Home	WAN	LAN	SIP	CODECS	System	Download	Configuration	Reset	Logout
WAN Status	WAN Settings	PPPoE	MAC Spoofing						

MAC Spoofing Configuration

WAN MAC Address (Spoofed):

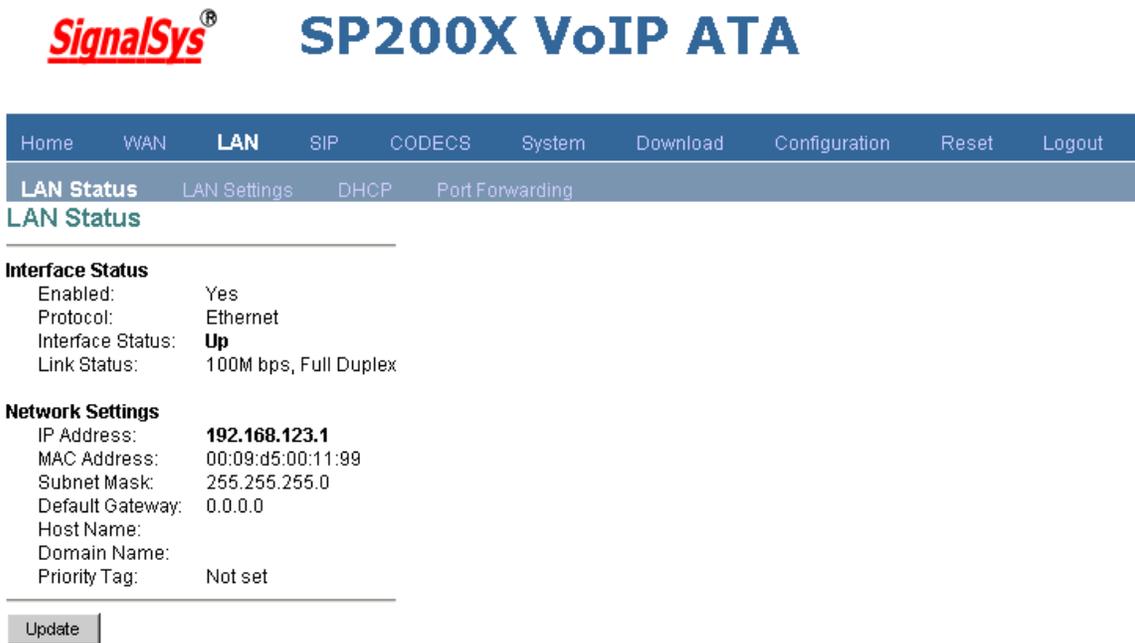
This sub-page allows the user to set the Ethernet hardware/MAC address to be used by the WAN interface. This is typically done to mimic ('spoof' or 'clone') the MAC address of one of the devices

connected on the private LAN interface. Enter the MAC (12 hex digits) address to assign to the WAN interface and press "Save MAC Spoofing Settings" to save and apply the new setting. Note that only administrator level privilege has access to this sub-page.

2.3 LAN

This page allows configuration of the device's local network settings. Sub-pages are available for the following on the left of page: the LAN interface status, configuration of the LAN interface settings, and configuration of DHCP server settings, and Port Forwarding (NAPT) Settings.

LAN Status



SignalSys[®] **SP200X VoIP ATA**

Home WAN **LAN** SIP CODECS System Download Configuration Reset Logout

LAN Status LAN Settings DHCP Port Forwarding

LAN Status

Interface Status

Enabled:	Yes
Protocol:	Ethernet
Interface Status:	Up
Link Status:	100M bps, Full Duplex

Network Settings

IP Address:	192.168.123.1
MAC Address:	00:09:d5:00:11:99
Subnet Mask:	255.255.255.0
Default Gateway:	0.0.0.0
Host Name:	
Domain Name:	
Priority Tag:	Not set

Update

This page displays the current status of the LAN interface, including the IP address and other network settings the interface is currently using.

LAN Settings



SP200X VoIP ATA

Home WAN **LAN** SIP CODECS System Download Configuration Reset Logout
LAN Status **LAN Settings** DHCP Port Forwarding

LAN Configuration

Network Settings

IP Address:
Subnet Mask:

Multicast Limits

Broadcast limit: % (of Ethernet connection bitrate)
Multicast limit: % (of Ethernet connection bitrate)

PHY Speed mode:

This sub-page allows the user to configure the private LAN interface settings. Assign an IP address to the LAN Ethernet port. This IP address is also the default router address for the devices on the private LAN. The default LAN interface IP address is set to 192.168.123.1. Enter the subnet mask for the private LAN. If you wish to set the broadcast and multicast limits for the bridge/router, enter these values as percentages of the LAN interface Ethernet bit rate. Leaving these values blank will imply values of 100%. Press "Save LAN Settings" to save and apply the LAN interface settings. Any new settings will take effect immediately.

DHCP



SP200X VoIP ATA

Home WAN LAN SIP CODECS System Download Configuration Reset Logout
LAN Status LAN Settings **DHCP** Port Forwarding

DHCP Server Configuration

Server Settings

Enabled Disabled

Client IP Address Range: 192.168.123. -

Client Network Information

Domain Name:
DNS Server 1: 2:

Static Address Assignments

Identify Using	Host Identifier	Internal Address	
<input type="text" value="Hostname"/>	<input type="text"/>	192.168.123.	<input type="text"/> <input type="button" value="Add"/>

This sub-page allows configuration of the private LAN DHCP server. The DHCP server settings do not apply if the device is operating in bridge only mode. Specify whether the device's internal DHCP server feature is enabled or disabled. Also indicate the IP address range to use for DHCP assignments to the LAN. Specify the domain name (optional) that is provided to LAN clients via DHCP. Two optional static DNS server IP addresses can be entered that will be provided to LAN clients via DHCP. These are in addition to the DNS servers automatically provided by WAN connections. Press "Save DHCP Settings" to save and apply the DHCP server settings. Any new settings will take effect immediately. In addition, up to 8 static DHCP address assignments can be configured. To add a static IP assignment, enter the LAN device's name (must be unique in the private network) and/or MAC address. Specify the static IP to be assigned and press the "Add" button. To remove an existing entry, press the "Remove" button next to the entry to be removed. To view the current active DHCP client binding table (the internal list of devices for which the server has provided an IP address lease), press "View DHCP Table". A popup window will appear displaying the list of bindings. Press "Update" to update the binding information. Press "Close" to close the binding table window. To clear the device's internal DHCP client binding table, press "Clear client binding table". Be aware that doing this will destroy the DHCP server's knowledge of the LAN clients it has provisioned, and may result in client problems when a client tries to renew its lease.

Port Forwarding (NAPT)



SP200X VoIP ATA

Home	WAN	LAN	SIP	CODECS	System	Download	Configuration	Reset	Logout
LAN Status	LAN Settings	DHCP	Port Forwarding						

Port Forwarding Configuration

Reserved Ports
 The following ports have been reserved by the CPE, and may not be forwarded to the LAN
68, 5060-5070, 8000-8015, 7001-7005, 5555, 80, 23, 161

Port Forwarding to LAN

Port Range	Protocol	Destination Address	
<input type="text"/> - <input type="text"/>	Both	192.168.123.	<input type="text"/> <input type="button" value="Add"/>

DeMilitarized Zone
 If specified, packets which port are not listed above will be forwarded to this DMZ host
 192.168.123.

This page allows the user to customize the devices port forwarding feature. The port forwarding feature does not apply if the device is operating in bridge only mode. Port forwarding provides WAN access to the internal LAN, by specifying that traffic over certain ports are to be directed at particular LAN hosts. This feature is available only in router mode. Up to 8 ports forwarding entries ("Pinholes") can be configured. To add a port forwarding entry, configure the Port Range to be forwarded, the Protocol to be forward (TCP, UDP or both), and destination LAN IP Address, press the "Add" button to add the entry. To remove an existing entry, click the "Remove" button next to that entry you wish to remove. Note that certain port numbers are reserved by the CPE for its own internal use. These ports may not be used for port forwarding to the LAN. Ports which

may be reserved by the device include those used by VoIP call signaling, RTP packets, HTTP and SNMP. All reserved (unavailable) ports will be displayed on this page.

2.4 SIP Server



Home	WAN	LAN	SIP	CODECS	System	Download	Configuration	Reset	Logout	
Server	Extensions	Digit Map	User 1	User 2	OOB Signaling	ToGDfServ	Tone	Ring	Service Code	Phone Book

SIP Server Configuration

Primary Server Settings	Secondary Server Settings
(Current Server: 192.168.1.200 ; Domain: 192.168.1.200)	(Current Server: 0 ; Domain:)
* Address: <input type="text" value="192.168.1.200"/> (IP or FQDN)	* Address: <input type="text"/> (IP or FQDN)
* Port: <input type="text" value="5060"/>	* Port: <input type="text" value="5060"/>
Domain Name: <input type="text" value="192.168.1.200"/>	Domain Name: <input type="text"/>
<input checked="" type="checkbox"/> Send Registration Request with Expire Time <input type="text" value="3600"/>	<input type="checkbox"/> Send Registration Request with Expire Time <input type="text"/>
Outbound Proxy IP: <input type="text" value="192.168.1.201"/> (IP or FQDN)	Outbound Proxy IP: <input type="text"/> (IP or FQDN)
Outbound Proxy Port: <input type="text" value="5060"/>	Outbound Proxy Port: <input type="text" value="5060"/>

RTP Port Number Setting(5000-65535) -

NAT Traversal Settings

NONE
 UPnP Control Point
 STUN Server IP: (IP or FQDN) STUN Server Port:

* Leaving a setting blank will force the unit to use the information obtained via DHCP and/or DNS.

This page allows configuration of the SIP server and endpoint settings. Enter the address and port value of the SIP server. The address may be an IP address or the name of the server. If no SIP server address is entered, the device will attempt to self provision a SIP server using a DNS query. For this to be successful, ensure that the DNS settings on the device include a DNS server address that is configured with the SIP server address and will respond to the query, and the appropriate domain name of the network. If you wish to specify a special SIP domain name, you may enter the domain name here. If no domain name is entered, the SIP domain name will be set to that of the network (i.e. that which is obtained via DHCP, or specified on the LAN settings page). The currently provisioned SIP Server and Domain are displayed beside "SIP Server Settings" for informational purposes. Select whether or not to send a Registration Request to the SIP server by checking the box next to "Send Registration Request". For the endpoint, set the dial plan to be used by all lines (refer to section 4.0 for details on the dialplan representation), and select the transport method to be used for SIP signaling (either UDP or TCP). For each line on the endpoint, enter the Line Phone Number, Caller-ID Name, signaling port value, authentication Username and Password, and select if AEC is to be performed on this line. Press "Save SIP Settings" to save the new values.

Extensions

SignalSys[®] SP200X VoIP ATA

The screenshot shows the web interface for the SignalSys SP200X VoIP ATA. The top navigation bar includes links for Home, WAN, LAN, SIP, CODECS, System, Download, Configuration, Reset, and Logout. Below this, a secondary navigation bar shows links for Server, Extensions (selected), Dial Map, User 1, User 2, OOB Signaling, ToS/DiffServ, Tone, Ring, Service Code, and Phone Book. The main content area is titled "SIP Extensions" and contains a list of configuration options with checkboxes:

- Support PRACK method with provisional response reliability
- Encode SIP URI with user parameter
- Session Timer use UPDATE method
- Call Hold using c=0.0.0.0 (RFC 2543) in SDP
- enable Global Number support (E.164)
- send NOTIFY for REFER request
- send Message Waiting Indicator (MWI) SUBSCRIBE command
- No Authorization Header in re-REGISTER
- Check existence of To Tag in INVITE 2xx response

Below the SIP Extensions section is the "SIP Timers" section, which includes input fields for timer values in seconds:

- Send INVITE with Timer header value: Seconds
- SIP Session Timer value: Seconds
- SIP Keep Alive Timer value: Seconds
- Conditional Call Forwarding Timer: Seconds
- Inter Digit Timer: Seconds.

This page allows specification of the SIP signaling stack behavior under certain scenarios.

1. Support PRACK method: If you wish for the SIP stack to implement reliable transmission of provisional responses according to RFC 3262 (using the PRACK method), check the option.
2. Encode SIP URI with user parameter: include the user parameter "user=phone" in the SIP URI headers.
3. Session Timer use UPDATE method: Session timer use update instead of reinvite.
4. Call Hold using C=0.0.0.0: using the call hold method described in RFC2543. If unchecked, the call hold would follow RFC3263 method.
5. Enable Global Number support (E.164): add prefix "+" for dialed numbers in sip invitation.
6. Send NOTIFY for REFER request: send out NOTIFY request to transferor for unattended and attended call transfer.
7. Send Message Waiting Indicator (MWI) SUBSCRIBE command: send SUBSCRIBE after registered to server to check if there are any messages to be read.
8. Send INVITE with Timer header: encode Timer header in all INVITE requests for ringing timeout.
9. Enable SIP session timer: If you wish for the SIP stack to implement a session timer according to "draft-sipsession-timer", select the option.
10. SIP T1 Timer, SIP T2 Timer, and SIP T4 Timer: please refer to RFC3261.

Press "Save SIP Extension Settings" to save the new values.

User 1

SignalSys® SP200X VoIP ATA

Home	WAN	LAN	SIP	CODECS	System	Download	Configuration	Reset	Logout	
Server	Extensions	Digit Map	User 1	User 2	OOB Signaling	ToSIP#Serv	Tone	Ring	Service Code	Phone Book

User 1 Configuration

Line 1	Phone Number	CallerID Name	Port	UserName	Password
Primary Server	408552103	408552103	5060	408552103	*****
Secondary Server			5060		

Line1 AEC Control ON

Line1 Gain Control

Input Gain Control (-12 ~ 18)db db

Output Gain Control (-12 ~ 18)db db

Supplementary Service Subscription

- Enable Call Waiting (Reject second incoming call)
- Enable Caller-ID Display
- Reject anonymous call
- Block Caller-ID in outgoing call

This sub-page and the next User2 sub-page allow the user to configure the device with phone number, caller ID, username and password specified by the service provider.

User 2

SignalSys® SP200X VoIP ATA

Home	WAN	LAN	SIP	CODECS	System	Download	Configuration	Reset	Logout	
Server	Extensions	Digit Map	User 1	User 2	OOB Signaling	ToSIP#Serv	Tone	Ring	Service Code	Phone Book

User 2 Configuration

Line 2	Phone Number	CallerID Name	Port	UserName	Password
Primary Server	1234	1234	5061	1234	****
Secondary Server			5060		

Line2 AEC Control ON

Line2 Gain Control

Input Gain Control (-12 ~ 18)db db

Output Gain Control (-12 ~ 18)db db

Supplementary Service Subscription

- Enable Call Waiting (Reject second incoming call)
- Enable Caller-ID Display
- Reject anonymous call
- Block Caller-ID in outgoing call

OOB Signaling



SP200X VoIP ATA

Home	WAN	LAN	SIP	CODECS	System	Download	Configuration	Reset	Logout	
Server	Extensions	Digit Map	User 1	User 2	OOB Signalling	ToS/DiffServ	Tone	Ring	Service Code	Phone Book

RTP Telephone Event Configuration

Send DTMF Events:

RFC2833 signalling using payload value:

Regenerate OOB DTMF tone

This sub-page allows configuration of the out-of-band signaling options for SIP. Select whether OOB telephone event signaling is to be done using the SIP INFO message, or to be done via RFC2833 RTP signaling.

ToS/DiffServ



SP200X VoIP ATA

Home	WAN	LAN	SIP	CODECS	System	Download	Configuration	Reset	Logout	
Server	Extensions	Digit Map	User 1	User 2	OOB Signalling	ToS/DiffServ	Tone	Ring	Service Code	Phone Book

ToS/DiffServ

Call Signalling Packets: (2 Hex digit byte value)

RTP Packets: (2 Hex digit byte value)

This sub-page is used to configure the Type-of-Service/Diffserv byte values that are to be used in the IP header of all transmitted SIP signaling packets and RTP packets. The ToS/DiffServ byte values are entered as two-digit hexadecimal values. If no special ToS/DiffServ value is to be used for a particular traffic type, enter "00" or leave the setting empty.

Press "Save ToS/DiffServ Settings" to save these new settings.

Tone

SignalSys® SP200X VoIP ATA

Home	WAN	LAN	SIP	CODECS	System	Download	Configuration	Reset	Logout	
Server	Extensions	Digit Map	User 1	User 2	COB Signaling	ToSIPUser	Tone	Ring	Service Code	Phone Book

Tone Configuration

Dial Tone:	<input type="text"/>
Recall Dial Tone:	<input type="text"/>
Confirm Tone:	<input type="text"/>
Ring Back Tone:	<input type="text"/>
Busy Tone:	<input type="text"/>
Reorder Tone:	<input type="text"/>
Receiver-Off-Hook Tone:	<input type="text"/>
Message-Waiting Indicator Tone:	<input type="text"/>
Call-Waiting Indicator Tone:	<input type="text"/>

This sub-page is used to configure Tones which applies in order to acknowledge users.

Dial Tone: The tone you hear when you pick up handset

Recall Dial Tone: The tone when you hold callee and prepare to make another call.

Confirm Tone: The tone after you've set up some service, like DND (Do Not Disturb), Call Forwarding, etc.

Ring Back Tone: The audible ringing you hear before callee picks up and answers your call.

Busy Tone: The tone indicates the number you dialed is in busy now.

Reorder Tone: The tone you hear if you dial an invalid number or the call is not available.

Receiver-Off-Hook Tone: The tone to alert you to place the handset on-hook.

Message-Waiting-Indicator Tone: The tone to notify you to call for message box.

Call-Waiting-Indicator Tone: The tone to make you aware of the second incoming call while you're in conversations.

Ring

The screenshot shows the 'Ring Configuration' page of the SignalSys SP200X VoIP ATA. The page has a blue header with navigation tabs: Home, WAN, LAN, SIP, CODECS, System, Download, Configuration, Reset, Logout. Below the header is a sub-menu with tabs: Server, Extensions, Digit Map, User 1, User 2, COB Signaling, ToS/DiffServ, Tone, Ring (selected), Service Code, Phone Book. The main content area is titled 'Ring Configuration' and contains three sections: 'Default Ring' with three input fields for Default Ring, CallWaiting, and Reminder Ring; 'Distinctive Ring Configuration' with eight input fields for Distinct Ring 1 through 8; and a 'Save Ring Settings' button at the bottom.

This sub-page is used to configure Ring Cadences required by Rings, Call-Waiting-Indicator, and Distinctive Ring features.

- Ring Configuration:

Default Ring: Default ring cadence when the phone rings.

Call-Waiting Reminder Ring: Ring cadence of Call-Waiting Reminder Ring.

- Distinctive Ring Configuration:

Distinctive Ring 1-8: Ring cadences provided for distinctive ring function.

You may customize them according the fixed format. For example, *ON(500),OFF(500),R* Will cause 500 milliseconds ring on, then 500 milliseconds off, and repeat steadily.

Service Code

The screenshot shows the 'Service Code Configuration' page of the SignalSys SP200X VoIP ATA. The page has a blue header with navigation tabs: Home, WAN, LAN, SIP, CODECS, System, Download, Configuration, Reset, Logout. Below the header is a sub-menu with tabs: Server, Extensions, Digit Map, User 1, User 2, COB Signaling, ToS/DiffServ, Tone, Ring, Service Code (selected), Phone Book. The main content area is titled 'Service Code Configuration' and contains a list of service code settings, each with a checkbox: Conditional Call Forwarding, Call Forwarding On Busy, Call Forwarding On, Call Forwarding Off, Do Not Disturb On, Do Not Disturb Off, Call Transfer, Call Return, and Speed Dial. Below the list is a note: '- use *00# or #00# format, xx=01-99'. At the bottom is a 'Save Service Code Settings' button.

This sub-page is used to configure Service Code, you can subscribe for service by setting through keypad.

Service Code Configuration:

Conditional Call Forwarding: 70#

Call Forward On: 72#

Call Forward Off: #72#

Do Not Disturb On: 74#

Do Not Disturb Off: #74#

Call Transfer: 98#

Call Return: 69#

Speed Dial: 68

Note: Do NOT change the service code values unless there is a conflict between the settings of your VoIP device and the settings provided by your service provider.

2.5 CODECS

SignalSys[®] SP200X VoIP ATA

Home WAN LAN SIP **CODECS** System Download Configuration Reset Logout

CODECS
Audio/CODEC Configuration

CODECS

Selected	Silence Suppression
<input checked="" type="checkbox"/> G711U	<input type="text" value="OFF"/>
<input checked="" type="checkbox"/> G711A	<input type="text" value="OFF"/>
<input type="checkbox"/> G723	<input type="text" value="OFF"/>
<input type="checkbox"/> G726	<input type="text" value="OFF"/>
<input checked="" type="checkbox"/> G729	<input type="text" value="OFF"/>

Packetization

Jitter Buffer

Adaptive Jitter Buffer: (maximum playout delay in milliseconds)
 (minimum playout delay in milliseconds)

Fixed Jitter Buffer: (fixed playout delay in milliseconds)

Automatically switch to Fixed Jitter Buffer upon fax/modem tone detection

If the device is running one of the four VoIP applications, this page is available for configuring the audio CODEC parameters, as well as the Jitter Buffer settings for the CODEC decoders. Enter which CODECs are to be supported. For some protocols (e.g. H.323 and SIP), the G711U and G711A protocols are always supported by default. For MGCP and H.248, it is possible to remove these CODECs from the devices list of supported "capabilities". Select which complex codec is to be supported. Due to memory limitations, it is not possible to select more than one complex codec. Select the packetization period to be used for each selected CODEC. For MGCP, a range of packetizations may be provided for each CODEC (to be advertised in the device's "capabilities" set). Select whether Silence Suppression is to be supported for each CODEC. The Jitter Buffer settings apply to all active CODEC decoders. You may choose between an adaptive jitter buffer and a fixed jitter buffer. For an adaptive jitter buffer, choose the maximum allowable playout delay (in milliseconds). For a fixed jitter buffer, choose the fixed playout delay

(in milliseconds). Finally, select whether or not a decoder should automatically switch from an adaptive jitter buffer to a fixed jitter buffer upon fax/modem tone detection. Adaptive jitter buffers are sometimes detrimental to fax transmission over G711 CODECs if they have to adapt too rapidly or too extensively due to inconsistent and widespread packet delays. In these adverse network conditions, a fixed jitter buffer provides superior performance when handling incoming fax transmissions over G711 CODECs. Press "Save CODEC Settings" to save the new CODEC parameters.

2.6 System Security



SP200X VoIP ATA

Home	WAN	LAN	SIP	CODECS	System	Download	Configuration	Reset	Logout
Security	Timeout	Localization	Handset	Port Number	SNMP	Service Access			

Set Security Password

Password is currently installed

Account: admin

Old password:

New password:

Confirm new password:

The password needed to access the VoIP device via web interface can be set in the sub-page. First enter the old password, then enter the new password and confirm new password. Click on the "Change Password" button to save the change. The VoIP device will log the user out and redirect the user to the login page.

Set Security Timeout



SP200X VoIP ATA

Home	WAN	LAN	SIP	CODECS	System	Download	Configuration	Reset	Logout
Security	Timeout	Localization	Handset	Port Number	SNMP	Service Access			

Set Web System Timeout

HTTP Authentication Timeout: (Seconds)

This field will active, the password must be setting.

Click "System" item on the top menu.

Click "Security" on the left menu.

In HTTP Authentication Timeout field, input timeout value you want. Then press change time button. After Http Authentication timeout value expired, it will redirect to password protected page.

Localization



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Localization

Country:

NTP Server:

Time Zone:

Adjust clock for daylight savings

Timezone:

Find the current time from a list of cities.

Country Caller ID:

The caller ID can find out who's calling you and keep track of how often they call.

Users should set the country field according to their geographical location, otherwise the Caller ID function might not work properly.

Timezone setting:

Click "System" on the top menu.

Click "Localization" on the left menu.

In NTP Server field, enter a NTP server IP address.

If you want to use the default NTP server, this field should be blank.

In Time Zone drop down menu, select one time zone.

In Adjust clock for daylight savings checkbox, if your country has daylight savings time, you can enable it.

Press Save Localization Settings button, then system will redirect to the web page of reset.

Country Caller ID setting:

Click "System" on the top menu.

Click "Localization" on the left menu.

In Country drop down menu, select one country.

Press Save Localization Settings button, then system will redirect to the web page of reset.

Handset

The screenshot shows the SignalSys SP200X VoIP ATA web interface. At the top left is the SignalSys logo. To its right is the title "SP200X VoIP ATA". Below the title is a navigation menu with tabs: Home, WAN, LAN, SIP, CODECS, System (highlighted), Download, Configuration, Reset, Logout. A secondary menu below it includes Security, Timeout, Localization, Handset (highlighted), Port Number, SNMP, and Service Access. The main heading is "Media Hub Handset Configuration". Underneath is a section titled "Control Timer Values" with two input fields: "Hook Flash Timer Min:" and "Hook Flash Timer Max:", both followed by "Milliseconds". A note below the fields reads: "*Please enter a multiple of 10.(ex:10,20,30...)". At the bottom of the section is a "Save Handset Settings" button.

This sub-page allows user to configure the flash hook time interval.

Port Number

The screenshot shows the SignalSys SP200X VoIP ATA web interface for Port Number configuration. At the top left is the SignalSys logo. To its right is the title "SP200X VoIP ATA". Below the title is a navigation menu with tabs: Home, WAN, LAN, SIP, CODECS, System, Download, Configuration, Reset, Logout. A secondary menu below it includes Security, Timeout, Localization, Handset, Port Number (highlighted), SNMP, and Service Access. The main heading is "Http Server Listen Port Number Setting". Underneath is an input field labeled "HTTP Server Port Number:". At the bottom of the section is a "Change Port Number" button.

This allows the device to use different port number for the http server. The default port number is 80.

SNMP



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SNMP Configuration

SNMP Trap Configuration

IP address: Trap Community:

SNMP Community Configuration

Read Community: Write Community:

SNMP System Configuration

System Description:

System Objectid:

This sub-page is used for configuring the device's SNMP manager. Configure the SNMP Trap Host IP address and community, the SNMP read and write community parameters, and the SNMP System Description and System Object ID parameters. Press "Save SNMP Settings" to apply the new values. These settings will only take effect when the device is rebooted.

Service Access



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Security	Timeout	Localization	Handset	Port Number	SNMP	Service Access			

Service Access Configuration

Select which interfaces are allowed access to the services listed below:

	LAN	WAN
HTTP (Web access):	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>
SNMP:	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>
TELNET:	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>

Save Service Access Settings

If the device is a multi-interface device (bridge/router), then this sub-page is available to enable/disable access to certain system level network services on the device's interfaces. From this page, an administrator can choose to allow or disallow HTTP, SNMP and Telnet access from devices on the LAN or WAN or both. Care must be taken while configuring the HTTP access, as once you have saved you will no longer be able to access the device's web pages if you have disabled HTTP on the interface you are currently accessing the device on. Press "Save Service Access Settings" to save and apply the new access settings.

2.7 Download

This page provides two options for downloading a new firmware application image to the device. If you wish to download the new firmware image using TFTP, enter the filename of the ROM image and enter the IP address of the TFTP server on which this file resides. To initiate the TFTP download process, press "Start TFTP Download." If the ROM image is stored on the same local machine you are using to access the device's web pages, you can choose to download the ROM file to the device using an HTTP post. Enter the filename of the ROM image or press "Browse" to help locate the file. To initiate the HTTP download process, press "Start HTTP Download." If the main application is executing at the time, the device will automatically reboot itself into the downloader mode and begin the download process. If the downloader application is executing at the time, the download process will begin. The download status will be displayed when the image download process is complete.

Download



SP200X VoIP ATA

Home	WAN	LAN	SIP	CODECS	System	Download	Configuration	Reset	Logout
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Download

Warning! The download process will reset the unit into the download mode. This will terminate all network connections and reset

TFTP Download method (Select remote TFTP server IP address and filename)

TFTP Server IP:

Filename:

HTTP Download method (Select filename on local browser machine)

Filename:

URL Download method (Currently ftp://, http:// and https:// are supported)

URL:

HTTP Download method:

When using http to upgrade firmware, it will check firmware version before starting download process. In Filename field, press Browsing Button. Press Start HTTP Download button to start downloading file. If firmware version doesn't fit in with old version, it won't allow updating.

2.8 Configuration Backup



SP200X VoIP ATA

Home	WAN	LAN	SIP	CODECS	System	Download	Configuration	Reset	Logout
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Backup Restore

Configure File Backup

Backup configuration values of system settings to a file from the device:

Click "Configuration" item on the top menu.

Click "Backup" item on the left menu.

Press Backup Configure File button to save configuration file.

Restore



SP200X VoIP ATA

Home	WAN	LAN	SIP	CODECS	System	Download	Configuration	Reset	Logout
Backup	Restore								

Configure Restore

Configure Restore method (Select filename on local browser machine)

Filename:

This page is used to restore configuration values of system settings from a previously saved configuration file, or default factory values that stored inside the device.

Restore configuration from a file:

Click "Configuration" item on the top menu.

Click "Restore" item on the left menu.

Press Browsing button to select file by backup from local machine.

Press Start Download button to process downloading file.

After downloading file is finished, the web system will redirect to restart device.

2.9 Reset



SP200X VoIP ATA

Home	WAN	LAN	SIP	CODECS	System	Download	Configuration	Reset	Logout
Reset									

Reset

You must reboot to make your changes active.

Warning! Resetting the system will terminate all network connections and reset your browser connection.

Reset and execute Main Application

Reset and execute Downloader Application

This page provides options for resetting the device. Select whether you wish to reset the device and start executing the main (default) application, or whether you wish to reset the device and start executing the internal downloader application. Press "Reset" to reset the device

2.10 Logout

This sub-page shows the user that s/he has log out from the device.

3.0 IVR

To use IVR, one should pick up the phone, and then dial four consecutive asterisks (****.) And hang up the phone will stop IVR.

CODE STATUS USER INPUT

**** Menu Enter choice code.

100# Network status None.

110# DHCP setting

1# to enable

2# to disable

or # back to menu

120# IP address setting

Use "*" to instant of ".", and "#" to end.

Ex: 172*16*230*227#

Or # back to menu.

130# Gateway setting Use "*" to instant of ".", and "#" to end.

Or # back to menu.

140# Net mask setting Use "*" to instant of ".", and "#" to end.

Or # back to menu.

4.0 DIAL PLANS

The SIP code will allow provisioning (via web browser) of the dial plan. A dial plan gives the unit a map to determine when a complete number has been entered and should be passed to the SIP server for resolution into an IP address. Dial plans are expressed using the same syntax as used by MGCP NCS specification.

The formal syntax of the dial plan is described by the following notation:

Digit ::= "0" | "1" | "2" | "3" | "4" | "5" | "6" | "7" | "8" | "9"

Timer ::= "T" | "t"

Letter ::= Digit | Timer | "#" | "*" | "A" | "a" | "B" | "b" | "C" | "c" | "D" | "d"

Range ::= "X" | "x" -- matches any digit

| "[" Letters "]" -- matches any of the specified letters

Letters ::= Subrange | Subrange Letters

Subrange ::= Letter -- matches the specified letter

| Digit "-" Digit -- matches any digit between first and last

Position ::= Letter | Range

StringElement ::= Position -- matches any occurrence of the position

| Position "." -- matches an arbitrary number of occurrences including 0

String ::= StringElement | StringElement String

StringList ::= String | String "|" StringList

DialPlan ::= String | "(" StringList ")"

A dial plan, according to this syntax, is defined either by a (case insensitive) string or by a list of strings. Regardless of the above syntax a timer is only allowed if it appears in the last position in a string (12T3 is not valid). Each string is an alternate numbering scheme. The unit will process the dial plan by comparing the current dial string against the dial plan, if the result is under-qualified (partial matches at least one entry) then it will do nothing further. If the result matches or is over-qualified (no further digits could possibly produce a match) then send the string to the SIP server and clear the dial string. The Timer T is activated when it is all that is required to produce a match. The period of timer T is 4 seconds. For example a dial plan of (xxxTjxxxx) will match

immediately if 5 digits are entered, it will also match after a 4 second pause when 3 digits are entered.

4.1 Sample Dial Plans

Simple Dial Plan

This allows dialing of 7 digital numbers (e.g. 5551234) or an operator on 0. Dial plan is (0T|xxxxxxx)

4.2 Non-Dialed Line Dial Plan

As soon as handset is lifted the unit contacts the SIP server (used for systems where DTMF detection is done in-call). Dial plan is (x.) i.e. match against 0 (or more) digits. Note: the dot '.'

4.3 Complex Dial Plan

Local operator on 0, long distance operator on 00, four digit local extension number starting with 3,4 or 5, seven digit local numbers are prefixed by an 8, two digit star services (e.g. 69), ten digit long distance prefixed by 91, and international numbers starting with 9011+variable number of digits. Dial plan for this is:

(0T|00T|[[3-5]xxx|8xxxxxxx]*xx|91xxxxxxxxxx|9011x.T)

General Troubleshooting

5.1 LED

LED "Power": ON/OFF (indicates power connection)

LED "LAN": ON/OFF (indicates LAN connection)

LED "WAN": ON/OFF (indicates WAN connection)

LED "Status": ON/OFF (indicate Registration status)

LED "Phone": ON/OFF (indicates phone in use)

5.2 Mass Deployment

If the "Mass Deployment" is selected, the device will take approximate **20 seconds** to startup. The LED "Phone" is ON when the router is in "Mass Deployment".

6. Make Phone Calls

6.1 Make a call

If you have your own dial plan, just dial the phone number. If you have no dial plan or using default dial plan, dial the phone number. If the "# used as quick dial" is checked, you may dial the phone number followed by #.

6.2 3-Way Conference

While two parties are in the conversation, either party can add the third party in the 3-way conference by flash hook (hang up once while holding handset) and dial 7. After you hear a dial tone, please dial telephone number to add third party for conference. To drop the third party, flash and dial 8.

6.3 Call Waiting

During the conversation, when you hear the call waiting tone, flash and dial "*". It will switch you to the incoming call. Flash and "*" again, it will be switched to the previous conversation.

6.4 Call Forwarding

Pick up the phone and dial *2. After hearing the dial tone, dial the forwarding phone number and #. Then, hang-up the phone. The incoming calls will be forwarded.

To set call forwarding off, pick up the phone and dial *3.

APPENDIX

A. Determining the IP Address

This section provides instructions on how to determine the IP address of a unit, or how to boot up into a 'safe' mode that allows the IP address to be reconfigured. Instructions differ depending on the platform type.

A.1 General Instructions

The following general instructions can be followed to determine the IP address of a unit. Several mechanisms are available:

- If the device is thought to be using DHCP to obtain an IP address, check the DHCP server log, leases file, or binding table (if available). Look for the MAC address of the unit. The MAC address is a 6-octet string and can be found on a barcode label on the device.
- Use a packet sniffer attached to the same network as the device. Set the sniffer to filter packets based on the unit's MAC address, and initiate some data packet transfer (e.g. make a VoIP call). Analyze the captured packets to determine the device's IP address.
- If the SNMP trap destination has been previously set, check your SNMP management server for the IP address which is sent out as part of the Cold Start trap on system reset.

B. Forcing 'Safe' Modes

The SP200X can be forced into a 'safe' downloader mode by performing the following steps in order. In this description, the RESET button, which is the button at the back end of SP200X:

1. Power cycle the unit while holding down the RESET button. **Release the RESET button in less than 4 seconds.**
2. To force the unit to use a fixed IP that is in the config file, press and immediately release the RESET button again **within 2 seconds** of releasing it after step 1 above. After 2 seconds of button inactivity, the device will start the downloader network stack in fixed IP mode. If no fixed IP information exists in the configuration file, the device will default back to DHCP mode.
3. Finally, to force the unit to use **10.1.0.54** as the IP address, press and immediately release the RESET button again a third time **within 2 seconds** of releasing it after step 3 above. The device will now use 10.1.0.54 as the IP address.

REMINDER: There is a two second time-out after each press of the RESET button, after which the downloader network stack will boot into the selected mode.

1 Presses = FIXED IP
2 Presses = 10.1.0.54

If the wrong network mode is started by mistake, the process can be easily restarted by holding down the RESET button and power cycling the device again.