

How to Add the SP-11N SIP Module to a VoIP System



Create a “SIP Proxy Server” on the VoIP system.

Obtain the IP address of this server.

In the Phone VoIP system add a new user and note the **username** and **password**.

Open up a browser (Chrome, Internet Explorer, etc.) and enter the SP-11N’s IP address

How to obtain SP-11N IP address:

- The SP-11N module should be installed on a compatible mixer/amplifier
- Disconnect the power supply (AD-1210P) from the SP-11N
- SP-11N must be connected to a Network
- Turn up the volume for the specific channel in which the SP-11N is installed (w/spkrs)
- Reconnect power supply to SP-11N
- Result: In about 6 seconds the SP-11N will announce its IP address, which has been pre assigned via DHCP (DHCP Dynamic Host Configuration Protocol)

❖ Open up a web browser and type in the SP-11N IP address and this page will load up.

HOME	CONFIGURATION	DEFAULTS	UPDATE	REBOOT	MAC: 00:08:E1:03:74:3D FW V1.01 Feb.24, 2014
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SP-11N



SIP Paging Module

APPLICATION STATUS

Application Mode	SIP Mode
SIP PBX	
SIP ID	
Time till next Registration	0 seconds
Call State	Idle
Remote Party	

AUDIO STATUS

Current Set Volume	0 %
Output Peak Level	0 dBFS
Input Peak Level	0 dBFS

Help

Home page
Gives an overview of the most important settings of the unit.

APPLICATION STATUS

Application Mode
Shows the current mode of the application, and may take the following values:

- *Device is still booting ...*
The Boot process has not finished yet.
- *SIP mode*
The device is in SIP mode. The SIP server name, and the SIP ID are also shown in this case
- *Peer to peer mode*
The device is in P2P mode, and configured to call to only one remote peer. Incoming calls will be accepted only from this peer.

Time till next Registration
Shows the remaining time till the next registration attempt. The current registration status is shown with different colours of the text:

- Device not registered
- Registration in progress
- Device registered

Call Status
Shows the current call state, and may take one of the following values: - *idle*:
No audio stream is received, and the SIP client is accepting calls
- *Getting incoming call*:

- ❖ Click on “**Configuration**” to see this page
- ❖ Enter the **SIP Server (PBX)** “this is the SIP Proxy Server IP address”
- ❖ Enter the **SIP ID (username)** “this is the user name added to the VoIP system”
- ❖ Enter the **SIP Password (secret)** “this is you new user password created in the VoIP system”
- ❖ Click **Apply**

HOME **CONFIGURATION** DEFAULTS UPDATE REBOOT MAC: 00:08:E1:03:74:3D FW V1.01 Feb.24, 2014

SP-11N 

SIP Paging Module

Basic Settings
Advanced Settings

Apply Cancel

BASIC SETTINGS

SIP PROTOCOL SETTINGS

Peer to Peer No Yes

SIP Server (PBX)

SIP ID (username)

SIP Password (secret)

INBOUND CALL SETTINGS

Phone pickup mode

Hang up time seconds

CONTACT OUTPUT CONTROL C-OUT-1 C-OUT-2 C-OUT-3 C-OUT-4 C-OUT-5

No call (idle)

Start call

KEY 1 during call

KEY 2 during call

KEY 3 during call

KEY 4 during call

KEY 5 during call

Help

BASIC SETTINGS

SIP PROTOCOL

Peer to Peer
Choose whether peer to peer calls should be allowed.
NOTE:When using P2P, the device uses always the default SIP (port 5060) and RTP (port 5004) ports. Make sure the remote peers are configured to listen on the default ports as well.

SIP Server (PBX)
Enter here the hostname/IP address of the SIP server.

SIP ID
Enter the SIP ID (username) that has been created for this device.

SIP Password
Leave this field empty if the PBX doesn't require authentication.

INBOUND CALL SETTINGS

Phone Pickup Mode
autoanswer: the call is auto-answered.

Hang-up Time
Set hang up time after answer if it is necessary to have a time-limit of pagings.

Contact Output Control

❖ This page shows up while the SIP module is being restarted with a 5 second countdown

HOME CONFIGURATION DEFAULTS UPDATE REBOOT MAC: 00:08:E1:03:91:BB FW V1.01 Feb.24, 2014

SP-11N 

SIP Paging Module

Settings saved. Please wait, the device is restarting!

Basic Settings **3** Advanced Settings

Apply Cancel Please click [here](#) after the countdown if your browser doesn't support forwarding.

Help

BASIC SETTINGS

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SIP Password
Leave this field empty if the PBX doesn't require authentication.

INBOUND CALL SETTINGS

Phone Pickup Mode
autoanswer: the call is auto-answered.

Hang-up Time
Set hang up time after answer if it is necessary to have a time-limit of pagings.

Contact Output Control
Select output (OFF:Open or ON:Close) of each contact output in each status.
"Stat call" means the status when the module answers to a received call.
"Key 1 during call" means the status when other calling party pushes key "1" during a paging.
All outputs keep until "Finished the call" or "Other calling party pushes another key"

OUTBOUND CALLS

Call on Device Input 0
Extension to be called when input 0 is closed.
In case of using "Peer to Peer" mode, enter here the ID and the IP address of the remote peer, for example:
1234@192.168.0.123

- ❖ Once the SIP module has been reinitialized this page shows up with the new information
- ❖ Contact Outputs 1-5 normally open have been pre-assigned at the factory to keys 1-5 for individual paging on zones 1-5 as well as KEY “9” to all contacts 1-5 for an all call page

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SP-11N 

SIP Paging Module

BASIC SETTINGS

SIP PROTOCOL SETTINGS

Peer to Peer No Yes

SIP Server (PBX)

SIP ID (username)

SIP Password (secret)

INBOUND CALL SETTINGS

Phone pickup mode

Hang up time seconds

CONTACT OUTPUT CONTROL C-OUT-1 C-OUT-2 C-OUT-3 C-OUT-4 C-OUT-5

No call (idle)	<input type="text" value="OFF"/>				
Start call	<input type="text" value="OFF"/>				
KEY 1 during call	<input type="text" value="ON"/>	<input type="text" value="OFF"/>	<input type="text" value="OFF"/>	<input type="text" value="OFF"/>	<input type="text" value="OFF"/>
KEY 2 during call	<input type="text" value="OFF"/>	<input type="text" value="ON"/>	<input type="text" value="OFF"/>	<input type="text" value="OFF"/>	<input type="text" value="OFF"/>
KEY 3 during call	<input type="text" value="OFF"/>	<input type="text" value="OFF"/>	<input type="text" value="ON"/>	<input type="text" value="OFF"/>	<input type="text" value="OFF"/>
KEY 4 during call	<input type="text" value="OFF"/>	<input type="text" value="OFF"/>	<input type="text" value="OFF"/>	<input type="text" value="ON"/>	<input type="text" value="OFF"/>
KEY 5 during call	<input type="text" value="OFF"/>	<input type="text" value="OFF"/>	<input type="text" value="OFF"/>	<input type="text" value="OFF"/>	<input type="text" value="ON"/>
KEY 6 during call	<input type="text" value="OFF"/>				
KEY 7 during call	<input type="text" value="OFF"/>				
KEY 8 during call	<input type="text" value="OFF"/>				
KEY 9 during call	<input type="text" value="ON"/>				
KEY 0 during call	<input type="text" value="OFF"/>				

OUTBOUND CALL SETTINGS

Call on Device Inputs

Input 0 Call ID

Help

BASIC SETTINGS

SIP PROTOCOL

Peer to Peer
Choose whether peer to peer calls should be allowed.
NOTE:When using P2P, the device uses always the default SIP (port 5060) and RTP (port 5004) ports. Make sure the remote peers are configured to listen on the default ports as well.

SIP Server (PBX)
Enter here the hostname/IP address of the SIP server.

SIP ID
Enter the SIP ID (username) that has been created for this device.

SIP Password
Leave this field empty if the PBX doesn't require authentication.

INBOUND CALL SETTINGS

Phone Pickup Mode
autoanswer: the call is auto-answered.

Hang-up Time
Set hang up time after answer if it is necessary to have a time-limit of pagings.

Contact Output Control
Select output (OFF:Open or ON:Close) of each contact output in each status.
"Stat call" means the status when the module answers to a received call.
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All outputs keep until "Finished the call" or "Other calling party pushes another key"

OUTBOUND CALLS

Call on Device Input 0
Extension to be called when input 0 is closed.
In case of using "Peer to Peer" mode, enter here the ID and the IP address of the remote peer, for example:
1234@192.168.0.123

❖ Click “HOME” and the SIP module information will show up like this, showing the card’s IP address pre-assigned via DHCP

The screenshot shows the web interface for the SP-11N device. The top navigation bar includes 'HOME', 'CONFIGURATION', 'DEFAULTS', 'UPDATE', and 'REBOOT'. The 'HOME' button is highlighted with a red box. The page title is 'SP-11N' and the TOA logo is in the top right corner. The main content area is divided into two columns. The left column displays 'SIP Paging Module' information under 'APPLICATION STATUS', including 'Application Mode' (SIP Mode), 'SIP PBX' (192.168.1.2), 'SIP ID' (SModule), 'Time till next Registration' (1730 seconds), and 'Call State' (Idle). The right column displays 'Help' information under 'APPLICATION STATUS', including 'Application Mode' (SIP mode), 'Time till next Registration', and 'Call Status'. A green arrow points from a green text box to the '1730 seconds' value.

Section	Parameter	Value
APPLICATION STATUS	SIP Mode	SIP Mode
	SIP PBX	192.168.1.2
	SIP ID	SModule
	Time till next Registration	1730 seconds
	Call State	Idle
AUDIO STATUS	Current Set Volume	50 %
	Output Peak Level	-99 dBFS
	Input Peak Level	-99 dBFS

Green text shows SP-11N is now registered with the SIP Proxy Server

❖ Calls made to the SP-11N’s extension number created in the VoIP system will now be broadcasted

❖ For more information click on the menu bar and HELP text for that area will appear on the right pane.

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SP-11N

SIP Paging Module

APPLICATION STATUS

Application Mode	SIP Mode
SIP PBX	192.168.1.2
SIP ID	SModule
Time till next Registration	1730 seconds
Call State	Idle
Remote Party	

AUDIO STATUS

Current Set Volume	50 %
Output Peak Level	-99 dBFS
Input Peak Level	-99 dBFS

Help

Home page
Gives an overview of the most important settings of the unit.

APPLICATION STATUS

Application Mode
Shows the current mode of the application, and may take the following values:

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Time till next Registration
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- Device not registered
- Registration in progress
- Device registered

Call Status
Shows the current call state, and may take one of the following values: idle

Technical Specifications for the SP-11N module

The SP-11N, 900 Series SIP module can be used with the SIP-PBX system which meets the following conditions:

*SIP-PBX Systems which operate under SIP (Session Initiation Protocol /IETF RFC3261) and compatible with Data Transmission “IETF RFC2883”.

The **Session Initiation Protocol (SIP)** is a signaling communications protocol, widely used for controlling multimedia communication sessions such as voice and video calls over Internet Protocol (IP) networks. The protocol defines the messages that are sent between endpoints, which govern establishment, termination and other essential elements of a call. SIP can be used for creating, modifying and terminating sessions consisting of one or several media streams. SIP can be used for two-party (unicast) or multiparty (multicast) sessions. This sometimes causes setting and/or registration problems or difficulties, but SIP is a standardized protocol and its communication method is also standardized. Therefore, if any SIP-PBX system which meets the above conditions cannot or has difficulty to setup and/or register the SP-11N SIP module, it is out of our (TOA) control but the SIP Server company’s responsibility.

Specifications

SIP Compliance	RFC3261
Power Input	48 V PoE Class 0 (Max 12.95 W - Idle 3 W)
Physical Connection	RJ45