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

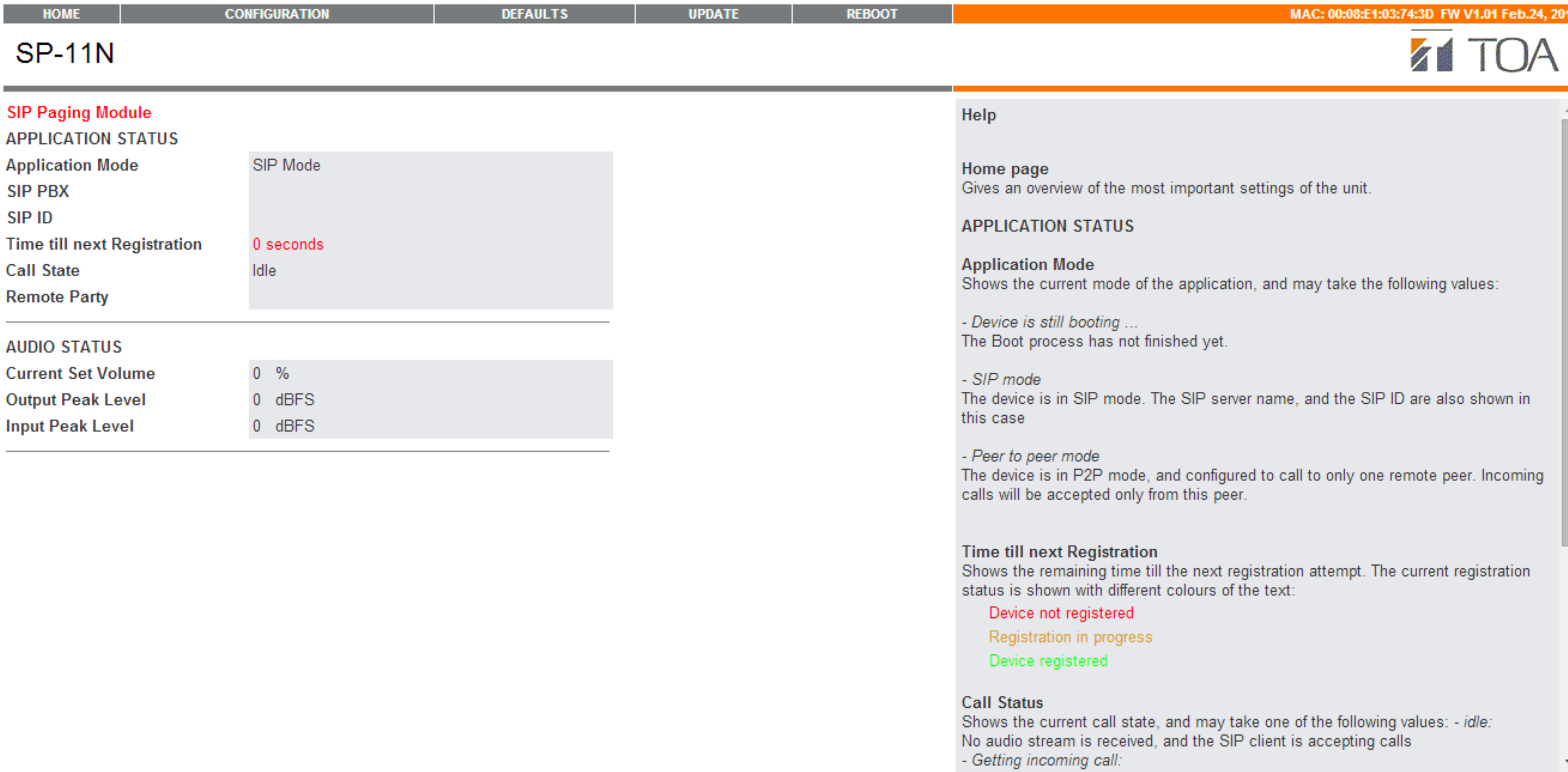
Create a “SIP Proxy Server” on the VoIP system.
Obtain the IP address of this server.
In the Phone VoIP system add a new user and note the username and password.

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

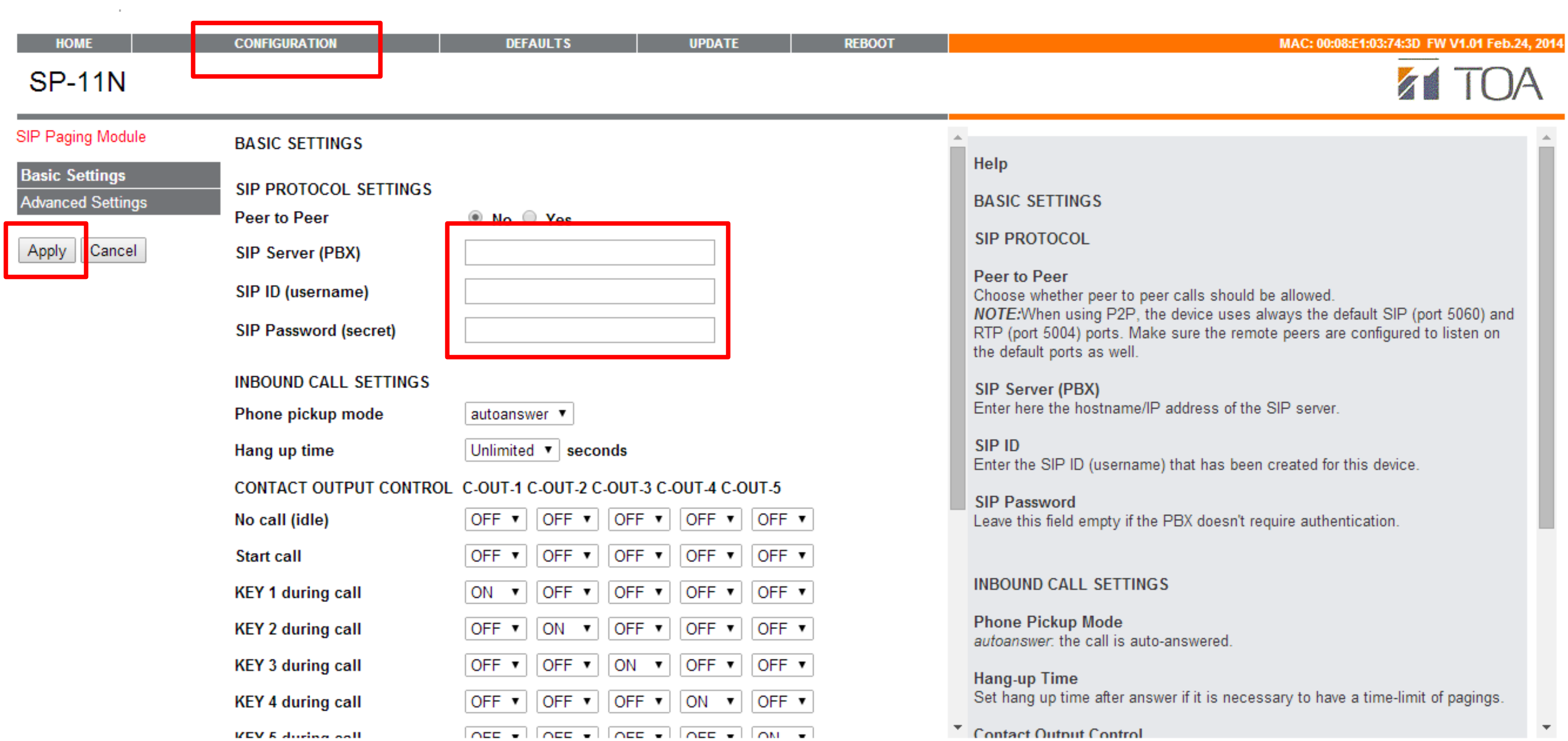
How to obtain SP-11N IP address:

- 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)

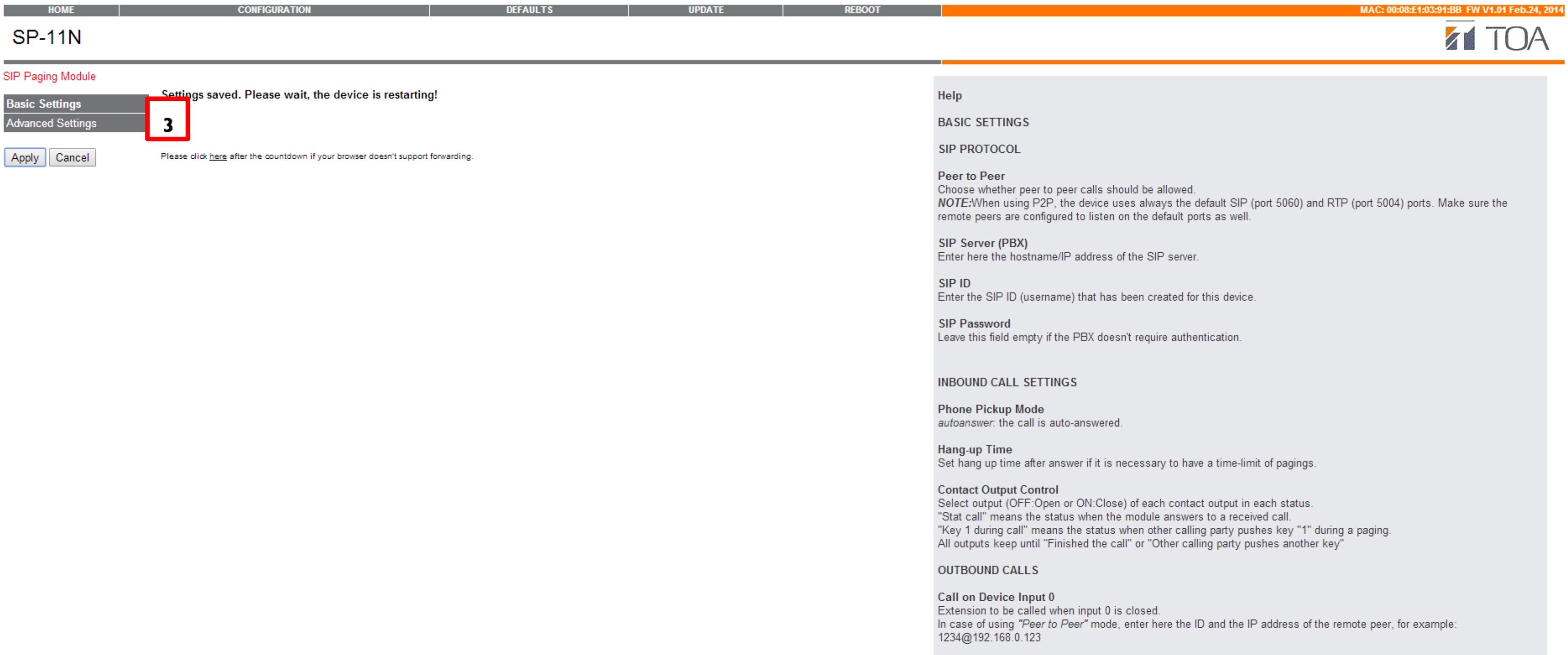
❖Open up a web browser and type in the SP-11N IP address and this page will load up.



- 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

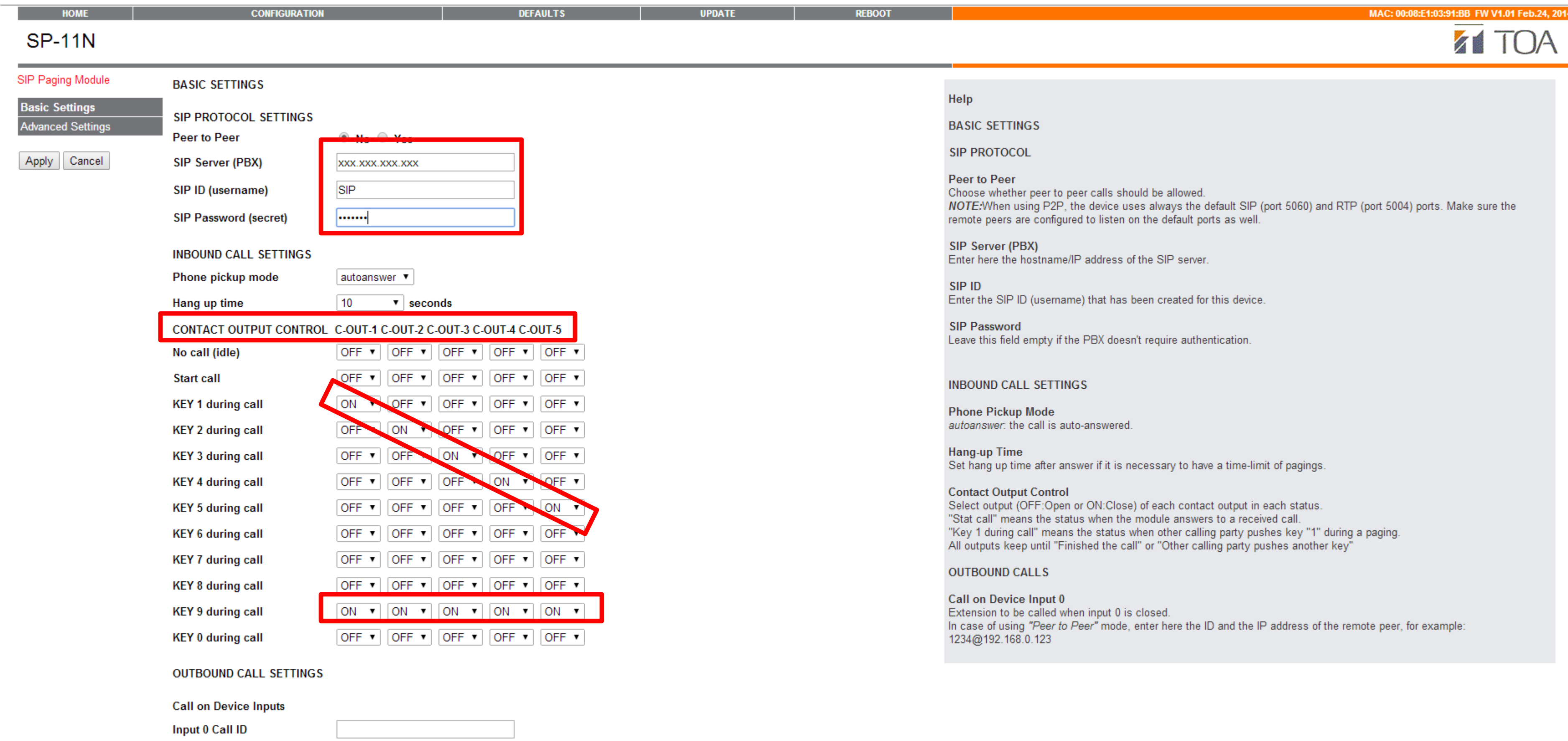


❖This page shows up while the SIP module is being restarted with a 5 second countdown

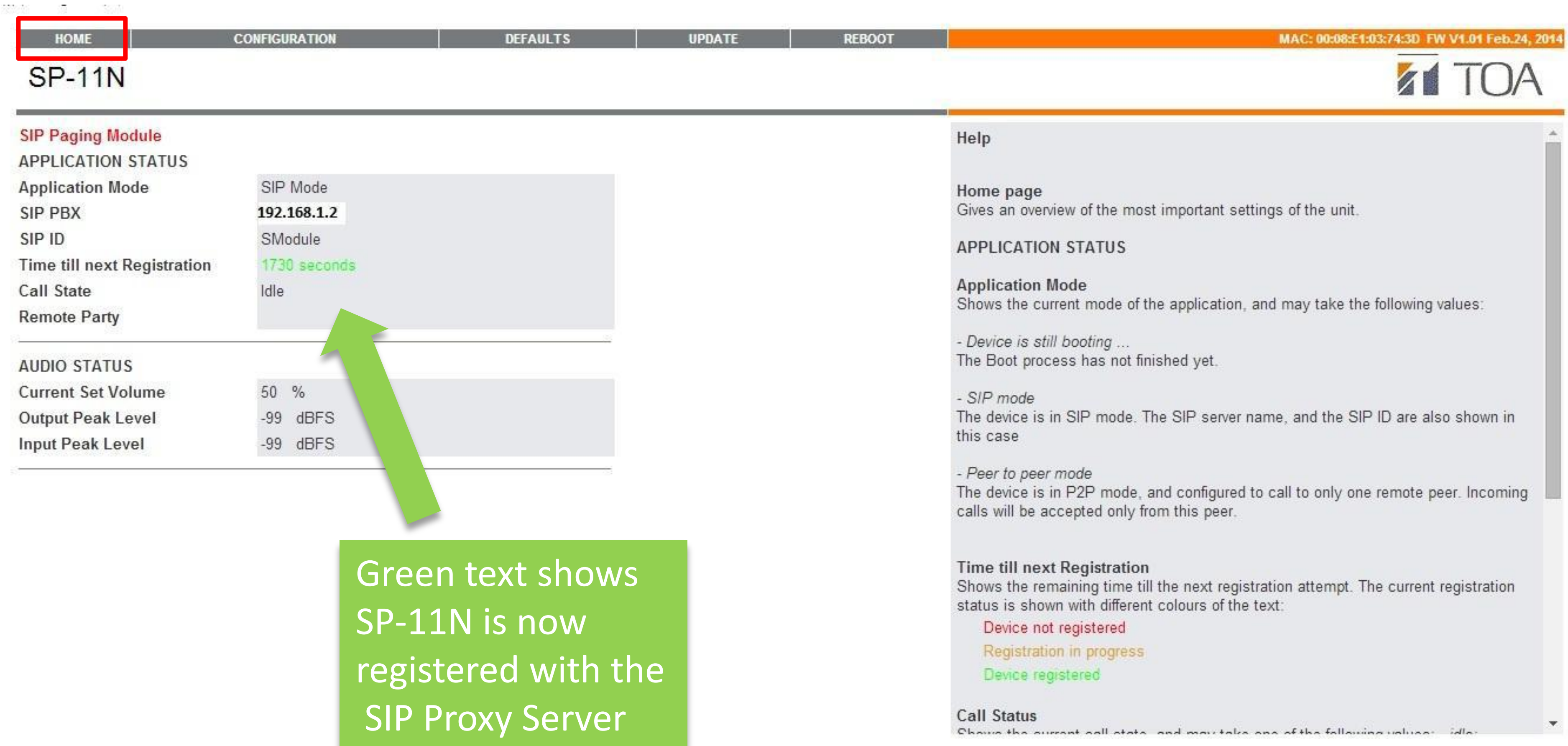


❖Once the SIP module has been reinitialized this page shows up with the new information

❖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



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



❖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.

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