SIP Trunking using the Optimum Business SIP Trunk Adaptor and the Talkswitch 848 VS IP PBX Version 7.11.006

Goal

The purpose of this document is to describe the steps needed to configure the Talkswitch IP PBX for proper operation with Optimum Business SIP Trunking.

Prerequisites

Please follow the instructions in the Optimum Business SIP Trunk Set-up Guide. The Set-up Guide was left by the Optimum Business technician at installation. If you do not have the Set-up Guide, go to **optimumbusiness.com/SIP** to download a copy.

Follow the instructions to configure the LAN side settings.

Important: The Optimum SIP Trunk Adaptor needs to convert out of band DTMF sent by the IP PBX to Inband. This is in step 3 of the Guide. Make sure you click the box next to “Convert Inband DTMF”.

This knowledgebase solution provides the configuration steps for both PBX registration and static or non-registration modes of PBX operation.

- Talkswitch – Version: 7.11.006
- Talkswitch System Software Model: 848 VS
- Java installed on Windows machine, Flash Player 10

optimum.

Talkswitch Configuration

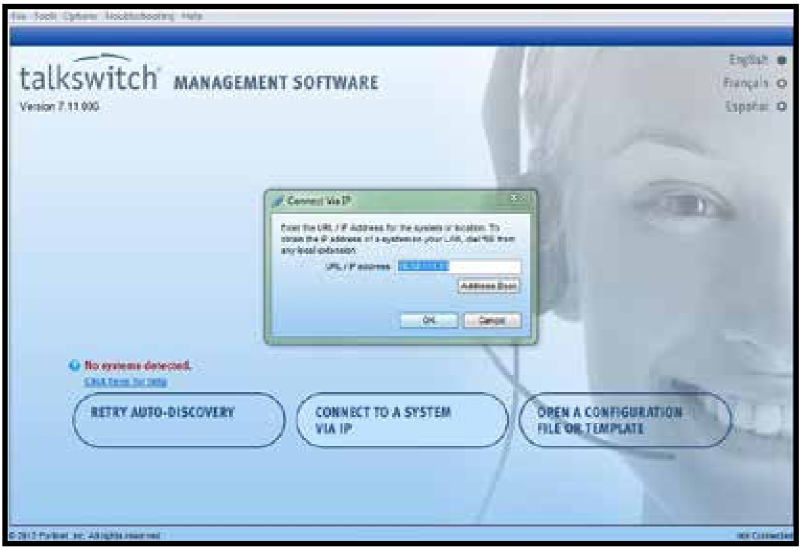
The steps below describe the minimum configuration required to enable the Talkswitch PBX to use Optimum Business SIP Trunking for inbound and outbound calling. Please refer to the Talkswitch product documentation for more information on other advanced PBX features.

When connecting to the Talkswitch from the LAN with a PC, make sure there is no other enabled NIC or Network (like wireless) on the PC. This will interfere with the configuration process.

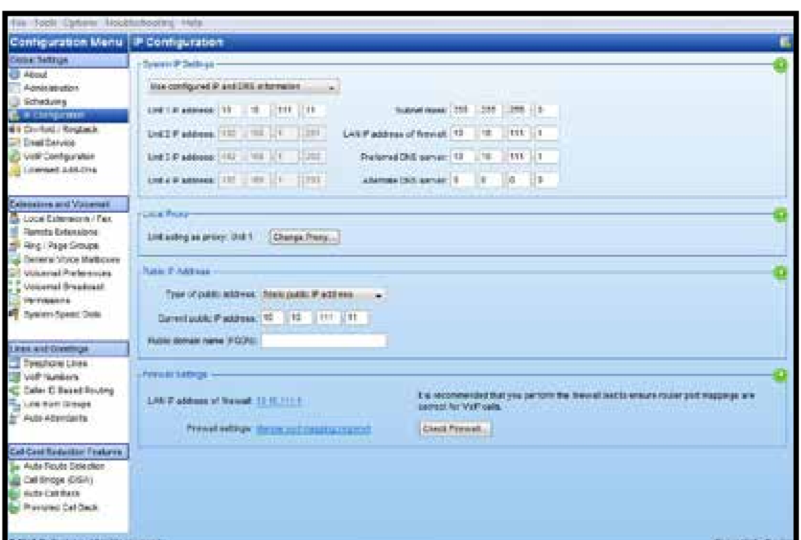
Configuration:

Plug PC into LAN side of PBX. By default a 192.168.3.1
By default the password will be 1234
In this configuration guide the IP address will be 10.10.111.11.
The Optimum Business SIP Trunk Adaptor needs to have the “Convert Inband DTMF” check box selected.

1 Start by detecting the Talkswitch and connecting to it via the LAN IP.

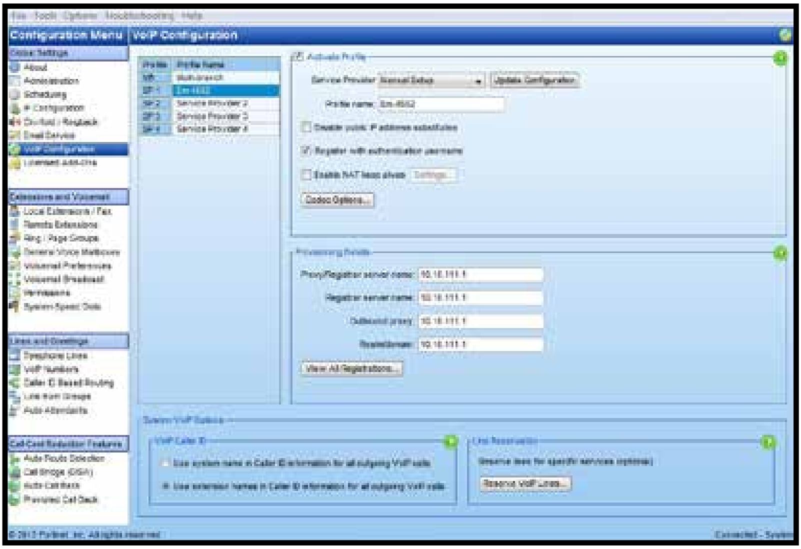


2 Configure the IP address settings for the Talkswitch to work properly with the network it will be residing on. Configuration Menu ▸ IP configuration.



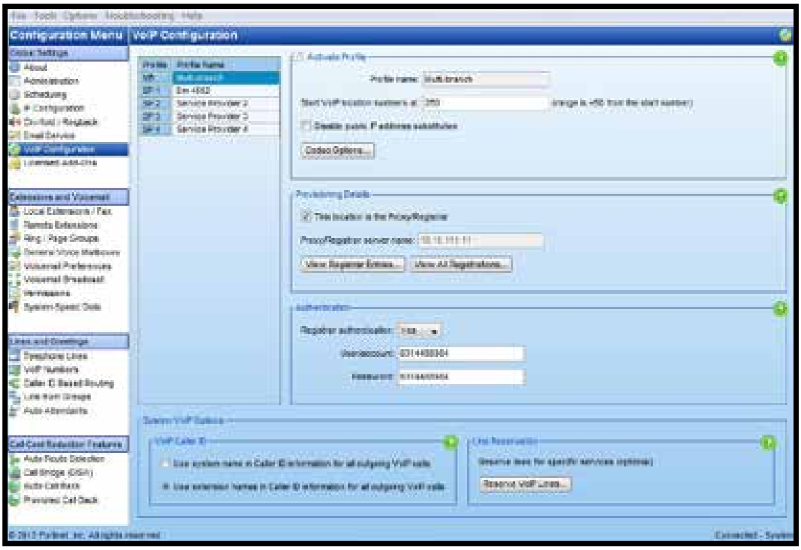
3 Configure the VoIP/SIP Server settings. Configuration Menu ▸ VoIP Configuration ▸ SP 1.

a Service Provider drop-down box, Select Manual Setup.
b Fill in the Profile Name.
c Under Codec Options make sure G.711u is selected and set as the preferred codec.
d Fill in the Proxy Server Name, Registrar Server Name, Outbound Proxy, and Realm/Domain fields with the SIP Servers IP address or domain name. Typically this is the LAN IP address of the Optimum Business SIP Trunk Adaptor. Note: for Non-registration Static mode, only fill in the Proxy Server and Realm/Domain to prevent the Talkswitch from registering).

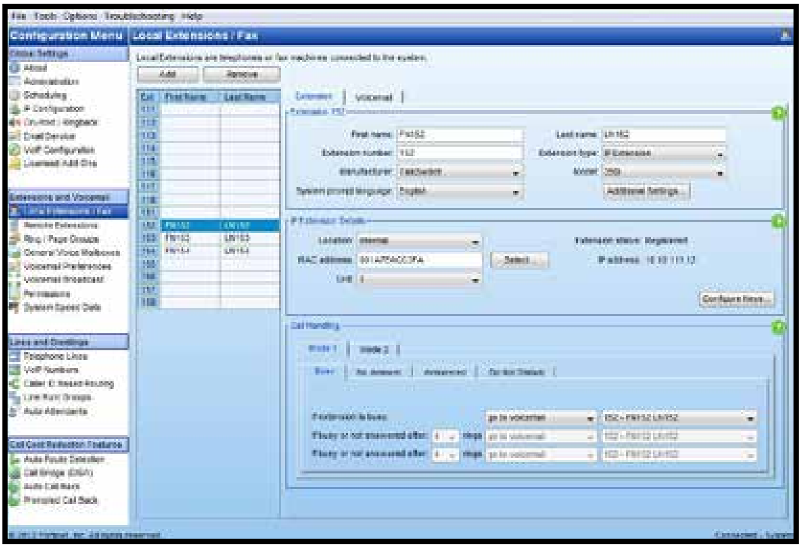


4 If setting up registration to the Optimum Business SIP Trunk Adaptor, configure the Talkswitch profile as well. Check the “This location is the Proxy/Registrar”, and configure the Authentication fields with the register information. Under the user/account name and password enter the Pilot DID, for this example we used 6314488984.

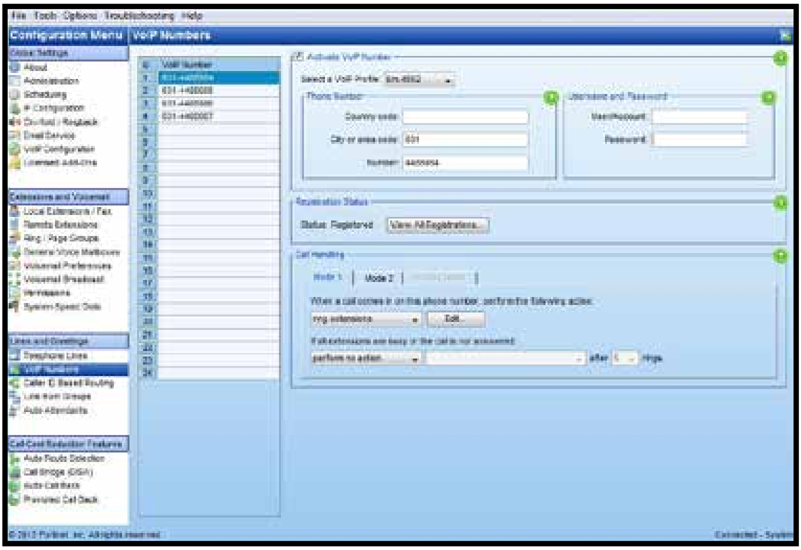
The user/account and password must match what was configured in the Optimum Business SIP Trunk Adaptor. This was step 3 of the Optimum Business SIP Trunk Set-up Guide.



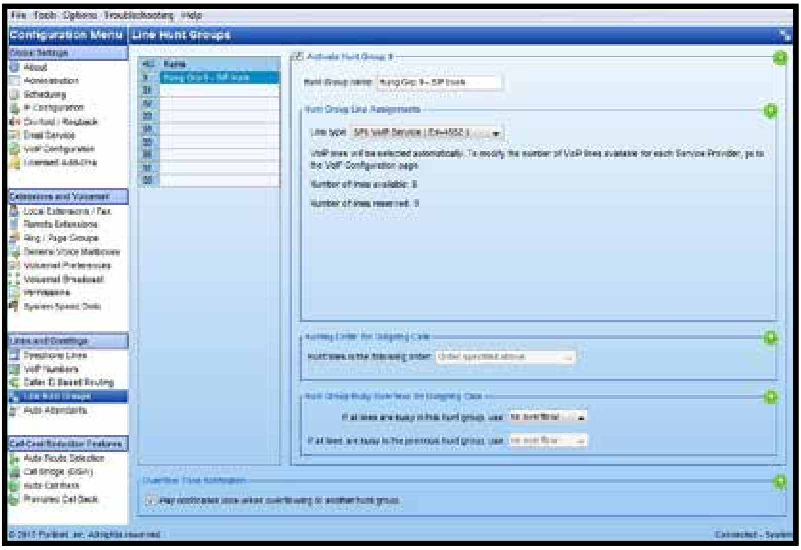
5 Set up each extension, Extensions and Voicemail ▸ Local Extensions / Fax. Fill in the MAC address of the phone and give the Extension a First and Last Name.



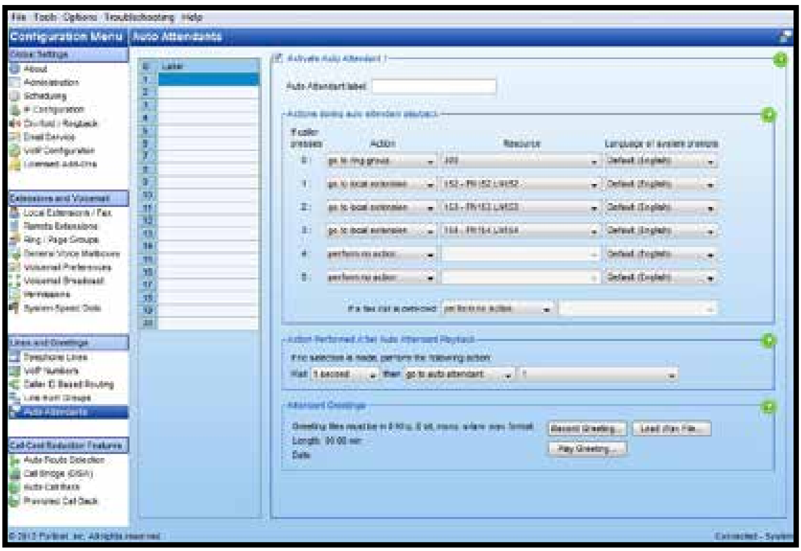
6 Set up the Incoming Call Routing, Lines and Greetings ▸ VoIP Numbers. Configure each line with a DID and set it to an Extension. Also set up one of the DID's to go to the Auto Attendant.



7 Set up the outbound Trunk. Lines and Greetings ▸ Line Hunt Group.
Check the box on top "Activate Hunt Group 9".



8 Configure the Auto Attendant. Lines and Greetings ▸ Auto Attendants (Menus). Next to each number that will be pressed select “go to local extension” as the Action and select the extension number as the Resource. Then select “go to auto attendant” as the action if no selection is made and enter the desired time to wait.



Due to the Cablevision DTMF network requirements, the DTMF tone duration generated by the phones and/or PBX may need to be increased from the default value of 180ms-200ms to 600ms. Make sure to check this setting in the IP PBX and/or Phones.