October 7, 2017

Obihai OBi202/200 ATA Setup Guide For Freephoneline (FPL) without Server Failover

version 1.43

by Webslinger on the Redflagdeals forums

I rewrote sections of my original guide to eliminate a potential issue arising from server failover. With all firmware versions up to and including 3.1.1 (Build: 5695EX), if your ATA boots up or is turned on without internet access being established first (modem and router haven't fully booted up or Internet Service is down), then the ATA will continually attempt to register with freephoneline.ca, which is not a SIP server. That's a website. Even when internet service has been established, afterwards, the ATA will not self-correct and attempt to register with voip.freephoneline.ca, voip2.freephoneline.ca, or voip4.freephoneline.ca: 6060. The ATA will just continually attempt to register with the website, freephoneline.ca, over and over again. Or the ATA may attempt to register with 10.0.0.1. The only fix is to force the ATA to reboot once internet access has been established. This potential problem occurs when Local DNS records and X_ProxyServerRedundancy are enabled in the ATA without internet access being available. Given that people are more likely to encounter a time when their ATA boots up before internet access has been established than encountering a specific Freephoneline SIP server that's down (most are up 99% of the time at least over the past two years), for most users setting up server failover in the ATA is not worthwhile due to the potential registration issue I'm describing here. Consequently, I have removed the server failover section from this guide. For those that know what they're doing, they can refer to the previous Server Failover version of this guide (versions before 1.30) found elsewhere. *This version of the guide that you're currently reading is safer to use.*

This guide was created while using OBi202/200 firmware 3.1.0 (Build: 5285) and is based on your ATA being at factory default settings. 3.1.1 (Build: 5695EX) will also work. Freephoneline's recommended settings can be found at
http://support.freephoneline.ca/hc/en-us/articles/212430746-VoIP-Unlock-Key-Credentials.

If you just want to skip to setting up your ATA immediately, skip to page 11. If you've used Obitalk.com previously to setup Freephoneline, skip to page 9 instead. But I do recommend you read everything, regardless.

Preamble

1. **Typically it's far better to have your own router with strong QoS functions and a restricted cone NAT firewall, disable whatever SIP ALG feature is enabled in the router, and stick whatever modem/router combo your ISP gives you into bridge mode.** These router combos issued by ISPs frequently have faulty (and hidden) SIP ALG/SPI features enabled with no way for the customer to disable them without getting a technical representative from his or her ISP to turn this feature off. Quite frequently, the first representative you speak to will have no idea how to accomplish this, much less know what SIP ALG is. Someone may try to enable DMZ in your modem/router combo or port forward; doing either is a huge security risk. Be aware if you reset your modem or when your ISP pushes a new firmware update to your modem/router combo, SIP ALG may be enabled again by default (and, therefore, it's simply better to have your own router with SIP ALG disabled in it).

To understand why SIP ALG often causes horrible problems, please visit
http://www.voip-info.org/wiki/view/Routers+SIP+ALG.

> "**SIP ALG problems**
> The main problem is the poor implementation at SIP protocol level of most commercial routers and the fact that this technology is just useful for outgoing calls, but not for incoming calls.

Lack of incoming calls
When a UA is switched on, it sends a REGISTER to the proxy in order to be localizable and receive incoming calls. This REGISTER is modified by the ALG feature (if not, the user wouldn't be reachable by the proxy since it indicated a private IP in REGISTER "Contact" header). Common routers just maintain the UDP "connection" open for a while (30-60 seconds) so after that time the port forwarding is ended, and incoming packets are discarded by the router. Many SIP proxies mantain the UDP keepalive by sending OPTIONS or NOTIFY messages to the UA, but they just do it when the UA has been detected as NATted during the registration. A SIP ALG router rewrites the REGISTER request so the proxy doesn't detect the NAT and doesn't maintain the keepalive (so incoming calls will be not possible).

Breaking SIP signalling
Many of the actual common routers with inbuilt SIP ALG modify SIP headers and the SDP body incorrectly, breaking SIP and making communication just impossible. Some of them do a whole replacing by searching a private address in all SIP headers and body and replace them with the router public mapped address (for example, replacing the private address if it appears in "Call-ID" header, which makes no sense at all). Many SIP ALG routers corrupt the SIP message when modifying it (i.e. missed semi-colon ";" in header parameters). Writing incorrect port values greater than 65536 is also common in many of these routers.

Disallows server side solutions
Even if you don't need a client side NAT solution (your SIP proxy gives you a server NAT solution), if your router has a SIP ALG function that is enabled and breaks SIP signalling, SIP ALG will make communication with your proxy impossible."

2. **If you have your own router, ensure that whatever modem/router combo your ISP gave you is in bridge mode.** Contact your ISP if need be. For Bell Hubs, visit
http://forums.redflagdeals.com/please-sticky-how-bypass-bell-hub-use-your-own-router-1993629/

3. When buying a router for VoIP, ensure you buy one that does **not** have a full cone NAT.
Visit https://www.think-like-a-computer.com/2011/09/16/types-of-nat/.

Mango from the Obitalk.com forums writes,

"Use a restricted cone NAT router, and do not use port forwarding or DMZ. Restricted cone NAT will only permit inbound traffic from the service provider you're registered to. If you have a full cone NAT router, it will allow traffic from any source. This is probably not what you intend.

If you have a Windows computer, you can test your router using the utility here:
http://www.dslreports.com/forum/remark,22292023.To run it, use stun stun.ekiga.net from a command prompt."
Essentially, you download the stun-test.zip file; extract the stun.exe file from within the zip file to an easily accessible location; use an elevated command prompt (visit http://www.thewindowsclub.com/how-to-run-command-prompt-as-an-administrator); change directory (cd) to the directory or location where you extracted stun.exe (visit http://www.digitalcitizen.life/command-prompt-how-use-basic-commands); and type "stun stun.ekiga.net" without the quotation marks followed by the enter/return button on your keyboard.

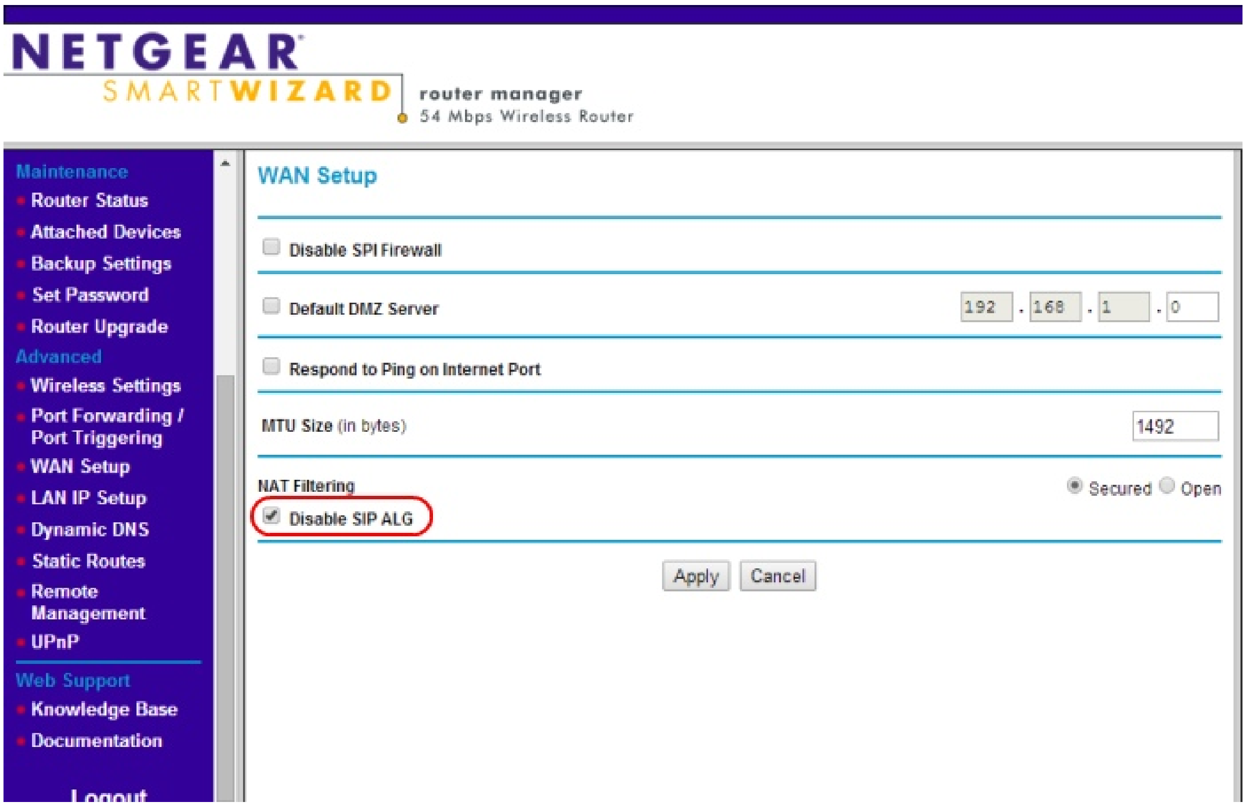
Asus routers, at the time of this writing, produce port restricted cone NAT routers, for example and are fine, provided you're using one with Asuswrt-Merlin, third party firmware installed: https://asuswrt.lostrealm.ca/about.

4. It's best to have a decent router for VoIP with strong QoS features.
Stick your ISP's modem in bridge mode, use your own router, and properly enable QoS for your ATA (if you're going to use adaptive QoS, give your ATA the highest priority for internet traffic and assign lower priorities for all other devices on your LAN). Refer to your router's manual.

I'm not a big fan of this site, but for a general QoS description, visit http://www.voipmechanic.com/qos-for-voip.htm (avoid anything this site says about the G.729 codec because you really don't want to be using this low bitrate codec unless you're using Freephoneline on a smartphone with a poor cellular data signal).

5. Now that you understand why SIP ALG is so horrible, disable it in your own router, unless you own an Asus router that has Asuswrt-Merlin (or a router with XWRT-Vortex) installed on it. **At the time of this writing, Asuswrt-Merlin, third party firmware for some Asus routers, works fine with Freephoneline with “SIP Passthrough + NAT helper” enabled. SIP Passthrough is the name for Asus’ SIP ALG feature.**

Visit https://www.obitalk.com/info/faq/sip-alg/disable-alg.



As of this writing, Apple routers don't offer this feature, but you might as well check. If you manage to disable SIP ALG in the router, reboot it.

DLINK router users may need to log into the admin page of their router, click the "Advanced" tab and then "Firewall Settings", navigate to "Application Level Gateway (ALG) Configuration", and uncheck SIP: Visit http://www.support.dlink.com/emulators/dir615_revC/310NA/adv_dmz.htm.

If you received a modem/router combo, from your ISP ask your ISP. It is typically better to stick the modem/router combo from your ISP in bridge mode and use an external router.

Save settings.

Turn off both router and ATA. Turn on router. Wait for router to be fully up and transmitting data before turning on your ATA. Turn on ATA.

6. Thanks to Mango, many of us now understand that in order for ATAs to remain registered and working properly with a VoIP SIP provider like Freephoneline, in particular after power failures, the following conditions must be met:

UDP Unreplied Timeout (in your router) < NAT Keep-alive Interval (in your ATA; for Obihai ATAs this is X_KeepAliveExpires) < UDP Assured Timeout (in your router) < SIP Registration Failure Retry Wait Time (or RegisterRetryInterval in Obihai ATAs)

“<“ means less than.

When a modem leases a new IP address, a problem can arise where prior associations using the old IP address are maintained in the router. When the ATA attempts to communicate using the old IP address, the response is unreplied, and then if the UDP Unreplied timeout is greater than the Keep Alive Interval (and UDP Unreplied timeout is often set to 30 by default in consumer routers) a problem arises where the corrupted connection persists. If UDP Unreplied timeout is, for example, 10, and the NAT Keep Alive Interval is 20, then the corrupted connection will timeout or close. A new connection will be created, and everything will work fine.

Another problem can occur when the Keep-Alive interval is greater than UDP Assured Timeout (often 180 by default in consumer routers): the NAT hole will close due to the ATA not communicating frequently enough with the SIP server. In turn, incoming calls may, intermittently, not reach the ATA. Again, X_Keepalives expires is supposed to be 20 with FPL.

Getting access to both UDP Unreplied Timeout and UDP Assured Timeout settings in consumer routers may be difficult, if not impossible. Asuswrt-Merlin, third party firmware for Asus routers, does offer easy access to these two settings, which are found under General–>Tools-->Other settings. In part, for this reason, I tend to use Asus routers. However, my understanding is that third party Tomato firmware has these two settings as well. So if your router supports Tomato firmware, that may be another option. Note that I will not be held accountable any damage resulting from failed firmware updates.

The keep alive interval for FPL is 20. The SIP Registration Failure Retry Wait Time is 120. I use 10 for UDP Unreplied Timeout and 117 for UDP Assured Timeout.

Hardware Specific Issues

A. **Netgear R7000 routers** (Note that I will not be held accountable any damage resulting from failed firmware updates.)

If you have a Netgear R7000 router, you may need to install third party XWRT-Vortex firmware. I recommend doing this anyway to obtain easy access to both UDP Unreplied and UDP Assured timeout settings. Afterwards, turn off the router and the ATA. Turn on the router. Wait for it to be fully up and running (including Wi-Fi). Then turn on the ATA. Download XWRT-Vortex here: http://xvtx.ru/xwrt/download.htm. In your router, navigate to Advanced Settings–>WAN–>NAT Passthrough–>SIP Passthrough. Change SIP Passthrough to “Enabled + NAT helper.” Click “Apply.”

B. **Nettis 4422 modem** from Carry Telecom (click the "Internet" tab) http://www.carrytel.ca/support.aspx.

Q : DSL - My VoIP phone does not work with Netis 4422 modem.

A: Please download the newest Netis firmware file at www.carrytel.ca/download/netis1228.zip and update the firmware file netis1228.img for your modem. The new firmware has been tested and working with most of Voip phone providers

C. **Asus VLAN**

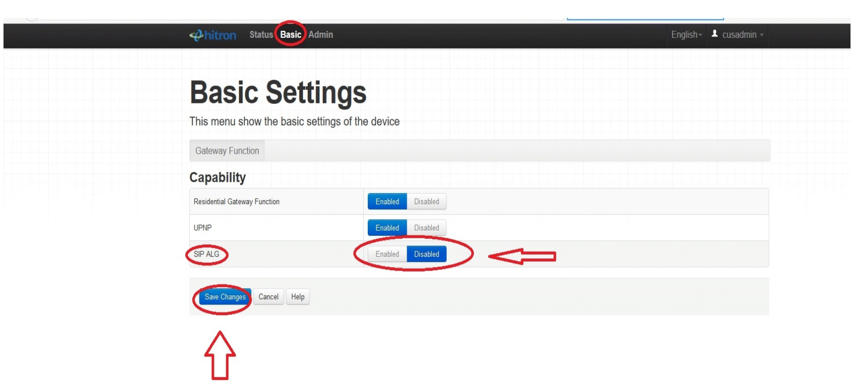
A number of people have been trying to eliminate Bell Hubs from their setups by using this guide: http://blog.ngpixel.com/post/104449747538/how-to-bypass-bell-fibe-hub-and-use-your-own-router.
At the time of this guide being written, NAT acceleration must be disabled in this setup in order for SIP services, including Freephoneline, to work properly. In your router, navigate to Advanced Settings-->LAN-->Switch Control-->NAT Acceleration. Select "disable." Click "apply."Then reboot your modem, router (wait for Wi-Fi SSIDs to populate first before rebooting ATA), and your ATA, in that order.

To determine whether you need NAT Acceleration enabled, visit https://routerguide.net/nat-acceleration-on-or-off/. If you do require NAT Acceleration to be enabled, don’t use VLAN with Asus routers.

D. **Hitron CGN series gateway modem/router combos (from Rogers, Shaw, or another ISP) or any modem/router combo from any ISP with SIP ALG forced on**

If you don’t have your own router, and if you can’t get someone from Rogers or your ISP to disable SIP ALG for you in their modem/router combo, your ATA may need to register with voip4.freephoneline.ca:6060. The purpose of voip4.freephoneline.ca:6060 is to help circumvent SIP ALG. So, if you’re experiencing one-way audio issues as a result of SIP ALG, this is the SIP server to try.

E. **Hitron CODA-4582 series gateway modem/router combo from Rogers (and possibly other ISPs)**
Open your web browser, and login at 192.168.0.1. Default username is cusadmin.
Select the “Basic” tab and disable “SIP ALG.” Click the “save changes” button.



Potential Problems With Setting up Freephoneline Using Obitalk.com

Configuration issues

A. Pianoguy on the Redflagdeals forum notes that "the OBi's default keep-alive message is literally keep-alive which doesn't garner a response. If you have a newer firmware version on your OBi200 (not available on OBi1) you can try the following:

Navigate to Voice Services-->SP(FPL)

X_KeepAliveEnable: Checked
X_KeepAliveExpires: 20
X_KeepAliveMsgType: notify"

At the time of this writing, if you're using Obitalk.com to provision Freephoneline, X_KeepAliveExpires will be set to 15, and X_KeepAliveMsgType will not be set properly.

So, if you're going to use Obitalk.com to provision Freephoneline, change these settings to the above recommended values.

B. Navigate to Service Providers-->ITSP Profile (FPL)-->RTP

KeepAliveInterval: 20

C. Navigate to Service Providers-->ITSP Profile (FPL)-->SIP

X_UsePublicAddressInVia: enabled

To help avoid one-way audio issues with Freephoneline, enable this setting.

D. Navigate to Service Providers-->ITSP Profile (FPL)-->SIP

Uncheck the box under the Value column to disable X_Use302ToCallForward

Freephoneline requires calls to be bridged if you want your ATA to forward calls.
If X_Use302ToCallForward is enabled, calls that are forwarded by the ATA (as opposed to using Freephoneline's Follow Me feature) will be dropped to voicemail

E. Navigate to Voice Services-->SP(FPL) Service

X_UserAgentPort would be better as a random port number between 30000 and 65535. Just pick a port number in that range.

By using a high random port you help to thwart SIP scanners and may also circumvent a faulty SIP ALG feature in your router.

F. Navigate to Service Providers–>ITSP Profile (FPL)–>General

DigitMap: (1xxxxxxxxxx|011XX.S3|[2-9]xxxxxxxxx|*98|[2-9]11)

Here’s an alternative example:
(1xxxxxxxxxx|011XX.S3|[2-9]xxxxxxxxx|[2-7]11|<811:8667970000>|*98|911)

The digitmap is appropriate for Toronto. For 811 you will need to look up the corresponding phone number for your area and replace the phone number after the colon. Note that dialing 811 in this example reaches Telehealth Ontario. https://www.fongo.com/government-service-numbers/

<1> means add or prepend 1 to the beginning of the phone number
<:1> also means add 1
<1:> means remove 1 from the beginning (and replace it with what follows after the colon)
[2-9] means match any single digit from 2 to 9
XX. means match any phone number you dial (also XX. is an indefinite variable, and without XX.S3, for example, your ATA will wait 10 seconds for you to finish entering in a phone number before dialing out). S3 reduces the wait (represents 3 seconds).
011XX. means any number starting with 011 (for international dialing), and again XX is an indefinite variable. Without the .S3, your ATA will wait 10 seconds.
011xxxxxxxxxxx. is wrong if the international phone number is less than 11 digits.

Mipd is for IP dialing.
[^*#]@@. is for SIP URI.
Neither is needed with Freephoneline. They should be removed.

[6-7]x*xxxxxxxxxxx. listed in another guide should never be used since it can't logically apply to anything.

XX. is usually not needed (and actually, inadvisable, since it can apply to anything and, due to it being an indefinite variable creates a 10 second timeout in the ATA while it waits for the user to finish entering a phone number).

*98 is for voicemail, which works with Freephoneline, by default.
For more information on Obihai Digitmaps, visit
http://www.obihai.com/docs/OBi-DigitMapCallRoute-Tutorial-v1-1.pdf

G. Navigate to Codecs–>Codec Profile (A or whatever the VoIP service you're using is assigned to . . . you can determine this under Voice services-->SP[freephoneline] Service-->X_CodecProfile)

i. Uncheck the default boxes for Enable (Fax Event) and T38ECM

ii. Check or enable the boxes under the Value column for Enable (Fax Event) and TC8ECM.

T.38 Fax protocol works with an OBi200 or OBi202 and Freephoneline.

H. Star Code Profiles (A & B)

Code28: *99, Blind Transfer, coll($Bxrn)

Blind Transfer can't be *98 with Freephoneline. *98 is meant for voicemail. Change Blind Transfer to *99 or something other than *98. Obitalk's preset configuration doesn't have this set properly for Freephoneline.

How do I stop Obihai from updating firmware without my permission and stop using Obitalk.com?

If you don't disable Obitalk Service (in addition to other settings), Obihai can remotely upgrade firmware on your ATA without your permission, in the middle of a storm with power flickering, while you're out somewhere. If the power goes out in the middle of a firmware upgrade, you run the risk of wrecking your ATA. Do you want a bricked device? I don't. An uninterruptible power supply (UPS) is a good idea, by the way.

If firmware can be upgraded remotely, Obihai has access to your ATA and can make changes. That's fine if you need technical support. It's not fine if you're into privacy.

After 1 year is up, in order to continue using Obitalk.com to upgrade firmware, you'll need to pay a $10 USD annual fee. However, you don't need to pay if you upgrade firmware manually. Best to learn now how to do it manually, unless, of course, you like paying annual fees.

There's two ways to update firmware for free even if the ATA is out of warranty:

A. Dialing ***6 (and then pressing "1" if an update is available)

http://www.obihai.com/docs/OBiProvisioningGuide.pdf (page 15)

This is a faster method than using the Obitalk web portal since you don't have to log into anything.
If no update is found for you, you'll have to do step B.

B. Manually updating firmware via the device:
http://www.obihai.com/docs/OBiDeviceAdminGuide.pdf (page 43)

Instructions are also found after visiting http://www.obihai.com/docs-downloads and clicking on "OBi Device Firmware."

Links to the latest OBi20x firmware can usually be found over here: https://www.obitalk.com/forum/index.php?topic=9983.0. Navigate to the last page.

Obihai does not frequently release updated firmware release notes. Older ones can be found here: https://www.obitalk.com/forum/index.php?topic=8982.0.

New firmware releases offer new options that sometimes aren't quickly added to the web portal. Consequently, it's best to stop relying on the web portal. You can use the Obitalk.com portal for initial setup (and keep in mind that you must use the web portal to activate Google Voice on your ATA), but eventually, people should learn to stop using it, in my opinion--unless, of course, you want to use OBiExtras ($4.99/month USD service). I don't.

To ensure Obihai and a ITSP provider can't, without your permission, update your device's firmware remotely, you must do the following (all three steps):

1. Dial ***1, and enter the IP address you're told into your web browser

2. Navigate to System Management-->Auto Provisioning

a) For Auto Firmware Update ---> ensure method is disabled

b) For ITSP Provisioning ---> change method to disabled (uncheck default box)

c) For OBiTalk Provisioning--->change method to disabled (uncheck default box)
Keep in mind changes made in Obitalk.com web portal will no longer transfer to your device after this change is made.

(submit)

3. Navigate to Voice Services-->OBiTALK Service-->Obitalk service Settings
a) uncheck enable and uncheck the default box
You will not be able to call other Obihai devices using their respective Obitalk numbers after disabling ObiTALK Service. Also the OBi Echo test will no longer work if you disable OBiTALK Service. If you email or send a support request to Obihai, Obihai Support will note that your ATA appears to be offline, and that they can't make adjustments to help you.

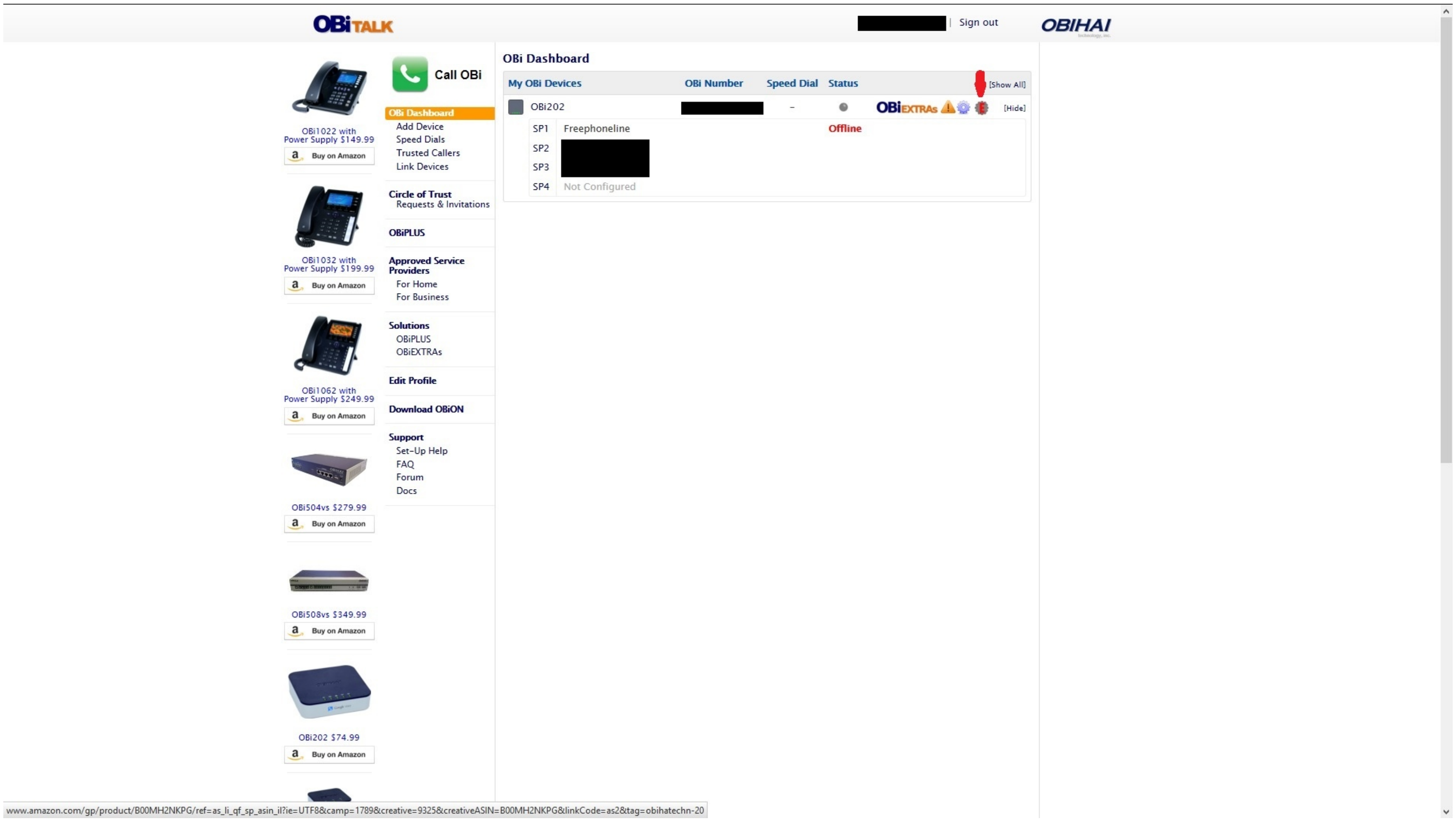
(submit/reboot)

<u>If you want to use Obitalk.com to make changes to your ATA, both Obitalk Service and Obitalk Provisioning must be enabled.</u>

Credit goes to Pianoguy on Redflagdeals for teaching me about this issue.

Have you already setup your Obihai device with Freephoneline and want to make changes?

<u>If you used the OBitalk web portal to configure your ATA, you need to continue using www.obitalk.com for now. Enter the expert menu (advanced configuration; it's an "E" icon).</u>



That grey cog wheel with the "E" is for the expert configuration menu.

It appears when logging in at http://www.obitalk.com, selecting "Edit Profile" on the left, then scrolling down under "Advanced Options" and finally selecting "Enable OBi Expert Entry from Dashboard."

Otherwise, dial ***1, and enter the IP you're told into your web browser.

<u>If you use the Obitalk web portal (www.obitalk.com) to configure your ATA, keep in mind that you must continue using it to configure your ATA unless you disable Obitalk Provisioning first. Otherwise whatever settings you change will eventually be overwritten</u>

<u>(they will be transferred from your Obitalk.com account to your ATA) by what you previously entered at obitalk.com anyway. If you wish to disable this behaviour, dial ***1. Enter that IP address into a web browser. Navigate to System Management-->OBiTalk Provisioning-->select Disabled for the method. Save. Reboot ATA. Afterwards, obitalk.com won't overwrite whatever changes you make via the device's interface (via IP address).</u>

<u>Pick one method (obitalk.com) or the other (IP address of device) for changing device settings. But do not use both methods. Keep in mind that activating Google Voice requires using the Obitalk.com web portal if your Obihai ATA is using post-OAuth 2.0 firmware.</u>

<u>How to Setup your Obihai ATA with Freephoneline</u>
(There are 14 sections/steps.)

<u>If you used the Obitalk web portal (www.obitalk.com) to configure your ATA, keep in mind that you must continue using it to configure your ATA unless you disable Obitalk Provisioning first. Otherwise whatever settings you change will eventually be overwritten (they will be transferred from your Obitalk.com account to your ATA) by what you previously entered at obitalk.com anyway. If you wish to disable this behaviour, dial ***1. Enter that IP address into a web browser. Navigate to System Management-->OBiTalk Provisioning-->select Disabled for the method. Save. Reboot ATA. Afterwards, obitalk.com won't overwrite whatever changes you make via the device's interface (via IP address).</u>

<u>Pick one method (obitalk.com) or the other (IP address of device) for changing device settings. But do not use both methods.</u>

This guide presumes that you are NOT using the Obitalk.com web portal, but the vast majority of the same settings apply.

i. **Before beginning the steps below make sure whatever modem/router combo your ISP gave you is in bridge mode <u>if you are using your own router</u>.** Call/contact your ISP if you have to. For Bell Hubs, visit http://forums.redflagdeals.com/please-sticky-how-bypass-bell-hub-use-your-own-router-1993629/

1. Hookup your ATA (power cord, ethernet cable from router to ATA, attach phone). Refer to the setup guide that came with your ATA.

Or watch this Youtube video from Obihai: https://youtu.be/iydg5lY1UA8.

OBi202/202 start guides are also located here: http://www.obihai.com/docs-downloads.

2. For the OBi202 ATA, you may need to enable WAN access to the ATA first: https://www.obitalk.com/info/faq/OBi202-sec/Howto-Access-Web-from-WAN.

"By default, web access to the Internet (WAN) IP address of the OBi202 is disabled. Accessing the Internet (WAN) IP will show, "The requested URL was not found!" or "The web page cannot be found" in your browser. Note: You may always access the web page from the LAN side of the OBi202, e.g. connect your PC to the LAN port of the OBi202. To enable access to the OBi202,

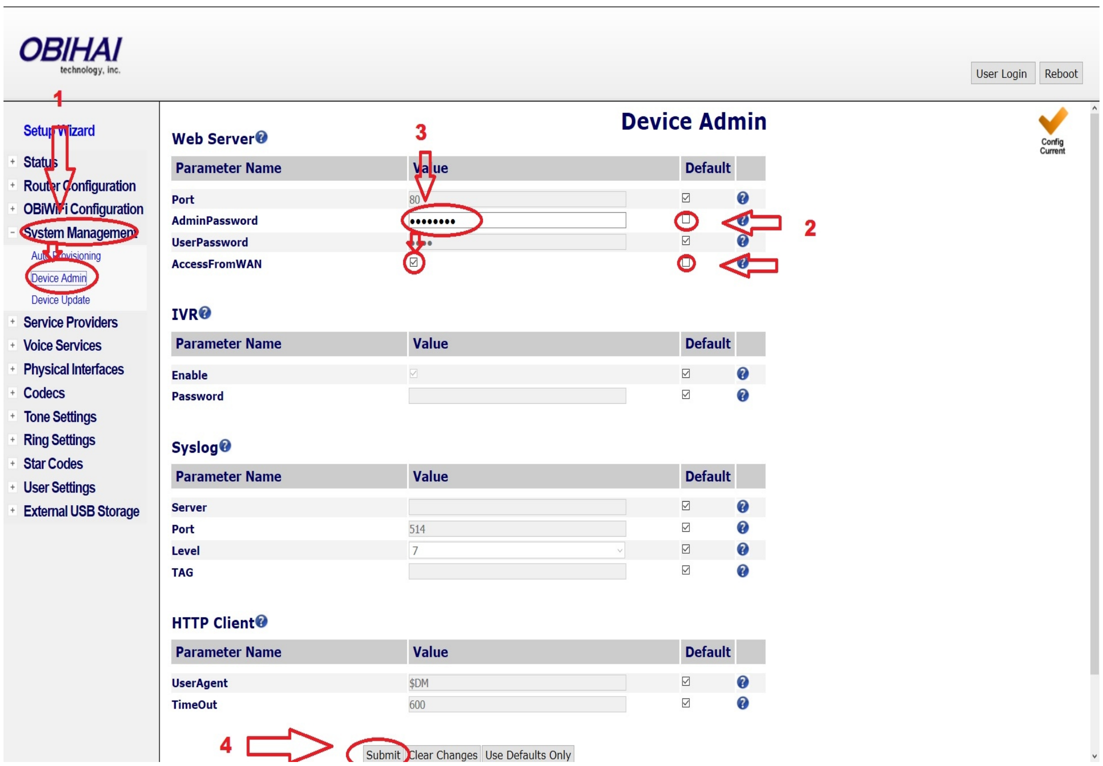
- Dial ***0 from the phone connected to the OBi202
- Enter 30#
- Press 1 to Enter a New Value
- Press 1# to Enable
- Press 1 to Save

To disable Internet (WAN) port access do the following:
- Dial ***0 from the phone connected to the OBi202
- Enter 30#
- Press 1 to Enter a New Value
- Press 0# to Disable
- Press 1 to Save

Alternatively, you may use the OBi Expert Configuration page via the OBiTALK web portal to enable WAN access.
- Sign in to www.OBiTALK.com (link)
- Select the OBi202 from the OBi Dashboard
- Go to: System Management - Device Admin
- In the Web Server area, you may enable AccessFromWAN by checking the available box."

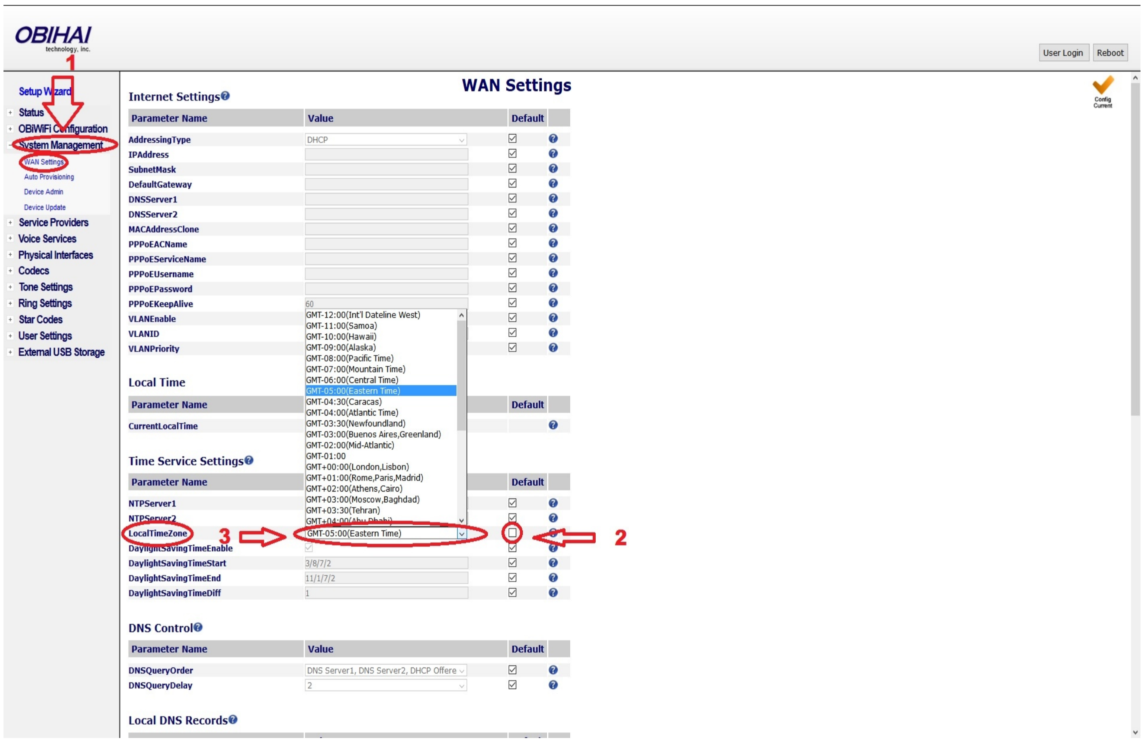
3. Dial ***1 on a phone connected to your ATA, and enter the IP address you're told into a web browser.

4. Login. The default username and password is admin. It's probably a good idea to change your admin password to something else. You can do this by navigating to System Management–>Device Admin–>Web Server–>Admin Password. Uncheck the default box on the right hand side. Type in a new password, and click the "Submit" button at the bottom. For security purposes, one should not use default passwords for devices that are connected to the internet.



Enabling access from WAN (refer to step 2) is only required if you can't access your ATA.

5. Set your time zone (Picture shown below is for an OBi200)



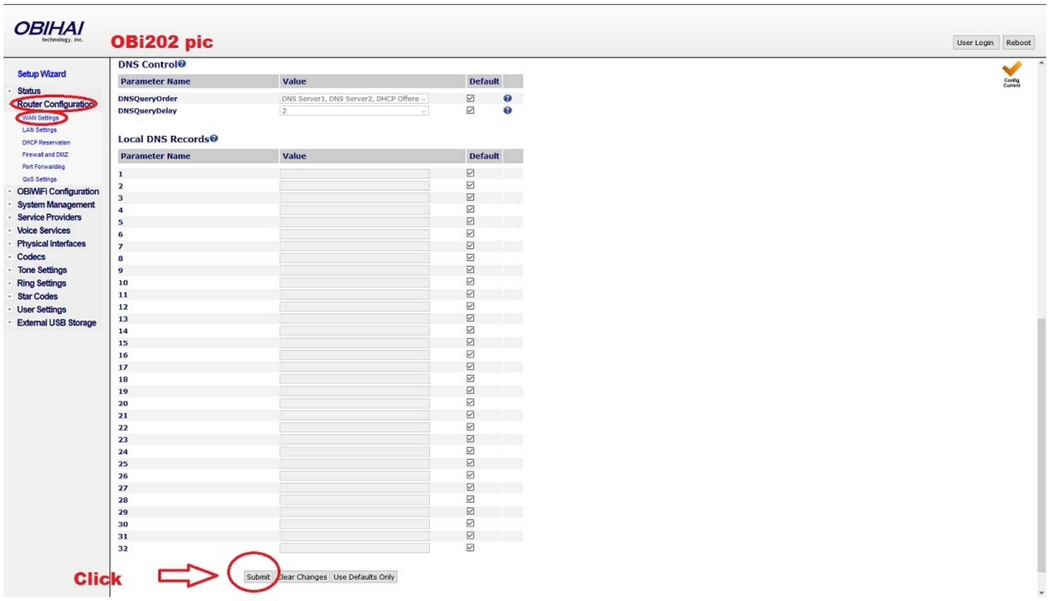
a. For OBi200, Navigate to System Management–>WAN Settings–>Internet Settings–>Time Service Settings–>LocalTimeZone

For OBi202, Navigate to Router Configuration–>WAN Settings–>Internet Settings–>Time Service Settings–>LocalTimeZone

The only difference here is the OBi202 setting is found under Router Configuration, whereas OBi200 owners need to look under System Management.

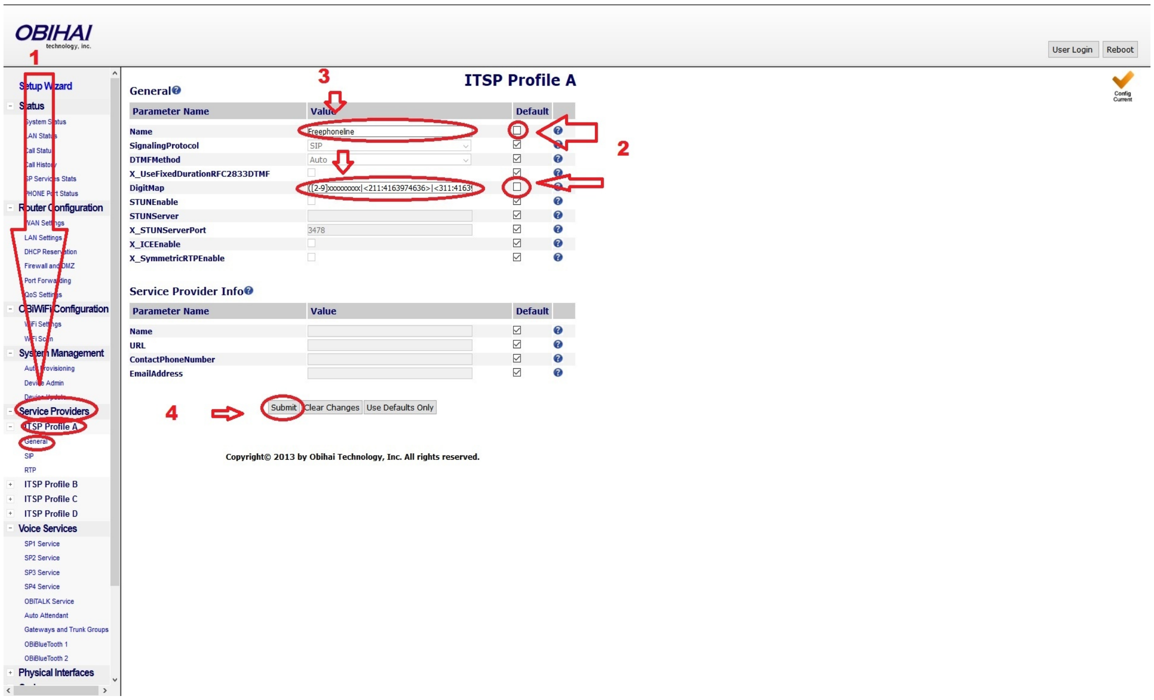
b. Uncheck the default box on the right

c. In the Value, dropdown box for LocalTimeZone, select your local time zone. Only select Eastern time zone if you’re located there.



d. Scroll down and click the “Submit” button (pic shown above is from an OBi202). Do NOT reboot your ATA.

6. Create a profile name and DigitMap for Freephoneline.



a. Navigate to (click on) Service Providers–>ITSP Profile A–>General

b. Uncheck the default boxes for “Name” and “Digitmap”

c. For Name, enter something descriptive (for example, Freephoneline).

d. For Digitmap, use

(1xxxxxxxxxx|011XX.S3|[2-9]xxxxxxxxx|*98|[2-9]11)

Here’s an alternative example:
(1xxxxxxxxxx|011XX.S3|[2-9]xxxxxxxxx|[2-7]11|<811:8667970000>|*98|911)

The digitmap is appropriate for Toronto. For 811 you will need to look up the corresponding phone number for your area and replace the phone number after the colon. Note that dialing 811 in this example reaches Telehealth Ontario. https://www.fongo.com/government-service-numbers/

<1> means add or prepend 1 to the beginning of the phone number

<:1> also means add 1

<1:> means remove 1 from the beginning (and replace with what follows after the colon)

[2-9] means match any single digit from 2 to 9

XX. means match any phone number you dial (also XX. is an indefinite variable, and without

XX.S3, for example, your ATA will wait 10 seconds for you to finish entering in a phone number before dialing out. S3 reduces the wait (represents 3 seconds)

011XX. means any number starting with 011 (for international dialing), and again XX is an indefinite variable. Without the .S3, your ATA will wait 10 seconds

011xxxxxxxxxxx. is wrong if the international phone number is less than 11 digits.

Mipd is for IP dialing
[^*#]@@. is for SIP URI
Neither is needed with Freephoneline. They should be removed.

[6-7]x*xxxxxxxxxxx. listed in another guide should never be used since it can't logically apply to anything.

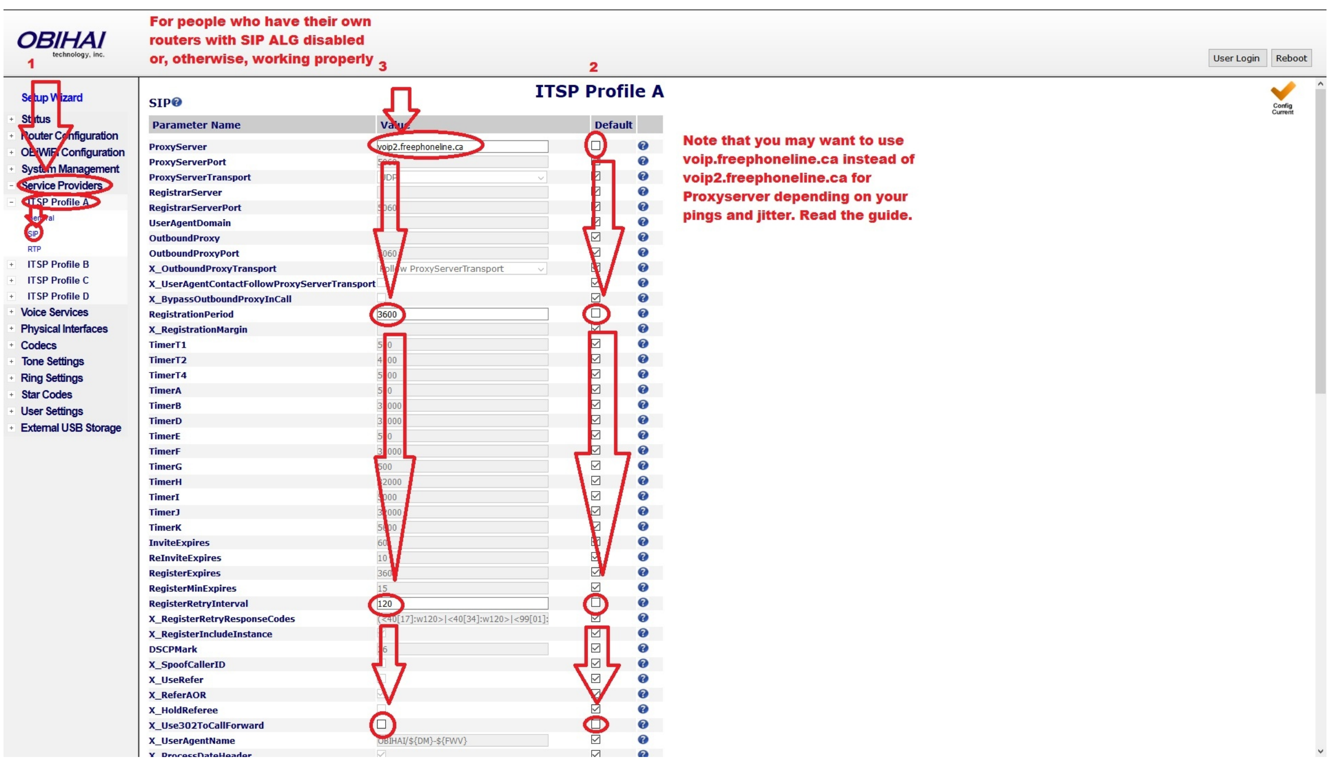
XX. is usually not needed (and actually, inadvisable, since it can apply to anything and, due to it being an indefinite variable creates a 10 second timeout in the ATA while it waits for the user to finish entering a phone number).

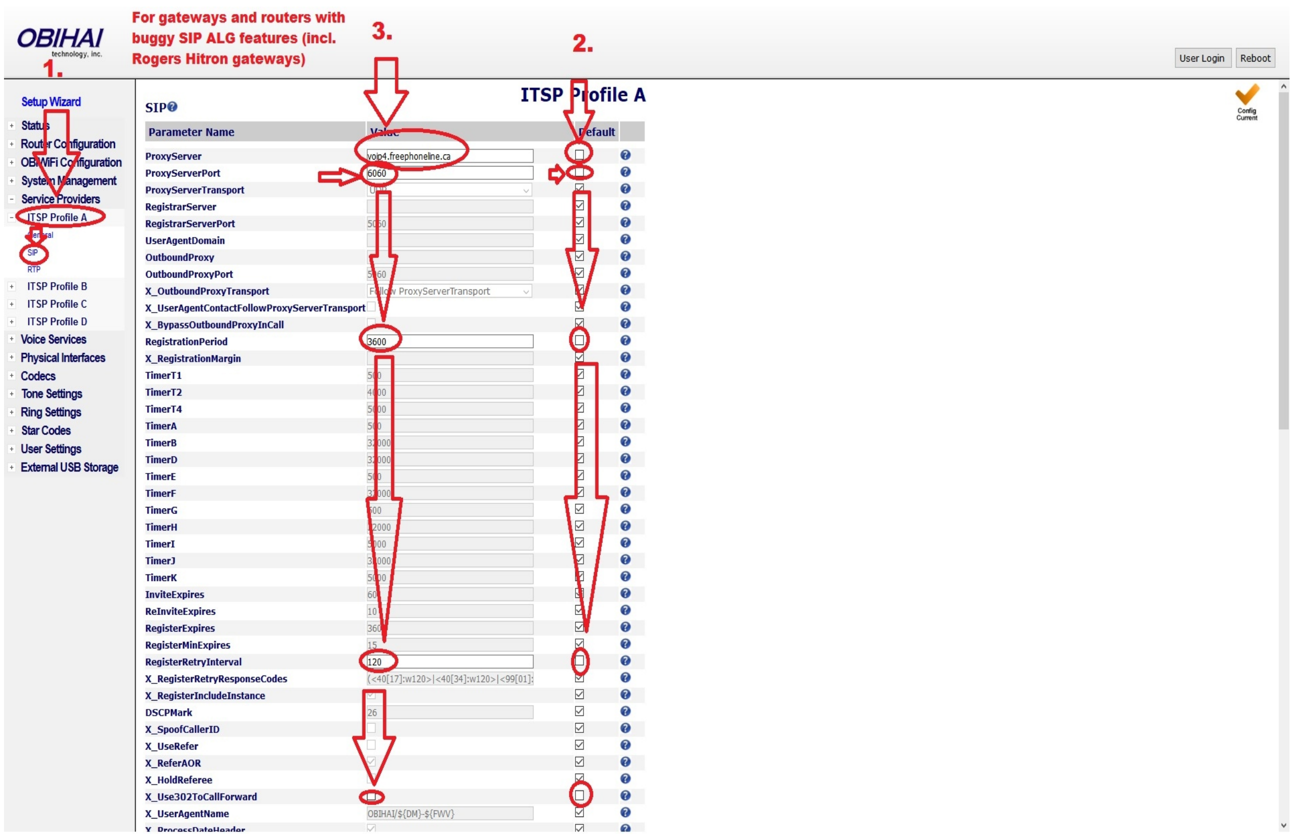
*98 is for voicemail, which works with Freephoneline, by default.

For more information on Obihai Digitmaps, visit
http://www.obihai.com/docs/OBi-DigitMapCallRoute-Tutorial-v1-1.pdf

e. Click the “Submit” button, but do NOT reboot the ATA yet.

7. Set Proxyserver and ITSP Profile settings for Freephoneline





a. Navigate to Service Providers–>ITSP Profile A–>SIP

b. Uncheck default boxes for Proxyserver, RegistrationPeriod, RegisterRetryInterval, and X_Use302ToCallForward.

c. For Proxyserver pay close attention below:

i) If you a) don't have your own router with SIP ALG disabled ***AND*** *b) are using a modem/router combo that has SIP ALG or Stateful Packet Inspection (SPI) enabled with no way for you to disable it (for example****, if you're using Rogers' Hitron modem/router combos that are older than the CODA-4582 series****) you will want to use voip4.freephoneline.ca:6060 only. That is, if you're experiencing one-way audio issues when your ATA is registered with voip.freephoneline.ca or voip2.freephoneline.ca, you're going to want to try voip4.freephoneline.ca:6060.*

If the underlined section applies to you, enter voip4.freephoneline.ca for ProxyServer. Also uncheck the default box for ProxyServerPort and enter 6060. You can skip ahead to step 7d on page 19.

ii) If you have your own router that was not issued by your ISP that has SIP ALG either disabled or working properly in it, then

test pings and jitter (you want little to no variation between pings) to the specific Freephoneline SIP servers you plan on using.

Use winmtr: http://winmtr.net/download-winmtr/. Ping about 100 times to each server.

My pings to

-voip.freephoneline.ca average 11 ms.
-voip2.freephoneline.ca average 12 ms
-voip4.freephoneline.ca average 27 ms

If you're using a Macintosh, maybe this helps:
https://www.reddit.com/r/TagPro/comments/2j6qx7/how_to_run_an_mtr_on_mac/

When using WinMTR, look at the very last hop or line. Look at your average ping and then maximum ping. Although WINMTR doesn't provide a jitter value, you can get an idea of what yours is by subtracting maximum ping from your average. Jitter is the difference between each successive ping. The bigger the difference, the bigger the problem.

Same with ping, which represents lag or delay. The lower your ping and jitter, the better.

You do not want high pings and lots of jitter (you do not want a lot of variation between each ping). If you get horrible results (pings over 200ms), to any server, you probably don't want to use that server. So you would want to give that server the lowest priority.

I get between 11 (voip.freephoneline.ca and voip2.freephoneline.ca)-24ms (voip4.freephonline.ca) on average, depending on the server I'm testing to. Preferably, you want pings below 100ms.

Anything over 200ms is unacceptable.

What you don't want to see is 40, 45, 50, 35, **500**, 40, 30, 45, **700**. That's bad jitter.
You want relatively consistent pings without a lot of variation.

One reason why jitter can occur is due to other devices on your LAN (local area network) using bandwidth. That's why properly enabling QoS in your router for your ATA is always a good idea. Refer to point 4 in the Preamble.

Bad jitter can produce broken up audio or choppiness during phone calls. Severe jitter can cause calls to drop. Ping affects delay.

I recommend testing pings/jitter between 8 p.m. and 11 p.m. to see if local congestion is a factor (this often is your ISP's fault). Sundays are the best days to test (because that's when most people in your area will be home). 8 p.m. - 11 p.m. is prime time. During prime time (between 8 p.m. and 11 p.m.) cable internet nodes may be oversubscribed in your area and face congestion issues (and congestion can also exist with DSL). So I suggest testing services between 8 p.m. and 11 p.m., particularly on Sundays, when everyone in your area will be home.

Ping is a measurement of data packet transmission, and ping does affect delay or lag. All gamers know, almost inherently, that lag affects them negatively. A PC gamer will pound his or her keyboard in hope that a character will respond on his or her monitor, quickly, but when there's a delay or lag, reality doesn't meet expectation. A gamer can see this problem visually. Over VoIP, anything over 200-210 ms, you will typically start to encounter crosstalk due to increased delay, even if the untrained ear doesn't notice. All VoIP services are subject to the same scientific principles including the fact that speed of transmission affects delay, and Freephoneline is not some magical service that is somehow exempt from issues arising from high pings and jitter.

When pings and, especially, jitter are high, it's a pretty horrible experience, just as it would be with any other VoIP service. When pings and jitter are fine, Freephoneline is great.

Lastly, anyone using any communication service (or even when playing online games or using other online services) should understand that the longer the path to the server being used, the greater the potential exists for a problem to occur somewhere along that path. Freephoneline's SIP servers are located in Ontario.

iii) My pings/jitter with voip2.freephoneline.ca tend to be better than those from voip.freephoneline.ca, which, in turn, are better than pings/jitter with voip4.freephoneline.ca. You can use voip4.freephoneline.ca:6060 if you get better results using it even if SIP ALG works perfectly fine for you. That is, even if you don't have a Rogers Hitron gateway (or modem/router combo), for example, you can still use voip4.freephoneline.ca:6060. However, the reason that server exists is to help those who have SIP ALG issues, and faulty SIP ALG functions are not exclusive to Rogers Hitron gateways. Broken SIP ALG features exist in other modem/router combos as well (and in consumer routers).

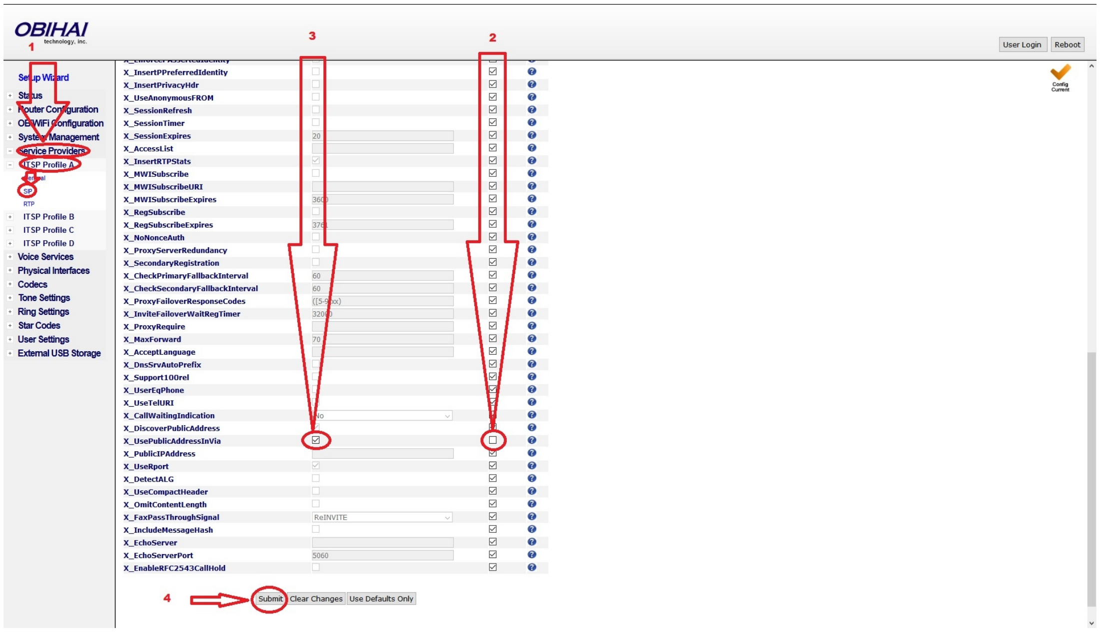
For ProxyServer, enter the server you get the best results from (choose from voip.freephoneline.ca, voip2.freephoneline.ca, and voip4.freephoneline.ca). Note that if you choose voip4.freephoneline.ca, you will also need to uncheck the default box for ProxyServerPort and enter 6060.

d. For RegistrationPeriod use 3600

e. RegistrationRetryInterval is 120

f. Uncheck the box under the Value column to disable X_Use302ToCallForward

Freephoneline requires calls to be bridged if you want your ATA to forward calls.
If X_Use302ToCallForward is enabled, calls that are forwarded by the ATA (as opposed to using Freephoneline's Follow Me feature) will be dropped to voicemail.



g. Scroll down a little. Uncheck default box for X_UsePublicAddressInVia

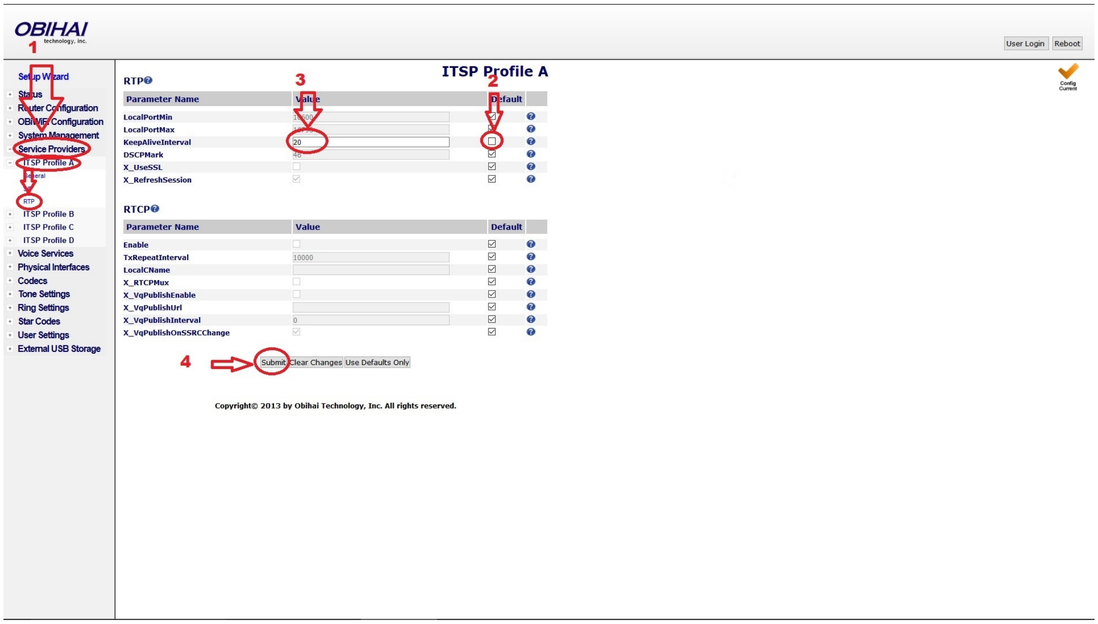
h. Check or enable X_UsePublicAddressinVia (check the box under the Value column)

This sends your public IP address (as determined by your Obihai ATA) in the VIA header that's sent to Freephoneline's server. This helps to ensure data is sent back to your public IP address as opposed to your LAN IP address (192.100.1.x, for example). If Freephoneline were to send data to 192.100.1.x, it would never reach you. It needs to be sent to your WAN or public IP address first before your router can send or route the data to the local IP address of your Obihai ATA.

This setting should be enabled to help ensure one-way audio issues are not caused by Freephoneline's switches in the manner they are configured at the time of this writing.

i. Click the "Submit" button, but do NOT reboot your ATA.

8. Set KeepAliveInterval for RTP to 20



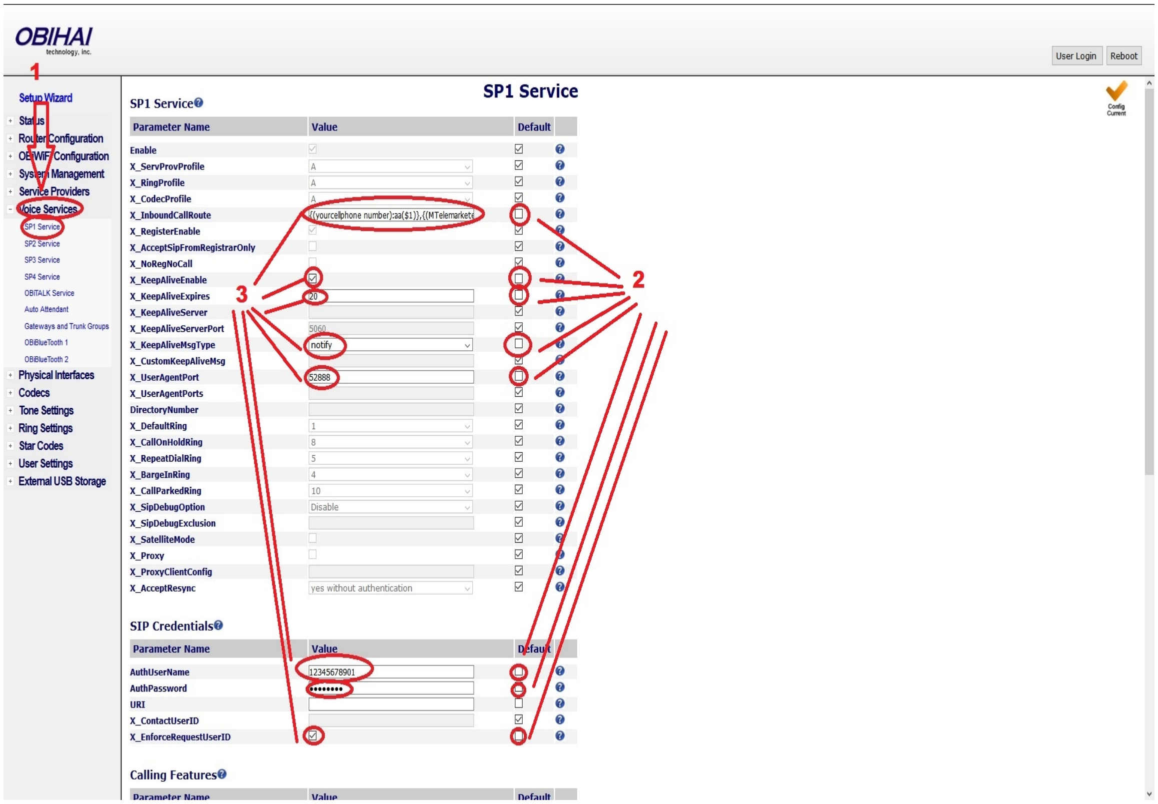
a. Navigate to Service Providers–>ITSP Profile A–>RTP

b. Uncheck default box for KeepAliveInterval

c. Set KeepAliveInterval to 20

d. Click the “Submit” button. Do NOT reboot your ATA

9. Enable and configure your SP1 Service



A. Navigate to Voice Services–>SP1 Service

B. X_InboundcallRoute

If you want to make changes to it, you will need to uncheck the default box for X_Inboundcallroute.

Entries in X_InboundcallRoute are processed from left to right.

By default with OBi200 ATAs, this value is ph
By default with OBi202 ATAs, this value is ph, ph2
The Obitalk.com portal shows {ph1,ph2} for OBi202 in some cases.

ph stands for phone port.
ph2 stands for phone port 2.
ph1 is phone port 1.

That is, all inbound calls to your Freephoneline phone number will be sent to your phone ports, and your attached phones, will, of course, ring. **That's fine. If that's all you want, then leave X_InboundcallRoute set to default, and move onto step 9C on page 25.**

i. But what if you wanted to use your cellphone to make free calls using Freephoneline or any other service that is configured on your ATA using regular cellular airtime (and not cellular data) by simply dialing your Freephoneline phone number?

We could send your incoming cellphone call to Obihai's Auto Attendant.

So now your X_InboundcallRoute would look like this:
{(your cellphone number):aa($1)},{ph}for OBi200s

or

{(your cellphone number):aa($1)},{ph,ph2}for OBi202s

aa stands for Auto Attendant.

If you hangup before auto attendant (aa) answers, then it will call you back (represented by $1).

When you call your Freephoneline phone number with your cellphone, your call will be sent to Obihai's Auto Attendant. And then all other incoming calls will be sent to the phone port(s).

If you call on your cell to your Freephoneline phone number, the Obihai Auto Attendant (AA) will ask you if you want to continue the call by pressing 1 (which will then ring the phones attached to your ATA), or press 2 to place a new call, or press 3 to have your ATA call you back (great if you have free incoming minutes).

You need to change "your cellphone number" to the actual cell phone number that appears in Freephoneline's call logs when you make calls with your cell phone. Ex. 4169999999. Basically, you want to pick a phone number (doesn't have to be your cell phone number) that you want to use to call in to access the services on your ATA. If you want, you could allow your relatives to access your services as well.

Ex. {(your cellphone number|your mom's phone number):aa($1)}

Also, refer to page 44 in this guide for a related question and answer.

ii. What about known Telemarketers? Do you still want your phone to ring if they call? I don't.

You will need to create a user defined digitmap and stick all of the phone numbers you want to block in it. Refer to the section at end of this guide on Telemarketers (step 13).

So then your X_Inboundcallroute might look like {(yourcellphone number):aa($1)},{(MTelemarketers):},{ph,ph2} for an OBi202

or

{(yourcellphone number):aa($1)},{(MTelemarketers):},{ph} for an OBi200

M stands for Digitmap.

If you don't want to use Auto Attendant, but do want to block telemarketers, just remove {(yourcellphone number):aa($1)} from your X_Inboundcallroute.

iii. What if I want to use Nomorobo to block telemarketers?

With an Obihai ATA and a toll free number from access number from Nomorobo, you can use Nomorobo with any service provider. To learn more about Nomorobo, visit www.nomorobo.com.

Keep in mind, if you have Follow Me enabled in your Freephoneline web portal, that in order for Nomorobo to work properly, your Nomorobo access number has to ring simultaneously (or first, if you're using sequential forwarding). Freephoneline doesn't allow toll free phone numbers to be used with Follow Me at the time of this writing. If you're having problems getting Nomorobo working with Follow Me enabled, disable Follow Me after logging into your Freephoneline web portal account: https://www.freephoneline.ca/followMeSettings.

a. Sign up at https://www.nomorobo.com.

b. Select "Landline/VoIP" for your phone type.

c. Select Vonage as your carrier (**or choose one that gives you a toll free Nomorobo phone number**). The bolded part is the important step.
I don't know if choosing other carriers will provide you with a toll free Nomorobo phone number, but maybe some will.

d. For phone number, enter your VoIP (Freephoneline) phone number that you want to protect with Nomorobo.

e. Click next (and keep the webpage open).

f. Navigate to Voice Services-->SP1 Service-->X_InboundCallRoute

Yours might look like, for OBi200 users, this:
{(yourcellphonenumber:aa($1)},{(MTelemarketers):},**{ph,sp1(1866732xxxx@tollfree.alcazarnetworks.com;ui=$1)}**

You just need to add the bolded part for Nomorobo to work.

For OBi202s, you would use
{ph,ph2,sp1(1866xxxxxxx@tollfree.alcazarnetworks.com;ui=$1)}

What happens with an incoming call is that both your phone port(s) and Nomorobo's service phone number ring simultaneously. ui=$1 makes the original caller ID information pass onto Nomorobo.

1866xxxxxxx represents your Nomorobo access phone number. You need to replace the xxxx with the actual digits for your Nomorobo access phone number.

You can also use switch.starcompartners.com instead of tollfree.alcazarnetworks.com. I'm not sure, out of all the free SIP tollfree termination services available that don't require registration, which is the most reliable.

g. We will come back to finishing up Nomorobo configuration at the appropriate time in this guide towards the end of step 13.

9 C. Refer back to the picture under step 9. Uncheck default boxes for X_KeepAliveEnable, X_KeepAliveExpires, X_KeepAliveMsgType, X_UserAgentPort, AuthUserName, AuthPassword, and X_EnforceRequestUserID.

D. Check or enable X_KeepAliveEnable.

E. Change X_KeepAliveExpires to 20.

F. Select “notify” from the dropdown box for X_KeepAliveMsgType.

G. X_UserAgentPort should be high random port number between 30000 and 65535. Just pick a port number in that range, and enter it.

By using a high random port you help to thwart SIP scanners and may also circumvent a faulty SIP ALG feature in your router.

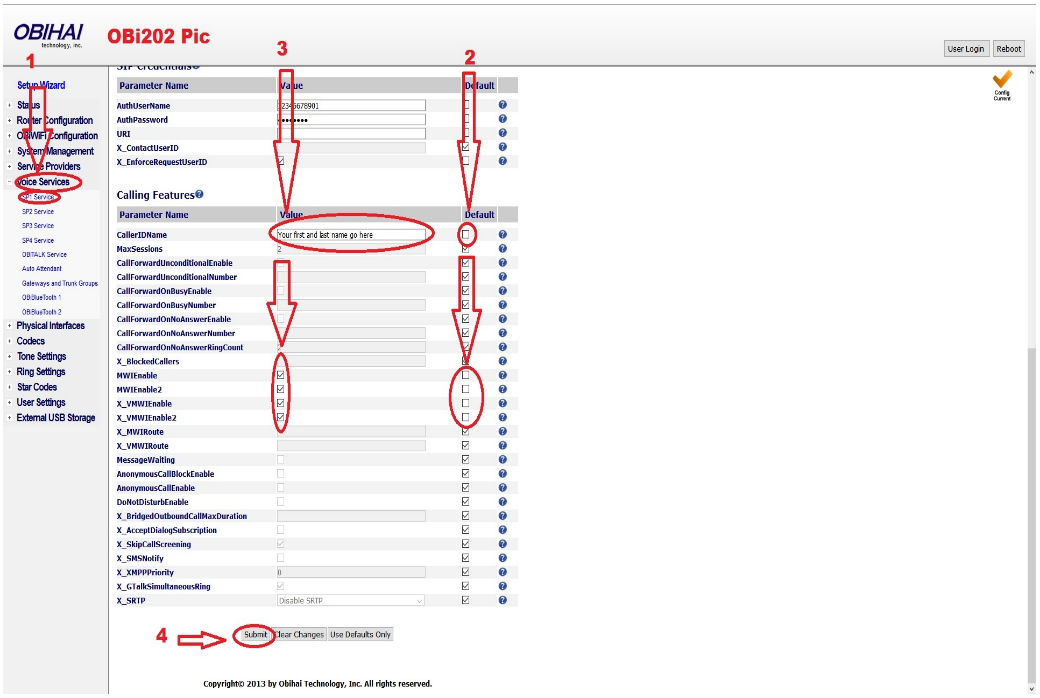
H. Under SIP Credentials, your AuthUserName is your Freephoneline SIP Username, which is found after logging into your account at https://www.freephoneline.ca/showSipSettings. Enter your SIP Username.

I. Your AuthPassword is your Freephoneline SIP Password, which is found found after logging into your account at https://www.freephoneline.ca/showSipSettings. Enter your SIP Password.

J. Check or enable X_EnforceRequestUserID to accept SIP invite requests only if the userids in those requests match AuthUserName or X_ContactUserID (found under Voice Services > SP(service you're using) Service-->SIP Credentials). This helps to thwart SIP scanners or crackers trying to access your devices and services.

SIP scanners are programs written by crackers (script kiddies). They look for ways to break into your home network by scanning for open ports. Typically, they'll scan for 5060, 5061 and a few others (some scan for a lot more than that). If a port is open, they can access your ATA (and, potentially, other devices on your LAN). Your phones will ring with caller ids appearing as 1001, 999, etc. These crackers will try to make free calls using your services. That's one reason why port forwarding is bad (port forwarding opens ports; it's a security issue).

Normally, to also help stop SIP Scanners I recommend enabling X_AcceptSipFromRegistrarOnly, but unless your ATA is registered with voip.freephoneline.ca, Fongo Mobile calls to your Freephoneline phone number will be dropped straight to voicemail. Fongo Mobile calls to Freephoneline phone numbers are SIP URI calls.



K. Scroll down, and under Calling Features, uncheck the default boxes for CallerIDName, MWIEnable, and X_VMWIEnable. For OBi202 users also uncheck the default boxes for MWIEnable2 and X_VMWIEnable2.

L. Scroll down to Calling Features–>CallerIDName

Enter your First name and last name. Your outbound Caller ID Name will show up on other people's caller ID displays when you call them.

M. Check the boxes under the Value column or enable MWIEnable and X_VMWIEnable. For OBi202 users, also enable MWIEnable2 and X_VMWIEnable2, if you want to receive stutter tones and flashing lights on phone port 2 when voicemail is left.

MWIenable stands for message waiting indicator (for voicemail). This creates the stutter tone. X_VMWIEnable stands for visual message waiting indicator (for voicemail). This creates the flashing light. Without these settings enabled, you won't know voicemail is waiting for you.

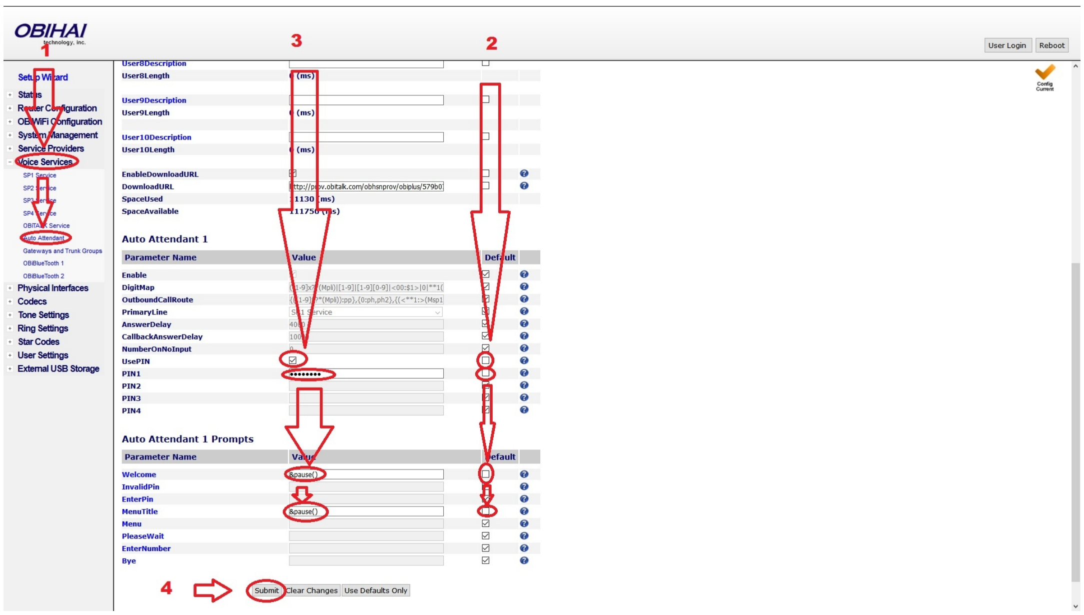
Please note that with Freephoneline, voicemail notification will not trigger until 10 minutes after a voicemail is left, and voicemail notification will not disappear until 10 minutes after your voicemail is deleted.

N. Click the "Submit" button, but do NOT reboot the ATA.

10. Create a PIN for the Auto Attendant, and shorten the voice prompts.

Refer to step 9B i on page 22.
You will need {(your cellphone number):aa($1)} in your X_Inboundcallroute.
Also, refer to page 44 in this guide for a related question and answer.

If you never plan on using the Auto Attendant, skip this section and move onto step 11 on page 28.



a. Navigate to Voice Services–>Auto Attendant–>Auto Attendant 1

b. Uncheck the default boxes for UsePIN and PIN1. Under Auto Attendant 1 Prompts, also uncheck the default boxes for Welcome and MenuTitle.

c. Check or enable UsePIN under the Value column.

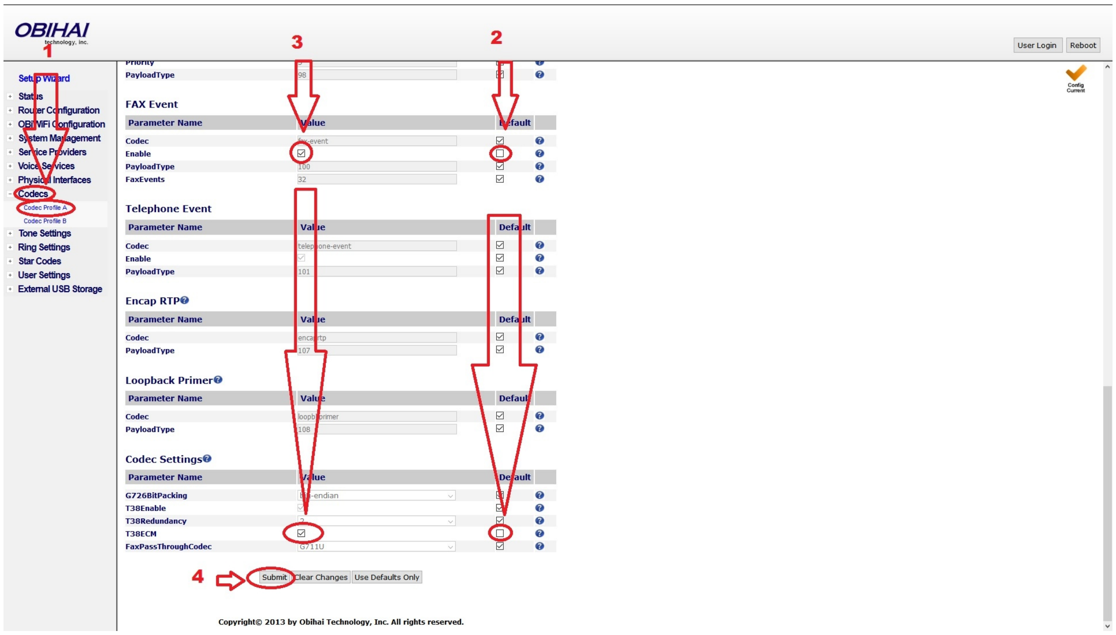
d. For PIN1, set a PIN or numerical password you want to use for when you dial into your ATA using your cellphone to access the Auto Attendant. Refer to step 9B i.

You can also create up to four different PINs. You can give one to you aunt; another to your mom, etc. The idea is you don’t want anyone calling in from a phone number and accessing your services without knowing the password.

e. To shorten the AutoAttendant 1 voice prompts, enter &pause() for both Welcome and Menutitle under the value column.

f. Click the “Submit” button, but do NOT reboot the ATA.

11. Enable T.38 fax protocol for use with Freephoneline



a. Navigate to Codecs–>Codec Profile A

b. Uncheck the default boxes for Enable (Fax Event) and T38ECM

c. Check or enable the boxes under the Value column for Enable (Fax Event) and TC8ECM.

d. Click the submit button, but do NOT reboot your ATA.

In your call status page (Under Status–>Call Status), during T.38 protocol fax transmission, you'll see the following: "Audio Codec = tx=; rx=G711U" (which doesn't specifically state T.38, but this is the only indication OBi20x seems to provide).

If you're not using T.38, you'll see "Audio Codec = tx=G711U; rx=G711U" instead, which just means you're using the G.711u codec only and not T.38 fax protocol.

If you’re having problems faxing,

i) I would increase volume slightly:
Navigate to Physical Interfaces-->Phone Port-->

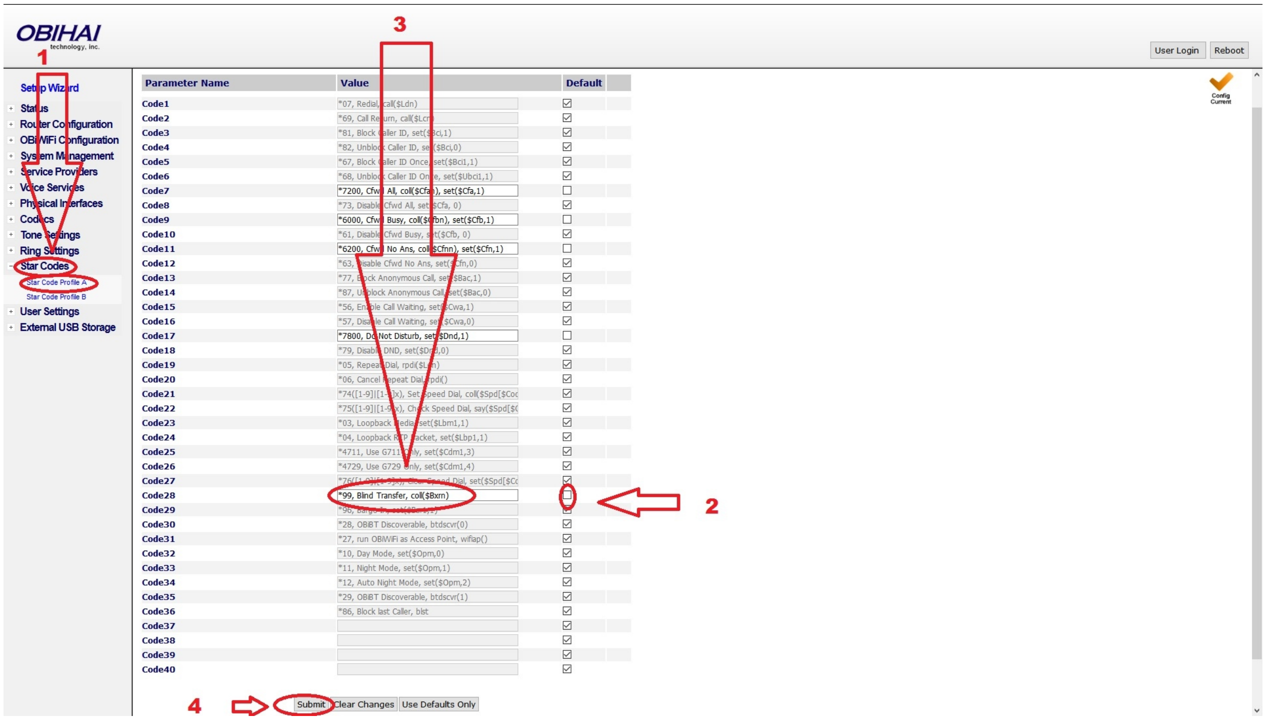
a) Change ChannelTxGain to -1
b) Change ChannelRxGain to 0

(submit/save/reboot)

ii) On your fax machine, lower baud rate to 9600 bps (I'm able to fax at faster rates than 9600, but if you can't without outgoing faxes failing, lower your baud rate to 9600)

iii) On your fax machine, turn off or disable ECM (both TX and RX)
http://www.voipmechanic.com/voip-fax-settings.htm

12. Change Blind Transfer from *98 to *99 to allow *98 to be used to check voicemail.



(You may notice that I've changed the defaults for Do Not Disturb and Call forwarding. That's because I don't want to accidentally dial *78, for example, and suddenly have all incoming calls dropping straight to Freephoneline's voicemail system because I enabled Do Not Disturb by accident. That's not necessarily something you need to do, but you can if you wish.)

a. Navigate to Star Codes–>Star Code Profile A

b. Uncheck the default box for Code 28.

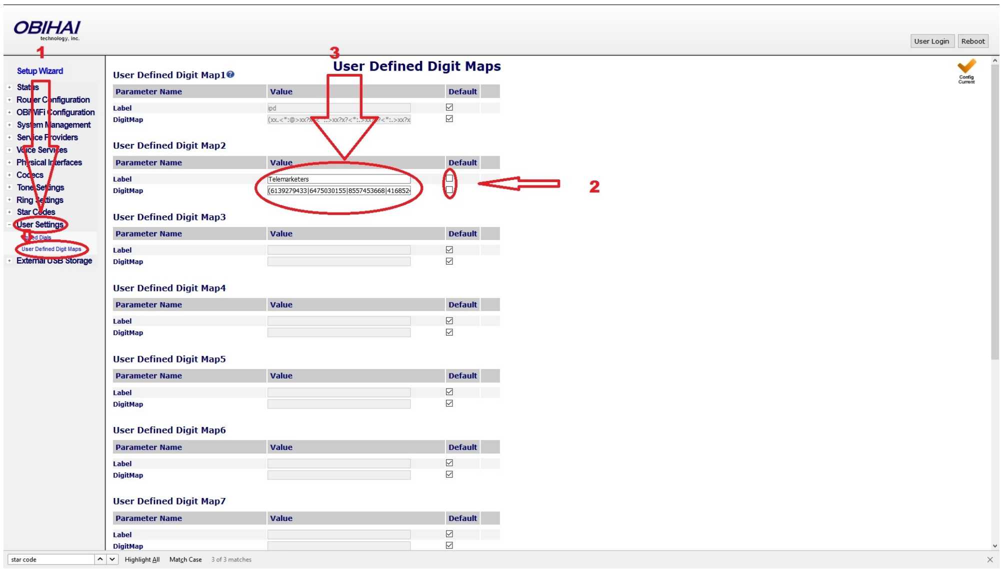
c. Change *98 to *99 in box under the Value column for Code 28.

Blind Transfer is neat by the way: http://www.obitalk.com/forum/index.php?topic=3039.0. Obviously, if you change Blind Transfer to *99, you need to use *99 for Blind transfer.

d. Click the "Submit" button, but do NOT reboot your ATA.

13. Create a list of phone numbers to block (ex. Telemarketers)

For the section, refer to step 9B ii. You'll need {(Mtelemarketers):}in your X_Inboundcallroute. If you're not interested in blocking callers you can skip to step 13 c on page 32.



With Freephoneline, Follow Me in your Freephoneline web portal must be disabled (unless you route MTelemarketers somewhere where the call is picked up immediately) for call blocking via your Obihai ATA to work. Login at https://www.freephoneline.ca/followMeSettings, and check your Follow Me settings.

To learn more about MTelemarketers (above) and blocking Telemarketers, visit http://www.toao.net/503-blocking-telemarketers-with-an-obi-ata. Good guide. Note that user defined digitmaps are limited to 511 characters.

If you have an OBi200 or OBi202, you can also navigate to Voice Services-->SP (service you're using)-->Calling Features-->X_BlockedCallers
You can enter 10 phone numbers, separated by commas, that you want to block per SP.

a. Navigate to User Settings-->User Defined Digit Maps

b. Uncheck the default boxes for Label and Digitmap for User Defined Digit Map 2

i. For the Label, enter Telemarketers.

ii. For the DigitMap, enter phone numbers you want to block.
For example, (6139279433|6475030155|8557453668)

Some people ask about blocking anonymous or unknown calls. (?|un@@.|Un@@.|anon@.|Anon@.)

? = no Caller ID
@ = any single alphanumeric (number or letter) except #
@. = zero or more occurrences of any length alphanumeric (number or letter) sequence except #
@@. = any single alphanumeric (number or letter) except # followed by zero or more occurrences of any length alphanumeric sequence except #

un@@. will catch unknown or anything starting with un followed by at least one more character (except #)
anon@. will catch anonymous or anything starting with anon (except #)

I do not generally recommend blocking anonymous calls since doctors and hospitals can call from them.

Note that you must enter phone numbers as they appear in your VoIP service's call log. For Freephoneline users, login at https://www.freephoneline.ca/callLogs.

Note that this method for Freephoneline drops all Telemarketer calls to Freephoneline's voicemail (FPL basically wants all incoming calls picked up no matter what because FPL makes money off of incoming termination fees to its network), but at least your phones won't ring.

Because of not wanting these telemarketer calls to drop to FPL's voicemail, boon1 came up with a cool idea for sending these calls to the auto attendant:
http://forums.redflagdeals.com/freephoneline-ca-free-local-soft-phone-line-lifetime-voip-821229/265/#p21660123.

However, for me, that's a bit of a problem because people in my household use the Auto Attendant to dial into and receive calls back from (and I don't want them to hear voice prompts that are intended for telemarketers).

Alternatively, if you have another ITSP, configured on SP2 for example, you could use{(MTelemarketers):sp2(phonenumbertosendtelemarketers)}in FPL's X_InboundCallRoute in place of {(MTelemarketers):} to send those telemarketing calls to another phone number.

If Freephoneline (FPL) is SP1, you can also use {(Mtelemarketers):sp1(phonenumbertosendtelemarketers)}in your X_Inboundcallroute to send the call to another phone number.

Or (for SIP calls) you can use
{(MTelemarketers):sp1(sipnumber@sipdomain.com)}

It doesn't really matter which option you choose. But if you don't want telemarketing calls to drop straight to FPL's voicemail, it is possible with an Obihai ATA, to route these calls elsewhere. Maybe you want to send them to Lenny: http://toao.net/595-lenny (keep in mind that sending telemarketers to Lenny will let telemarketers know your phone number is active).

iii. You can also create whitelist digitmaps.

Then you might want to have the following X_Inboundcallroute:

{(MWhitelist):ph}

All phone numbers calling that are listed in your Whitelist digitmap will ring your phone port. If you want to blacklist everyone that's not in your Whitelist, in your X_Inboundcallroute, you would remove the ph or {ph,ph2} that appears at the very end. You would just want {(MWhitelist):ph}.

Here's another example of how X_Inboundcallroute works:
{(MWhiteList):ph},{(MTelemarketers):},{ph}

So from left to right, if a whitelisted number calls you, it rings your phone port. If the number is not in your whitelist, the number is checked against your Telemarketer/blacklist digitmap, and if there's a match your phone doesn't ring (the call gets dropped to FPL's voicemail). If the number, by this point, still doesn't match your Telemarketer digitmap, your phone will ring. **Logically, that situation doesn't make sense because there's no point in creating a whitelist if your phone is going to ring anyway when anyone who is not in your blacklist calls. My point in showing this example is that if you're going to create a Whitelist, you need to remove ph at the very end of your X_Inboundcallroute.**

The following might be a better solution for Telemarketers for FPL users.

Here are the steps I took:

- Went to www.tropo.com
- Created a free developer account
- Verified account and logged in
- Found an audio file that plays SIT tones followed by a "We're sorry, you have reached a number that has been disconnected..."
- Clicked on "My Files" in Tropo and stuck the file in the www folder
- Selected "My Apps" and clicked "create application"
- Entered scammers for Basic information (you can put whatever you want here)
- Clicked on "new script"

Then Entered the following:

```
<?php
say("http://hosting.tropo.com/mytropoaccount#/www/disconnectedmessageaudiofilethatIadded.mp3");
say("http://hosting.tropo.com/mytropoaccount#/www/disconnectedmessageaudiofilethatIadded.mp3");
?>
```

- Saved the script as scammers.php (just has to end with .php)

- Clicked "create app"

- Scrolled down and picked a free Tropo phone number for Canada

- Stuck {(MTelemarketers):sp1(mytropophone#)} in X_InboundCallRoute for FPL in my OBi (where SP1 = FPL), but it doesn't matter what SP you use, as long as you call your Tropo phone number for free using it.

- Saved settings/ rebooted the ATA.

13 c. Scroll down, and click the "Submit" button. This time you can reboot your ATA by clicking the "reboot" button on the upper right-hand corner. Wait for about a minute before trying to access your ATA again.

13 d. **Nomorobo**

If you're not interested in using Nomorobo, skip to step 14 on page 34. Step 13 d (this step) is a continuation from step 9 iii g.

i. Test your Nomorobo number. Go back to the other web browser tab that has Nomorobo website loaded. Select "I'm ready. Call me now." Or choose the test call option.

ii. When your phone rings, answer it, to confirm that Nomorobo works.

That's it for Nomorobo. If you don't receive the Nomorobo verification call, you may not have to worry.

For example, I've received the following question from another Freephoneline user before:

"I tried following your guide to get Nomorobo working with Freephoneline, but for some reason when I dial number I got from website it rings once and then I get a busy signal (okay, that's what should happen). But when I click on `I'm ready. Call me now', my home phone never rings.

I've tried alcazar... Have Follow me - Disabled on FPL

Here is my Inbound route for an OBi202:

{(MTelemarketers):},{ph,ph2,sp1(1866xxxxxxx@tollfree.alcazarnetworks.com;ui=$1)}

Any ideas what could be my issue?"

A few thoughts occur to me.

A) Possibly Nomorobo is blocking the outbound call due your phone number not being owned by Vonage (or the provider you selected to obtain the toll free Nomorobo access number). If you don't see the test call in https://www.freephoneline.ca/doGetCallLogs, that might be one explanation. However, that's not a perfectly reliable method for determining what's happening because incoming phone numbers only appear in FPL's call log if the incoming call has been answered by you or FPL's answering machine.

B) If the test call to your FPL number is a SIP URI call from Nomoboro, the test call will never reach you because FPL blocks incoming SIP URI calls unless they're from Fongo Mobile.

C) Possibly FPL doesn't allow incoming Nomorobo verification calls anymore.

D) This is the most important point: it doesn't seem to really matter if you can't receive the Nomorobo verification call. If you don't verify the test call, Nomorobo still picks up the call with "please try your call again" when a telemarketer calls. If the test call is verified, the telemarketer hears this: https://soundcloud.com/nomorobo/nomorobo-robocaller-captcha. I might actually prefer if the test call isn't verified.

Regardless, I suspect if you are seeing "busy" in your Obihai ATA's call history, then your Nomorobo toll free number is working as intended.

Here's a quick way to test:

a) In your X_InboundCallRoute for Freephoneline, use
{ph,ph2,sp1(1866xxxxxxx@tollfree.alcazarnetworks.com;ui=3194327596)}

For OBi200 users, use{ph,sp1(1866xxxxxxx@tollfree.alcazarnetworks.com;ui=3194327596)}

1866xxxxxxx is your toll free Nomorobo access number. The xxxxxxx needs to be replaced.

b) Using your cellphone, dial your Freephoneline phone number.

The call should be answered within 3 seconds of ringing by Nomorobo ("please try your call again" message is played or something similar), and then call is disconnected by Nomorobo.

c) now, in your X_InboundCallRoute for Freephoneline,
try {ph,ph2,sp1(1866xxxxxxx@tollfree.alcazarnetworks.com;ui=yourcellnumber)}

For OBi200 users, try{ph,sp1(1866xxxxxxx@tollfree.alcazarnetworks.com;ui=yourcellnumber)}

1866xxxxxxx is your toll free Nomorobo access number. The xxxxxxx needs to be replaced. Similarly, youcellnumber needs to be replaced with your actual cell phone number.

d) Call your FPL number using your cell phone (or by using another provider other than FPL).

You should see the following in your ATA's call history (found by logging into ATA with a web browser and navigating to Status–>Call History):
Call Failed (486 Busy Here; SP1(1866xxxxxxx@tollfree.alcazarnetworks.com;ui=yourcellnumber)

That's what's supposed to happen. If there's a busy response from Nomorobo, then the call will continue to ring your phone ports (beyond 3 seconds).

If Nomorobo picks up the call, then your phone ports won't ring.

3194327596 is a known marketer.
http://www.obitalk.com/forum/index.php?topic=8802.msg58226#msg58226
(thanks to azrobert)
There's a related thread here: https://www.obitalk.com/forum/index.php?topic=10368.20.

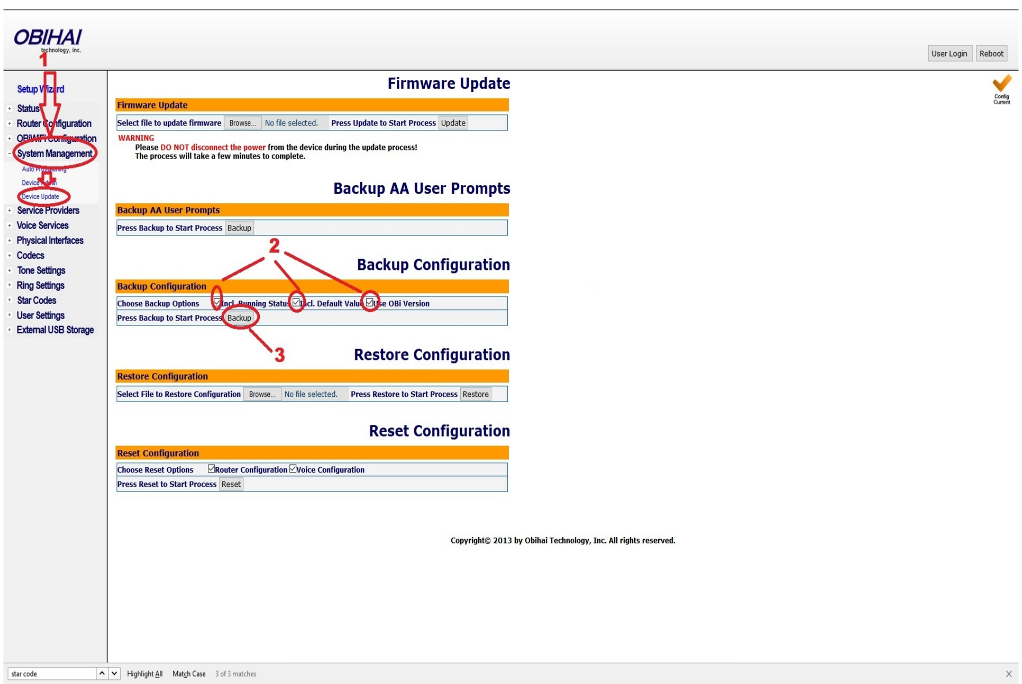
Should you have further questions concerning Nomorobo, you may want to try asking there. You do not, at this time of this writing, need to setup a dummy SIP trunk as described in the first page of that thread.

What happens with Nomorobo is that when the incoming caller ID isn't considered a telemarketer, Nomorobo responds with a busy signal, and your phone ports will still ring. If Nomorobo determines the caller is a telemarketer or robodialer, Nomorobo answers the call, and your phone ports stop ringing:
https://nomorobo.zendesk.com/hc/en-us/articles/205761825-What-Happens-When-Nomorobo-Blocks-a-Call-

Nomorobo may take a ring or two before answering the call (your phone will likely ring once per telemarketing call).

Keep in mind that, like Lenny, if Nomorobo answers the call, telemarketers will know your phone number is active and possibly add your phone number to other calling lists. So, in my opinion, using Tropo with SIT tones in combination with Nomorobo is the best solution. When Nomorobo answers the call, make note of the incoming phone number and add it to your MTelemarkers DigitMap.

14. Save your settings



a. Navigate to System Management–>Device Update

b. Under Backup Configuration, check the boxes for Incl. Running Status, Incl. Default Value, and Use Obi Version.

c. Click the “Backup” button, and save your file.

Note that passwords are not backed up. So if you have to restore your configuration, you will need to enter your AuthPassword again.

* If you want to perform a factory reset at any point, you can also do so here by clicking the reset button under “Reset Configuration.”

* If you wish to restore your backup file, you can by clicking the “browse” button under “Restore Configuration”, selecting your backup file, and then clicking the “Restore” button.

That’s all, folks!

But the following pages contain more tips and tricks you may find useful.

How Do I Check Voicemail Using Freephoneline?

1) If you followed this guide you can pick up a phone attached to your ATA and dial *98.

2) You can call your Freephoneline phone number from your Obihai ATA using Freephoneline.

3) Dial a Freephoneline voicemail remote access phone number (useful from your cellphone) followed by your FPL account phone number (starting with 1) + #, followed by your voicemail password + #: http://www.freephoneline.ca/vmAccessNumbers

4) You can also access voicemail by logging in at https://www.freephoneline.ca/mailbox. However, at the time of this writing, deleting voicemail from Freephoneline's portal triggers a bug, whereby your voicemail notification or message waiting indicator will never go away unless you send in a ticket request to Freephoneline and ask for your account to be fixed.

Note that with FPL, it takes 10 minutes after a message is left before receiving a message waiting notification on your phone. Similarly, it takes 10 minutes after voicemails are deleted before the Messsage Waiting indicator disappears.

Did you know you can also record an active call?

1. Dial ***1

2. Enter that IP address into a web browser

3. Navigate to Status-->Call status

4. A call needs to be in progress in order to record. Click "call status" when a call is in progress.

5. Click the record button.

6. A window will eventually popup. Click save.

7. An .au audio file will start downloading onto your computer.

Did you know that you can host a large conference calling session using your Freephoneline phone number?

Visit https://www.freephoneline.ca/features/conference-calling/

Only on the OBi202: Press # for Phone Port Collaboration

(this information is from an Obihai Advertisement)

Did you know that you can have a mini phone system with the OBi202? While the Phone Port 1 and Phone Port 2 can function independently so you and another person can be on two different calls at the same time, the two phone ports to work together. …Just Press #

- Call the Other Phone – You can press # to call from one phone to the other phone.

- Call Transfer – While on a call, press the hook or Flash button and then press # to ring the other phone. All three of you can talk together or just hang-up to transfer the call to the other phone.

- Join-in on the Other Phone's Call – If the phone on phone port 1 is on a call, from the phone on phone port 2 press # to join-in on the call.

- Incoming Call Pick-Up – If the phone connected to phone port 1 is ringing, pick-up the phone connected to phone port 2 and press # , then say "Hello?"

<u>Are you getting one way audio issues with an OBi200/202 and Freephoneline? Are incoming calls not ringing? Can you not hear one side of the conversation (you can hear the caller, but the caller can't hear you or vice versa)?</u>

<u>Simple things to check</u>

a) "If you're getting an `invalid account' error messages, or if people trying to call you are hearing "invalid account" or a busy signal, please log in to your account online at https://www.freephoneline.ca/followMeSettings, and reset your Follow Me settings (or disable it). Please ensure your temporary FPL number is not listed as one of the Follow Me numbers."

b) If you have calls going straight to voicemail, login at https://www.freephoneline.ca/voicemailSettings, and ensure "Rings before voicemail" is greater than 1.

c) Also, check in your ATA to ensure you don't have "Do Not Disturb" enabled. This is found after logging into your ATA or at Obitalk.com under Voice Services-->SP(FPL) Service-->Calling Features-->DoNotDisturbEnable. Ensure there is no checkmark under "Value".
Navigate to Voice Services-->SP (FPL) Service-->Calling Features
a) Ensure DoNotDisturbEnable is unchecked
b) Ensure CallForwardUnconditionalEnable is unchecked
c) Ensure CallForwardOnBusyEnable is unchecked
d) Ensure CallForwardOnNoAnswerEnable is uncheked
e) Ensure AnonymousCallBlockEnable is unchecked

<u>Follow These Steps:</u>

1) Make sure whatever modem/router combo your ISP gave you is in bridge mode <u>if you are using your own router.</u> Call/contact your ISP if you have to. For Bell Hubs, visit http://forums.redflagdeals.com/please-sticky-how-bypass-bell-hub-use-your-own-router-1993629/

<u>2) Refer to Page 4 of the Preamble under Hardware Related Issues and also page 3 for how to disable SIP ALG in your router.</u>

3) Disable SIP ALG in whatever router you're using (see step 5 on page 3).

4) In your Obihai ATA or at Obitalk.com, Navigate to Voice Services-->SP(FPL) Service-->X_UserAgentPort

X_UserAgentPort should be a random port number between 30000 and 65535. Just pick a port number in that range.

By using a high random port you help to thwart SIP scanners and may also circumvent a faulty SIP ALG feature in your router.

5) Navigate to Service Providers-->ITSP Profile (FPL)-->SIP

i) ensure X_DiscoverPublicAddress is enabled (it is by default)

ii) enable X_UsePublicAddressInVia (it's not by default)
You will need to uncheck default, device default, and Obitalk settings boxes. Then check the box to enable the feature.

Retest
When I say Retest, retest always includes the following: A. Turn off both router and ATA. B. Turn on router. Wait for router to be fully up and transmitting data. C. Turn on ATA.

Then retest by calling your FPL phone number. If the problem is solved, don't continue.

6. If that doesn't work, you can also try enabling X_DetectALG (Navigate to Service Providers-->ITSP Profile (FPL)-->SIP)

(submit/save/reboot ATA)

7. Retest
When I say Retest, retest always includes the following: A. Turn off both router and ATA. B. Turn on router. Wait for router to be fully up and transmitting data. C. Turn on ATA.

Then retest by calling your FPL phone number. If the problem is solved, don't continue.

8. If that still doesn't work, disable X_DetectALG. And submit/save/reboot ATA.

9. Try voip4.freephoneline.ca:6060

Refer to the underlined notes in section 7c on page 17 of the setup guide. That is, try voip4.freephoneline.ca for the ProxyServer and 6060 for the ProxyServerPort

Then retest by calling your FPL phone number. If the problem is solved, don't continue.

voip4.freephoneline.ca:6060 is a SIP server whose purpose is to help those with SIP ALG issues (can't disable it in the user's router or gateway, for example).

10. Try this at your own risk: use voip3.freephoneline.ca for ProxyServer and 5060 for the ProxyServerPort
Refer to step 7c on page 17 of the setup guide.

voip3.freephoneline.ca is intended for testing purposes only--or for those who receive explicit permission to use it. Using it for an extended period may get your account banned. However, if using voip3.freephoneline.ca does work, you should open up a ticket with support and let them know that you can't get two-way audio any other way: https://support.fongo.com/anonymous_requests/new. For the issue type, select VoIP Unlock Key–>My Account Inquiry. Ask for a "forced account registration."

If no one responds to your support ticket, provide the ticket number in a private message to Fongo Support: http://forum.fongo.com/ucp.php?i=pm&mode=compose&u=7852

FPL configures its SIP servers differently than many other VoIP providers.
voip3.freephoneline.ca conforms more to the norm. But using it without permission can get your account banned. If you'd like to avoid getting your account banned, then just skip to step #12.

11. Retest. When I say Retest, retest always includes the following: A. Turn off both router and ATA. B. Turn on router. Wait for router to be fully up and transmitting data. C. Turn on ATA.

Then retest by calling your FPL phone number.

12. If none of that helps, then, unfortunately, you're pretty much stuck with port forwarding your RTP (UDP) port range 16660-16798 from your router to your ATA. For reference, that range can be found under ITSP Profile (FPL)-->RTP. Then look at LocalPortMin and LocalPortMax. RTP packets need to reach your ATA in order for you get incoming audio. Quite often, when the one way audio issue occurs, this is the problem. RTP packets are not reaching your ATA. **Ideally, one should not have to port forward in order to achieve proper two-way audio, since port forwarding does create security issues. Port forwarding should only be done when everything else fails.**

Refer to the port forwarding section of your router manual to learn how to port forward to your ATA. If a router was given to you by your ISP, call your ISP.

13. Retest. When I say Retest, retest always includes the following: A. Turn off both router and ATA. B. Turn on router. Wait for router to be fully up and transmitting data. C. Turn on ATA.

Then retest by calling your FPL phone number.

14. This really shouldn't be necessary, but it's possible you may need to also port forward (from your router to your ATA) the UDP port you randomly selected in step #4. Retest afterwards.

15. Thanks to Mango, many of us now understand that in order for ATAs to remain registered and working properly with a VoIP SIP provider like Freephoneline, in particular after power failures, the following conditions must be met:

UDP Unreplied Timeout (in your router) < NAT Keep-alive Interval (in your ATA; for Obihai ATAs this is X_KeepAliveExpires) < UDP Assured Timeout (in your router) < SIP Registration Failure Retry Wait Time (or RegisterRetryInterval in Obihai ATAs)

"<" means less than.

A problem can occur when the Keep-Alive interval is greater than UDP Assured Timeout (often 180 by default in consumer routers): the NAT hole will close due to the ATA not communicating frequently enough with the SIP server. In turn, incoming calls may, intermittently, not reach the ATA. Again, X_Keepalives expires is supposed to be 20 with FPL.

Getting access to both UDP Unreplied Timeout and UDP Assured Timeout settings in consumer routers may be difficult, if not impossible. Asuswrt-Merlin, third party firmware for Asus routers, does offer easy access to these two settings, which are found under General–>Tools-->Other settings. In part, for this reason, I tend to use Asus routers that work with Asuswrt-Merlin. However, my understanding is that third party Tomato firmware has these two settings as well. So if your router supports Tomato firmware, that may be another option.

The keep alive interval for FPL is 20. The SIP Registration Failure Retry Wait Time is 120. I use 10 for UDP Unreplied Timeout and 117 for UDP Assured Timeout.

16. If all else fails, open a support ticket at https://support.fongo.com/anonymous_requests/new, and try posting at http://forum.fongo.com/viewtopic.php?f=8&t=15120&start=300. For the issue type, select VoIP Unlock Key–>My Account Inquiry. Ask for a "forced account registration." If no responds to your support ticket, provide the ticket number in a private message to Fongo Support after registering and logging into the forums: http://forum.fongo.com/ucp.php?i=pm&mode=compose&u=7852.

I'm getting choppy audio, what should I do?

You're experiencing jitter.

Generally speaking it's best to have a decent router for VoIP with strong QoS features. Stick your ISP's modem in bridge mode, use your own router, and properly enable QoS for your ATA. Refer to your router's manual or contact your ISP if your router was provided by your ISP.

I'm not a big fan of this site, but for a general QoS description, visit http://www.voipmechanic.com/qos-for-voip.htm (avoid anything it says about G729 codec).

When you test below, pick the location that is closest to your VoIP service provider's server location.

1) The typical reaction would be to try enabling QoS properly in your router for your ATA. Refer to your router's manual.

2) Another possibility is you're dealing with congestion during prime time (8p.m. to 11 p.m., especially on Sundays). That's an ISP issue (possibly oversold its service in your area). With Rogers or a cable ISP, you could very well be dealing with local node congestion.

Try running http://vac.visualware.com/ at 8p.m. (especially on a Sunday).

After visiting the link, choose a test location that's closest to server (Freephoneline's servers, at this time, are in southern Ontario) you're using. A MOS score below 4.0 is bad news. It means call quality will not be good. The advanced (+) tab provides interesting info.

You should also try the winmtr test I mention in step 7c ii of the setup guide around 8 p.m. to the server you're using.

If the problem only occurs during prime time (as opposed to weekday mornings), then I would probably start thinking your ISP is to blame.

3) Another possibility is that your ISP uses poor routing tables to Freephoneline's SIP servers. This usually causes pings to increase (not necessarily jitter). Large ISPs typically won't do anything if you complain. Smaller ones might.

But no other devices are being used when I experience choppy audio!

A lot of people say that without realizing other devices and/or programs may actually be using bandwidth in the background. It's really not a good idea, in general, to be using a router that doesn't have a good QoS feature for VoIP.

But if what you claim is really true, then you may be dealing the possibility of congestion during prime time (8p.m. to 11 p.m., especially on Sundays). That's an ISP issue (possibly oversold its service in your area/local node congestion).

You should also try the winmtr test (or if you're on a MAC, maybe this helps): https://www.reddit.com/r/TagPro/comments/2j6qx7/how_to_run_an_mtr_on_mac/

Again, if the problem only occurs during prime time (as opposed to weekday mornings) and especially Sunday evenings, then I would probably start thinking your ISP is to blame. Sunday evening is when everyone in your neighbourhood is home.

How do I set my Obihai ATA's Ethernet port to 100Mbit full duplex (it's not by default)?

http://www.obihai.com/support/troubleshooting/sg/drop

Dial *** 0
Enter option 27 and press #
Press 1 to set a new value
Enter a value of 1
Press 1 to confirm/save
Hang up
Your ATA should reboot at this point.

I'm noticing static. What should I do?

1. First thing I would do is try a different phone.

2. Some users have reported switching the ethernet port to full duplex fixes static noise. Force your Obihai ATA to use 100Mbit Full duplex by doing the following:

Dial *** 0
Enter option 27 and press #
Press 1 to set a new value
Enter a value of 1
Press 1 to confirm/save
Hang up
Your ATA should reboot at this point.

3. Definitely try moving your ATA away from other devices to see if that helps.

4. Some other people mentioned the power supply adapter being an issue here: https://www.obitalk.com/forum/index.php?topic=3814.0

But that was for the OBi110/100 series, I think.

If the issue is the adapter, you may be able to get Obihai to send you a new one, but I think I would just ask for a replacement from the point of purchase (probably Newegg Canada). Also visit http://www.obitalk.com/forum/index.php?topic=8754.0.

If you try dialing *** and still get static, obviously the issue isn't your VoIP provider.

How do I connect an ATA to my house, so that all existing phone jacks work?

You need to disconnect your Telco company's line at the demarc--or make sure power from it is not running to your existing phone jacks. Otherwise, you run this risk of frying your ATA. Visit http://www.voipmyhouse.com/#thesolution.

Also, check out bogolisk and canadaodyowner's pictures/posts over here: http://forums.redflagdeals.com/merged-freephoneline-ca-free-local-soft-phone-line-lifetime-voip-821229/331/. Better to ask them about it than me.

There's a related DSL thread over here: http://forums.redflagdeals.com/phone-jack-wiring-question-voip-1293295/#p16243162.

Are Freephoneline's SIP servers down? My ATA isn't registered.

A. Visit http://status.fongo.com/ to check server status.

B. If the service status website doesn't note any issues, then chances are the problem is on your end. In your Obihai ATA or at Obitalk.com, Navigate to Voice Services-->SP(FPL) Service-->X_UserAgentPort. X_UserAgentPort should be a random port number between 30000 and 65535. Just pick a port number in that range. Change to a new port number in that range. Click the "submit" button, and reboot the ATA. (If you use Obitalk.com to change settings, you will need need to use Obitalk.com).

If changing X_UserAgentPort works, you were dealing with a corrupted NAT connection in your router.

Possibly a NAT router connection was never disconnected or never timed out properly. And, then, the ATA keeps the corrupted connection in a persistent state over and over again. (Credit goes to Mango for this information). Possibly, this problem is due to the router's UDP timeout being in excess of the ATA's Failure Retry timer (RegisterRetryInterval with Obihai ATAs). With FPL, that's 120 seconds.

Thanks to Mango, many of us now understand that in order for ATAs to remain registered and working properly with a VoIP SIP provider like Freephoneline, in particular after power failures, the following conditions must be met:

UDP Unreplied Timeout (in your router) < NAT Keep-alive Interval (in your ATA; for Obihai ATAs this is X_KeepAliveExpires) < UDP Assured Timeout (in your router) < SIP Registration Failure Retry Wait Time (or RegisterRetryInterval in Obihai ATAs)

"<" means less than.

When a modem leases a new IP address, a problem can arise where prior associations using the old IP address are maintained in the router. When the ATA attempts to communicate using the old IP address, the response is unreplied, and then if the UDP Unreplied timeout is greater than the Keep Alive Interval (and UDP Unreplied timeout is often set to 30 by default in consumer routers) a problem arises where the corrupted connection persists. If UDP Unreplied timeout is, for example, 10, and the NAT Keep Alive Interval is 20, then the corrupted connection will timeout or close. A new connection will be created, and everything will work fine.

Another problem can occur when the Keep-Alive interval is greater than UDP Assured Timeout (often 180 by default in consumer routers): the NAT hole will close due to the ATA not communicating frequently enough with the SIP server. In turn, incoming calls may, intermittently, not reach the ATA. Again, X_KeepaliveExpires is supposed to be 20 with FPL.

Getting access to both UDP Unreplied Timeout and UDP Assured Timeout settings in consumer routers may be difficult, if not impossible. Asuswrt-Merlin, third party firmware for Asus routers, does offer easy access to these two settings, which are found under Tools-->Other settings. In part, for this reason, I tend to use Asus routers. However, my understanding is that third party Tomato firmware has these two settings as well. So if your router supports Tomato firmware, that may be another option.

The keep alive interval for FPL is 20. The SIP Registration Failure Retry Wait Time is 120. I use 10 for UDP Unreplied Timeout and 117 for UDP Assured Timeout.

C. Double check your Registration timers (refer to page 19). For RegistrationPeriod use 3600, and RegistrationRetryInterval should be 120. If your ATA makes more than 5 registration attempts in 5 minutes, you may end up being temporarily IP banned by the specific FPL server the ATA was sending registration requests to. If you're temporarily IP banned, you could then try switching ProxyServer to a different FPL server than the one you were previously using (voip.freephoneline.ca, voip2.freephoneline.ca, or voip4.freephoneline.ca:6060), unless you need to use voip4.freephoneline.ca:6060 because you have SIP ALG forced on in your router. The purpose of voip4.freephoneline.ca:6060 is to circumvent SIP ALG features in routers. ***If you followed this guide properly, chances are you haven't been temporarily IP banned and should skip ahead to step D.***

D. You can also try rebooting your modem–>router (wait for it to be fully up and transmitting data)–>ATA (in that order).

Why am I not receiving Caller ID Name information on all incoming calls?

CNAM lookups are dependent upon your VoIP service provider--and not the Obihai ATA.

Freephoneline doesn't do CNAM lookups (I.E., Freephoneline doesn't lookup the name of the caller, using the incoming phone number, in an external database). Unless CNAM is being sent by the call provider, which often is not the case with cellular calls, only the phone numbers are going to appear. CNAM lookups cost money, and for a moderately priced one time fee, FPL isn't going to do CNAM lookups.

http://forum.fongo.com/viewtopic.php?f=10&t=4868&p=25678 (a lot of replies in that thread are incorrect; CNAM will show up if it's sent by the provider)
"Just a reminder, as has been provided in other threads, when the call display feature is enabled, it will only show the information that is being sent by the provider. If that provider only sends a number, we can only display a number. This is especially common for cell phone providers and a number of digital service providers."--Fongo_mike"

So, with an incoming call to your FPL #, an incoming phone number will appear on your callerid, but a name only shows if that info is also sent by the provider. Other VoIP services typically charge a fee per call to do a lookup in a CNAM database (phone number is queried for a name in a database). FPL doesn't offer that option. If CNAM info is passed on by the provider, FPL will display CNAM.

I've heard scary stuff about VoIP 911. Isn't it unreliable?

VoIP E911 is a two step process. With Freephonline, after dialing 911, the initial E911 call centre, which does have my name, address, and call back number, still has to transfer the call to local dispatch (PSAP), which doesn't have my name, address, and phone number.

It's important, when signing up to a VoIP service you're planning on using 911 with that you always keep your address updated on file with them. If you move, update your address. Your VoIP service sends that information to the E911 call centre/Northern911, which they will keep on file.

In some rare instances, I suppose it's possible that Northern911 (I'm guessing this is what FPL and other VoIP services in Canada use, but I'm not sure) may not transfer to the correct local dispatch (PSAP) number (human error happens). Some people I configured services for in the past were very paranoid about VoIP E911 and forced me to do a test call. Worked fine. That is, the first person I reached had name and address info; they ask for confirmation. And the call was promptly transferred to local dispatch and correct address info was given to local dispatch, verbally, by the first call centre. Worked fine each and every time I was asked to test.

How does this compare to 911 with a landline?

Landline 911 is not a two-step process. You don't need to keep your address updated. Landlines are the most reliable for 911 calls. But landlines don't work after your telephone lines have been knocked out by a storm.

How does this compare with Mobile 911?

Mobile 911 is not a two step process. However, they do not have your exact address, but they should have an approximate location (they should at least have the cellular site/tower that's carrying your call), especially if you're in a major city (they may have latitude and longitude). If you're in a rural area, location based on cellular towers may not be very precise. 70%+ of 911 calls are now coming from mobile phones according to the CRTC. Going forward, this is where improvements are going to be made.

Also, keep in mind that with FPL each E911 call is $35. If you dial 911 less than twice a year (or less than every 3 years with Anveo's $1.20 USD/monthly fee) vs. paying $1.50 USD/month with Callcentric or VoIP.ms, you're ahead with FPL. And you're paying an ongoing minimum monthly fee of $3.98 with Ooma. Ask yourself how often you're calling 911. If you're a senior citizen with a lot of health issues, maybe FPL is a bad idea. (And I don't mean to belittle this point. Everyone gets old. Health is a serious matter.) Otherwise, you'll end up way ahead using a FPL in the long run (in terms of cost).

Here's the thing . . . I used to talk to FPL reps several years ago over the phone, back when they allowed tech support calls. And even then an e911 fee was listed (but not in the FAQs), and I inquired about it. I was told the fee was intended to dissuade people from test calling 911--and that people wouldn't actually be charged.

Fast forward to now, and the $35 per call E911 fee is listed in the FAQs. It's listed all over the place. It's certainly enough to prevent me from testing 911 on FPL. Reps are now saying you will be charged no matter what when you dial 911. Is that true? Maybe. Is that enough to scare me from testing 911? Sure. Has anyone been charged yet? I don't know. Anyway, no one is going to be calling 911 using FPL unless it's really necessary now, and if that's the intent, I'm fine with it. And if I really need E911 as a backup (my smartphone is always nearby), it's there for me. In the meantime, I'm not paying ongoing monthly fees for something I'm not using.

Obihai OBi200/202 ATAs with the OBiBT adapter can be paired with smartphones over bluetooth: http://www.obihai.com/obibt.
Then with an Obihai OBi 200/202 ATA, you'd add {911:bt} in your OutboundCallRoute, and then all of your 911 calls on your phones go out over your smartphone's 911 cellular service, **provided your smartphone remains within bluetooth range of the ATA**.

By the way, there's also Anveo's E911 service ($25 USD per year) available through the Obitalk.com web portal, as an alternative 911 service (limited to a maximum of 5 e911 calls per year): https://www.anveo.com/e911obi.asp (click the link for more information). People asking for help with this Anveo E911 service should probably ask canadaodyowner, who is using this service and is also a Freephoneline customer: http://forums.redflagdeals.com/freephoneline-ca-free-local-soft-phone-line-lifetime-voip-821229/338/#p24980477. I have no experience with Anveo's special E911 service.

VoIP E911 is available all the time under these conditions:

1) You have electricity. An uninterruptible power supply (UPS) is always a good idea.

2) Your internet service isn't out.

3) Your VoIP service isn't down.

I don't know anyone who doesn't have a smartphone.

Call Waiting doesn’t work when Follow Me is enabled!

You’re correct. Call waiting won’t work when Follow Me is enabled unless the second incoming call is from a Freephoneline, Fongo Mobile, or Fongo Home Phone number. Calls between FPL, Fongo Mobile, or Fongo Home Phone number, at the time of this writing, are SIP URI calls. SIP URI calls don’t count towards the 2 channel limit (2 outbound, 2 inbound, or 1 outbound and 1 inbound call) that Freephoneline uses.

With Follow Me enabled, when there's already an existing call in progress, the second incoming call will not be forwarded unless it's from another FPL, Fongo Mobile, or Fongo Home Phone number. Instead, the incoming caller (unless it's from a FPL, Fongo Mobile, or Fongo Home Phone number) will hear the message, "The user you are trying to reach is not available", and that second call will also not appear in your online Freephoneline call logs. Only answered calls appear in FPL's call logs (calls answered by FPL’s voicemail system count as answered calls).

With Follow Me enabled, when there's already an existing call in progress, the second incoming call will be forwarded if it's from another FPL, Fongo Mobile, or Fongo Home phone number. However, if this second incoming call is not answered by one of the Follow Me destinations, the call will not be answered by FPL's voicemail. Instead the caller will hear, "The user you are trying to reach is not available."

If someone is leaving a message on FPL’s voicemail, and another call comes in, FPL’s voicemail won’t answer the second call!

You’re right, again. Unless the second call is coming from another FPL, Fongo Mobile, or Fongo Home Phone number, Freephoneline’s voicemail won’t answer the second call.

The Call Forward When Unavailable (Follow Me) Option is confusing!

Indeed. What actually happens with "Call Forward When Unavailable" is selected is this: if the incoming call would otherwise go to FPL's voicemail, the call is forwarded. That has nothing to do with your ATA being offline or your number not working. Someone could simply not answer the call despite your FPL number working, and the call will still be forwarded.

Ring ring ring ring. Oh, no one is picking up. I, FPL, would normally send the call to voicemail at this point, but I'll just forward the call.

That's basically what happens.

In other words, even when the ATA is online and registered, if the call would normally go to FPL's voicemail, the call is still forwarded. The trigger for "Call Forward When Unavailable" seems to be FPL's voicemail--and not the ATA being unregistered. FPL's voicemail doesn't actually pickup though; the call is just forwarded at the moment that FPL's voicemail would normally pickup. The call is then forwarded to your follow me numbers. If no one picks up the call after all that, the call gets dropped, finally, to FPL's voicemail.

Is there a way to only use the Auto Attendant when we want rather than all the time? It can get annoying if I just need to call home to talk to kids but have to press 1 each time when I use my cellphone.

Probably the best solution is what I've underlined and italicized below (you can skip down to it). You can follow my thought process in full here though (basically, I don't think a good solution comes from the ATA–but, rather, from your smartphone).

You could use something similar to the following example (in your X_InboundCallRoute for FPL):

For OBi202
{(cellphone#1|cellphone#2):aa($1);d=6,ph,ph2}

For OBi200
{(cellphone#1|cellphone#2):aa($1);d=6,ph}

Related info is over here: https://www.obitalk.com/forum/index.php?topic=66.0.

d=delay in seconds

So, d=6 is wait 6 seconds before sending the call to the Auto Attendant.

6 seconds = 3 rings, in my case

cellphone#1 and cellphone#2=the caller id (ex. 4161234567 and 4162345678) of incoming phone numbers you want to send to the Auto Attendant(AA).

($1) represents the AA calling you back if you hang up before it answers at the phone number you called in from

ph = phone port
ph2 = phone port #2 on an OBi202

In this example, your phone port(s) will ring 3 times, and then, if no one picks up, the call will be sent to the Auto Attendant, where you can press 1 to continue ringing your phone port(s), if you wish.

The only problem I have in this scenario is that it's inconvenient when you want to reach the Auto Attendant and someone at home picks up the phone first.

So, instead of using a delay (d=6) in your X_InboundCallRoute, ***I suggest inserting a pause or comma followed by 1 into your Smartphone's address book for your FPL phone number for dialing home. And then setup a phone contact number entry in your Smartphone for (when you want to reach) the Auto Attendant without the pause followed by 1.***

For iPhones, visit http://www.imore.com/daily-tip-automatically-dialing-extension-iphone. Ex. (416) 555-5555,1 will automatically ring your phone, and (416)555-5555 will reach the Auto Attendant.

For Android based smartphones, visit
http://www.dummies.com/consumer-electronics/smartphones/samsung-galaxy/how-to-add-pauses-when-you-dial-a-number-on-the-samsung-galaxy-note-3/

How to do I setup and collect log information from an OBi200 or OBi202 for troubleshooting purposes?

Visit https://www.obitalk.com/info/faq/Troubleshooting-sec/collect-syslog-from-OBi for full instructions.

The instructions mention entering an IP address; that's the LAN IP (ex. 192.168.1.x) of the computer or device you have syslogd.exe running on. You need to extract the file from the ZIP archive, and run it: http://www.obihai.com/docs/syslogd.zip.

What phone number can I dial to ensure my outbound calls are working properly?

Visit http://thetestcall.blogspot.ca/. This is a pretty great and free service! You can call either the 416 or 250 numbers for free. Dial #. And then choose one of the options from 1 to 9. There's a free echo test, DTMF test (check to make sure all your touchtone keys are recognized), Caller ID readback, etc.

If you just want an echo test, Freephoneline does provide one at (226)244-4444.

Where are Freephoneline's user to user support forums located?

Visit http://forum.fongo.com/viewforum.php?f=5

Where are Obihai's forums located?

Visit https://www.obitalk.com/forum/

Where is the Redflagdeals Freephonline thread located?

Visit http://forums.redflagdeals.com/freephoneline-ca-free-local-soft-phone-line-lifetime-voip-821229/

Where is the Dslreports VoIP forum located?

Visit http://www.dslreports.com/forum/voip

How do I get in contact with Freephoneline support?

Submit an online ticket at https://support.fongo.com/hc/en-us/requests/new

How do I contact Obihai Support?

Visit http://www.obihai.com/supportTicketFormA