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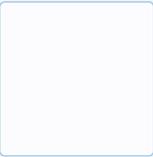
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A Denial of Service vulnerability has been found in all versions of Asterisk 1.8, 10, and 11. [Follow this link](#) for detail

Asterisk Security Alert 500 Apps Welcome

TRY THIS Obihai OBi202: Two phone ports, built-in router, and USB port

Discussion in 'Endpoints' started by Inactive, Mar 10, 2012.



Today I posted some photos and preliminary information on the Obihai OBi202 on my blog:

<http://michigantelephone.wordpress....s-plus-a-built-in-router-and-usb-port-part-1/>

This is not yet what you'd call a review (I hope to post a "Part 2" soon) but I at least wanted to get the information out there for those who may be thinking of purchasing a new device, but don't need it "today, if not sooner". It's supposed to be available around the end of March or the beginning of April.

Inactive
Inactive

Inactive, Mar 10, 2012 #1

lgaetz likes this.



I have posted more information on my blog over the past month (additional articles linked from the first, which has been edited a few times) but just wanted to note that the device went on sale today at Amazon, in case anyone is interested:

<http://www.amazon.com/OBi202-Phone-...3?s=electronics&ie=UTF8&qid=1334236599&sr=1-3>

Also it looks like they are planning to come out with a WiFi adapter that will plug into the USB port on the OBi202 and allow it to connect to the Internet over a Wi-Fi network:

<http://michigantelephone.wordpress....he-obihai-obi202s-usb-port-wifi-connectivity/>

Inactive, Apr 12, 2012 #2

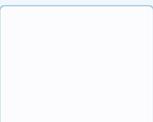
randy7376 likes this.



I've just ordered one. I'll be giving it a run through on its t38 support and comparing it to an SP2012.

atsak, Apr 12, 2012 #3

atsak
Guru



Obihai posted some new documentation links in their forum this afternoon:

OBi202 Data Sheet
OBi Device Admin Guide - PDF

Inactive
Inactive

I do have a question about this unit if I may... With the four accounts available in the unit, I see outside of the usual 1:1 ratio of GV account to PIAF trunk, with one unit per GV account, and potentially a wall of these.

With the Obi202, maybe this would work:

```
GV1 "A" <> obi202 <> "D" piaf
GV2 "B" <> obi202 <> "D" piaf
GV3 "C" <> obi202 <> "D" piaf
```

I can see all three GV accounts would pass through to the PIAF server, but how would one control which GV account would be the outbound number? Is one line permanently selected in the obi202 or can asterisk "tell" it which line to use (GV1, GV2, GV3) through a dial pattern?

Or, would this be better:

```
GV1 "A" <> obi202 <> "C" piaf
GV2 "B" <> obi202 <> "D" piaf
```

and

```
GV3 "A" <> obi202 <> "D" piaf
GV4 "B" <> obi202 <> "D" piaf
```

My attraction to the OBI202 is the idea that I could cut the number of little plastic boxes attached to my communications backboard in at least half, maybe thirds, depending which of the above scenarios is best.

Thank you for any thoughts....

frederic, Apr 21, 2012

#5

Inactive
Inactive

I had a brain fart when I originally posted this response, and after doing so I realized there's possibly a better way, so this is an edited version of what I originally posted.

For starters I will say that I think what you want to do is possible, although at least one of your Service Providers will need to be set to use SIP, not Google Voice. So at a maximum you could have three GV accounts and one SIP account — the SIP account is needed so you can receive calls from your Asterisk server. Note the SIP account does NOT actually have to connect to your Asterisk server (although it can) — as long as you have one, there's a technique you can use to use a Voice Gateway for calls from the Obihai to Asterisk, and you can send calls to the OBi202 by sending them to the appropriate SIP port. My article, [How to use the Obihai OBi100, OBi110, or OBi202 VoIP device as a gateway between Asterisk/F-PBX and Google Voice and/or the OBiTALK network \(UPDATED\)](#) explains this technique, but assumes you only have one Google Voice account.

In that article, I mention changing the X_InboundCallRoute setting for the service provider you send the incoming calls through (can be any SIP-based SP although it's usually easiest to send it to one that's set up as an extension off your Asterisk server, but in the method I propose you are sending calls to a specific port on the OBi202 so as long as that port corresponds to a SIP-based provider it should work). Anyway, the trick there is to use the X_InboundCallRoute setting to direct the calls to where you want them to go. For example, when the SIP-based service provider is an extension off of your Asterisk server, I show this example:

```
{>(011x.|1xxxxxxxxxx):spx},{ph}
```

... where **spx** is the service provider associated with your Google Voice account. This basically says that if the destination number matches the pattern for either a U.S. Canada number or an international call starting with 011, it should be sent to Google Voice, otherwise it should be assumed to be an internal call and ring the phone port (actually phone1 in this case since we did not specify ph2).

But note that we could do additional routing here. Unfortunately, at this point I must admit that my understanding of the intricacies of Obihai dial plans is not what it should be. I am 99% sure that using an OBi202 as a gateway to up to three GV accounts is possible, I'm just not sure of the correct dial plan syntax.

My assumption is that by using the proper syntax you could route calls in the OBi202 based on a prefix on the sent number. So let's say you have three extensions, 222 through 224, and you want to send calls to three different GV accounts. So at the outbound route level you could do something like this (this is very simplistic just to illustrate the principle - I'm showing routing for 11 digit NANP calls, 10 digit NANP calls, and international calls prefixed with 011 respectively):

```
**1+1NXXNXXXXXX/222
**11+NXXNXXXXXX/222
**1+011X./222
**2+1NXXNXXXXXX/223
**21+NXXNXXXXXX/223
**2+011X./223
**3+1NXXNXXXXXX/224
**31+NXXNXXXXXX/224
**3+011X./224
```

You would send all these to the same trunk (the CUSTOM trunk for outgoing calls to GV if you used my technique) but here is the part I'm not sure of. Once they get to the OBi202, you could theoretically do this in the X_InboundCallRoute setting:

```
{>(<**1:>011x.|<**1:>1xxxxxxxxx):sp1},{>(<**2:>011x.|<**2:>1xxxxxxxxx):sp2},{>(<**3:>011x.|<**3:>1xxxxxxxxx):sp3},{ph}
```

and that would direct the calls to the proper Google Voice accounts and hopefully strip off the ****x** prefixes, which Google Voice would reject. The thing I'm not 100% certain of is whether the above syntax is correct for the X_InboundCallRoute setting (particularly the **<**x>** portions that strip the prefixes... I copied that syntax from the Phone port OutboundCallRoute string but don't know if it works the same way in the X_InboundCallRoute).

EDIT: It doesn't, but adding a colon seems to work — instead of **<**x>** use **<**x:>** — I've fixed the string above to reflect this.

If it turns out that my suggested syntax won't work, you could ask about the correct syntax in the [OBiTALK forum](#) (assuming that no one here has a better idea). Just be aware that in that forum the person who will be most likely to answer has some inexplicable hatred of the OBiTALK portal and may instruct you to disable Auto Provisioning, but you don't have to do that if you don't want to; just make all his other suggested changes using the OBi Expert Configuration mode of the portal. Personally I don't care if anyone does or does not use it if they [know the pros and cons](#), but he tells people to disable it without explaining why and I personally think that's horrible advice for new Obihai users (though perhaps not so much for the more experienced readers of this forum, who presumably know what they are doing).

EDIT: Don't forget that by default the **MaxSessions** for any Service Provider is only two, so if you are using one of the SPs as a gateway between Asterisk and multiple Google Voice accounts, you'll probably want to raise that value to something a bit higher (I usually increase it to 10 for any SP that connects to my server, though that might be a bit above the maximum number of simultaneous connections that the OBi202 is capable of handling). The setting is under **SPx Service | Calling Features** and look for the **MaxSessions** option in that section.

EDIT 2: Also, because in my method I use a separate trunk for incoming as opposed to outgoing calls, and because the trunk for outgoing calls is simply a CUSTOM trunk that does not actually attempt to register with the OBi202, there is no reason you could not have multiple TRUNKS with the same Custom Dial String but different Dialed Number Manipulation Rules. This might be easier to implement than changing the existing Outbound Route Dial Patterns, especially if you have a rather large list of Dial Patterns in your outbound routes. So you could make two or three CUSTOM trunks with the **same** Custom Dial String, but in the first trunk you might use the following Dialed Number Manipulation Rules:

```
**1 + NXXNXXXXXX
**1 + X.
```

The first rule prepends ****11** to any ten digit NANP number that hits the trunk (to effectively add a "1" to the start of the number), while the second appends simply ****1** to any other number. Then for the second trunk...

****21 + NXXNXXXXXX**
****2 + X.**

And the third...

****31 + NXXNXXXXXX**
****3 + X.**

Now you can send calls from your existing outbound routes to any one of those three trunks as you choose (**don't** also add the ****n** code at the route level if you do this!). Coupled with the change to the X_InboundCallRoute string on the OBi202, this should route calls to the correct Google Voice account. So basically you can add the prepended routing code in either the outbound route or the trunk, depending on which makes more sense in your particular configuration.

Last edited by MichiganTelephone, Apr 21, 2012

Inactive, Apr 21, 2012

#6

Fixed my previous post - instead of **<**x>** try using **<**x:>** (adding the colon) in the X_InboundCallRoute settings. Also added the edit about increasing **MaxSessions**, and the second edit about optionally prepending the routing codes at the trunk level rather than in the outbound routes.

Last edited by MichiganTelephone, Apr 21, 2012

Inactive, Apr 21, 2012

#7

I meant to get back to this sooner so my apologies for not doing so.

Using the OBI202, I defined things as follows, using the various guides from Michigan Telephone as well as the PIAF OBI110 pdf.

Service1 - PIAF
Service2 - Google Voice Account 1
Service3 - Google Voice Account 2
Service4 - Google Voice Account 3

Inbound:

Call to GV1 --> Obi --> PIAF --> IVR2 (home business 1)
Call to GV2 --> Obi --> PIAF --> IVR3 (cub scouts)
Call to GV3 --> Obi --> PIAF --> Ext 9001 (my direct extension on my PIAF)

I also have the following, but unrelated to Obi stuff:
Comcast Business (8 lines) --> TMD400 (2) --> PIAF --> IVR1 (primary business)

The routes are setup so that:
9+number goes out Comcast, which is how it's been since 2005.
99+number goes out the Obi202, which routes through SP2 which is Google Voice Account 1.

I also set max sessions to six since each google voice number provides two simultaneous paths. I tested this using my eight Comcast Business lines, dialing the google numbers, dialing the "weather" extension, and pressing hold then dialing more.

I did notice that on the sixth simultaneous call, the voice quality of that particular call was noticed to be a little "fuzzy". I dialed the google numbers again in the same manner but in a different order, and again the sixth call didn't sound as good as the other five, regardless which google voice number is dialed last.

I did the reverse - I dialed out of the OBI202 six times to my PIAF to my Comcast Business numbers, putting each one into one of the parking lots and the same thing happened - the sixth simultaneous call through the OBI202 doesn't sound quite the same as the other five.

To eliminate networking as an issue, I unplugged the PIAF server and the OBI202 from our internal network, thus eliminating the switch, routers, vlan, QOS issues and used a simple crossover cable. I got the exact same results in both directions.

I think it's possible that the OBI202 doesn't have quite enough processing/memory capability to be used at it's max throughput. This personally doesn't bother me because the likelihood of all six paths across the six google voice channels being used simultaneously is essentially zero. But, this was an interesting discovery process nonetheless.

Okay, inbound and outbound work, caller ID seems to work in both directions, and I'm pleased with this.

I also eliminated all references to {ph} and/or {ph1,ph2} because my Obi mounts on a communications backboard and there is no analog phone.

The next step is to determine a method to select which google voice account on the OBI202 to dial through - this will for example allow me to use GV2/SP3 to call parents of cub scouts, so when they see the caller ID they see the number associated with cub scouts. This is to ensure the right number shows up in the call recipient's caller ID because we all know when people return calls they just hit "redial". Parents of cub scouts who call back in this fashion, get my main business line then they whine and complain they can't find me.

So, on and off next week, I'm going to work on some Obi dial plans as Michigan Telephone suggested above and see if I can make this work.

frederic, Jul 7, 2012

#8

wardmundy likes this.

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