Our First Experience with Skype Conference Calling

While we were using the latest version of Skype featuring conference calling and a set of tools to test the Internet connection we bumped into a four new sets of issues.

1. Bridging
2. Bandwidth
3. CPU Power
4. Quality of Service Tools

Here is what we discovered—

The concept of "Bridging"

Conference Calling uses one machine to bridge the calls. For example, Bimbobo in Denmark places a call to me in British Columbia, "*Hello Bill. I will now call Steven in North Carolina. Okay?*" Skype connects Bimbobo's machine to Steven's. "*Hello Steven. This is Bimbobo and Bill calling.*" Bill's voice call is bridged to Steven's machine through Bimbobo's. If Bimbobo's CPU and upload bandwidth are appropriate Bimbobo can call and "bridge" frankgoesskype in Germany into our conversation. He did. It didn't work well, because we hit the limit of Bimbobo's machine, but it did work!

Bandwidth Issues

How much bandwidth in required?

When our first conference call was set up Skype chose to use the iLBC Codec. This codec does an analog to digital conversion (and much more) using a bandwidth of about 18 kilobits per second. A two-way voice conversation must support upload bandwidth of twice 18 kbs or roughly 36 Kbs. With three parties on a call the **upload** bandwidth at the host machine was about 54 kbs (for the iLAC codec)

How much bandwidth do you have? Since most cable and DSL broadband connections are asymmetrical, i.e. the upload bandwidth is less than the download bandwidth it is a good idea to make a test so you understand

Notice of Disclosure

The comments made in this document reflect the opinions of the testing group, not of Skype Technologies Ltd.

None of us are technical gurus or geeks.

We are just experimenting with the toy and trying to understand where we can go with it.

We welcome feedback and technical input to correct our understanding.

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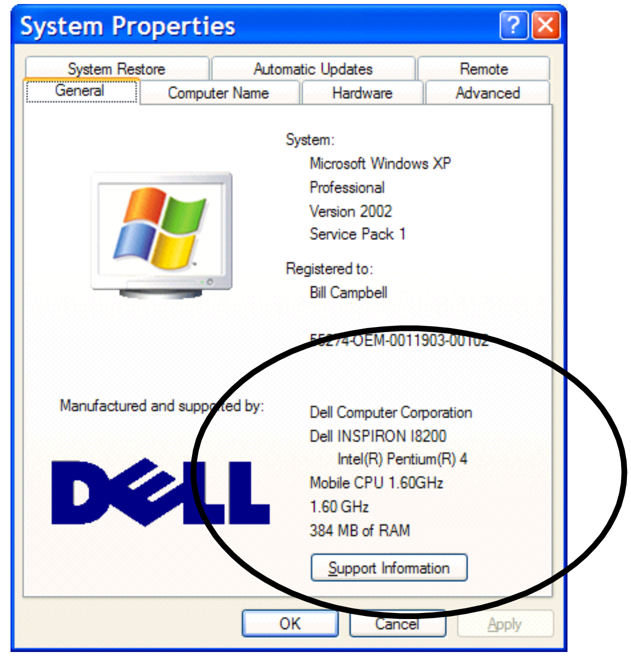
what your bandwidth resource you have to play with. There are many sites on the Internet that over this as a free service. We accessed the **connection page** at pcpitstop http://www.pcpitstop.com/internet/default.asp for our test. Bimbobo, whose machine was hosting or bridging our conference call, had a 128 kbs upload bandwidth so he still has some bandwidth to surf the net while he is in the call. About 74 kbs. Potentially enough to host/bridge several more parties to the conference.



If you are not sure of your processor type and speed, go to Control Panel>System as shown above.

CPU Power Issues

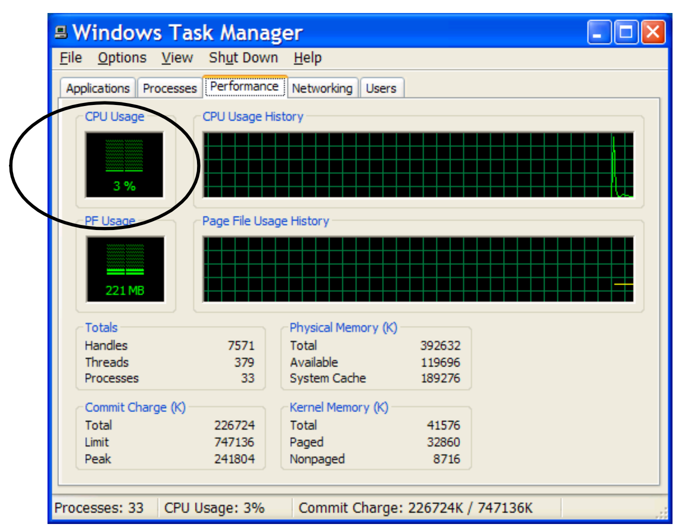
Adding a forth caller to our conference wasn't really possible, although we did accomplish the link. Bimbobo thought his computer was going to melt! The percent CPU utilization went off scale beyond 100. And soon his machine froze up instead of melting.

Executing a Codec takes processing power. Bimbobo's machine is a P III at 450 MHz. (Till his new baby arrives this Thursday!)

We could have guessed this was going to happen, because Bimbobo's machine typically has a forty percent CPU utilization for a Skype call.

Our Internet Connection was pretty clean. Round trip latency between Demark and British Columbia was 265 ms. Most of the time there was little or on packet loss, although at one point it rose to 13 percent it did not noticeably affect voice quality.

We suspect that a poor connection would call for even more CPU cycles.



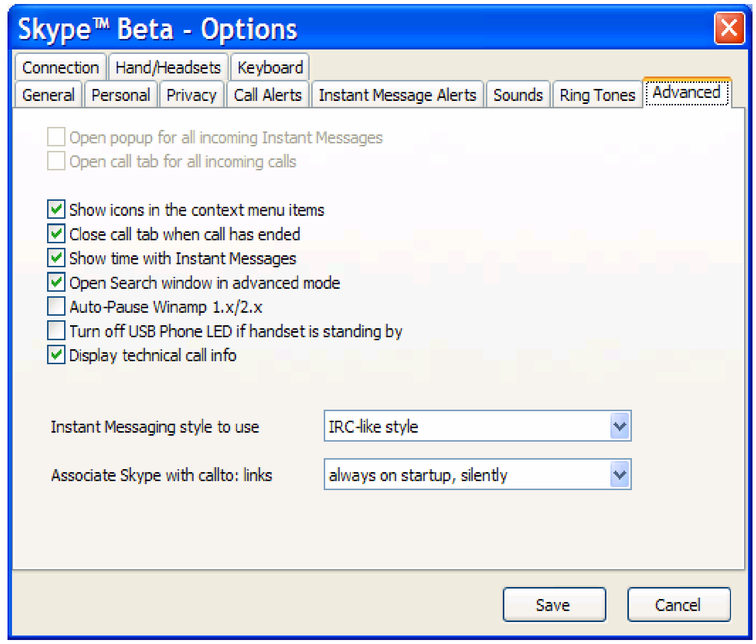
To check your CPU Utilization, hold down the Ctrl, Alt and Delete keys simultaneously. This will open Windows Task Manager. Select the Performance Tab as shown above.

Quality of Service Issues

What affects the quality of service?

The use of a handset or headset and meeting the minimum system requirements as specified by Skype on their Home Page. After that it is primarily four things:

1. Codec
2. Latency
3. Packet Loss
4. Jitter



Skype has added a new feature that allows you to do some simple measurements to help assess quality of service parameters. Tick, Display technical call info.
When is a call, wave the cursor over the black avatar of the of the person you are in conference or the host.
A list of service parameters is temporarily displayed. Such as packet loss.

Codec: This is controlled by Skype when it initiates a call. Skype chooses the best Codec for the worst end point in a connection. We have experienced three different codecs in operation on different types of calls.

1. The iLBC which gave us an upload bandwidth of about 18 Kilobits per second and a similar bandwidth for the incoming voice. All our call between Denmark and British Columbia has used this codec. (Probably because of the 450 MHz CPU)

2. Another codec iSAC that gives us double that bandwidth (about 31 Kilobits per second upload and 27 download)
   http://www.globalipsound.com/pdf/gips_iSAC.pdf
   This codec was used with all other conference parties when hosted on a P4 level machine.

3. Another codec with an unknown identifier that is double #2's bandwidth is used where an end point is a dialup modem (we are guessing here).

We will all learn more and know things more precisely when Skype staff allows us to peek under the hood.

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A document describing some of the components in Skype's voice engine can be found at:

http://www.globalipsound.com/pdf/gips_iLBC.pdf

The diagram on this page shows a plot of (MOS) *Mean Opinion Score* as rated subjectively by a number of listeners (or a set of tools) vs. a couple of common competitive codecs.

Latency: This is the length of time in milliseconds between the two end points. Most calls between Euro-land and North America we have experienced generally fall in around 250 ms. While intra-country calls are mostly under 100 ms. It very much depends on the time of day and traffic congestion. This could be one factor in determining the codec selected and thus overall sound quality.

More: http://rescomp.stanford.edu/~cheshire/rants/Latency.html
http://homepage.ntlworld.com/robin.d.h.walker/cmtips/latency.html

Packet Loss and Jitter

http://homepage.ntlworld.com/robin.d.h.walker/cmtips/loss.html

This site gives some audio files at different packet loss rates and other interesting things about Quality of Service
http://www.voiptroubleshooter.com/voiptr_packetloss.htm

http://www.voiptroubleshooter.com/voiptr_jitter.htm

Summary

CPU Power

To host a conference with five parties you need a big PC, a Pentium 4 or thereabouts. With a PIII CPU of 450 MHz you will be limited to hosting three parties. If you want five parties, have someone else host it

Bandwidth

An upload bandwidth of 30 kilobits per second per party is required. Five parties will require close to the 128 kbs supplied on low cost DSL like Bimbobo's.

Dialup. Well you can't host, but you can participate.

One highlight of our testing was a conference call hosted in Denmark, connecting Mallya in Tanzania to British Columbia. Mallya is on a satellite high speed connection with a round-trip latency of around 800 ms. The quality was remarkably good.

We all had a great deal of fun… franksgoeskype said there was a German word to describe our behaviour, kindergarten. ☺ ☺

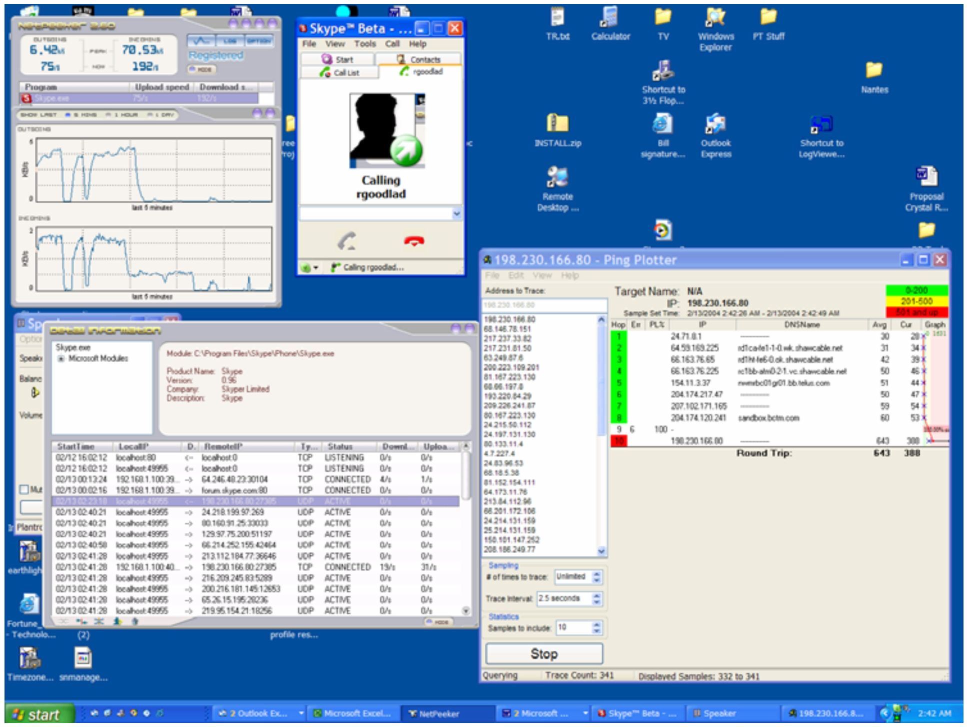
More stuff…

It is always nice to be able to troubleshoot a problem. Is the problem Skype, my machine, the Internet, Bimbobo or what??

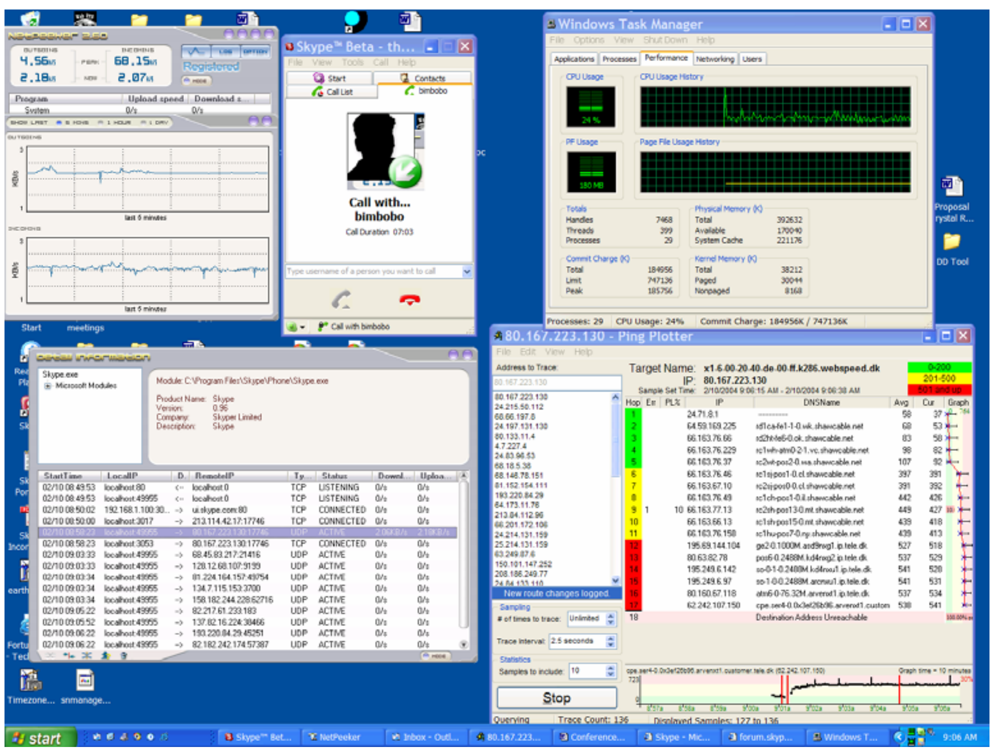
Here are some tools Steven, Bimbobo and I use…and what they tell us…

Ping Plotter, NetPeeker and Windows Task Manager

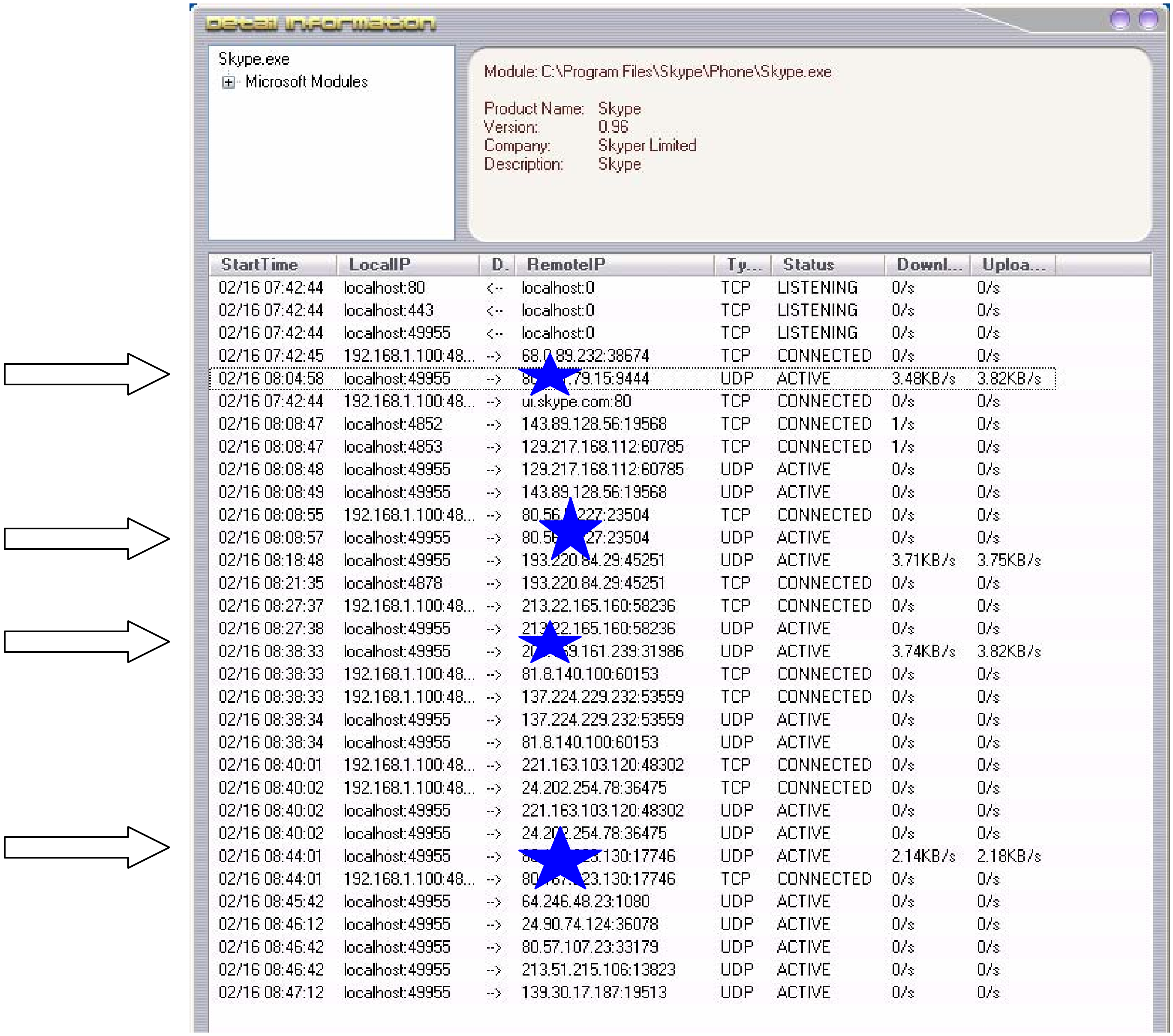
Examples of using the tools.



100 % packet loss!! Call to party using 1X or 3G Wireless (Cell) technology.



First Conference call hosted by Bimbobo’s “Little Engine that Could”



Detail Information

Skype.exe
Microsoft Modules

Module: C:\Program Files\Skype\Phone\Skype.exe

Product Name: Skype
Version: 0.96
Company: Skyper Limited
Description: Skype

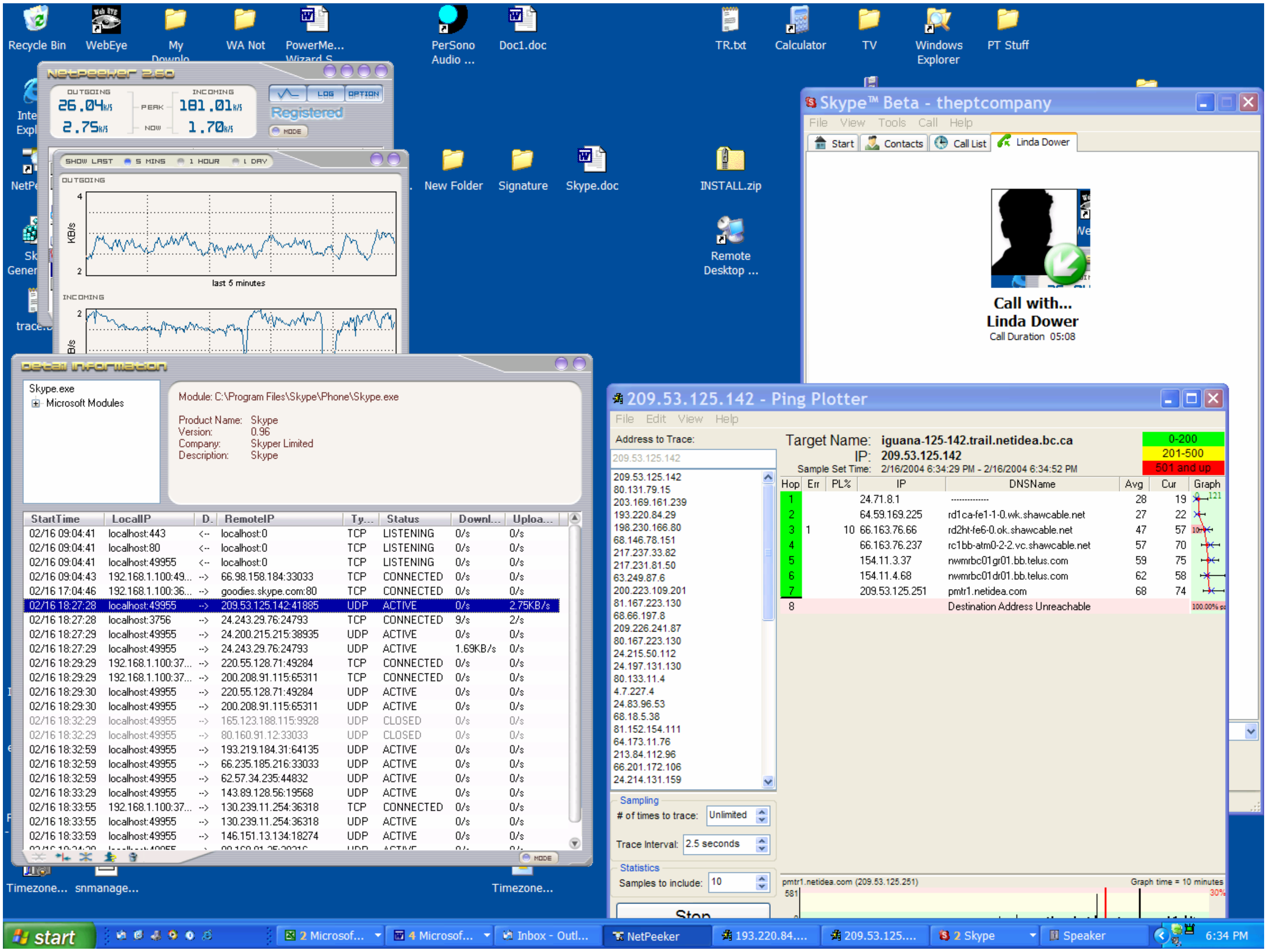
StartTime	LocalIP	D.	RemoteIP	Ty...	Status	Downl...	Uploa...
02/16 07:42:44	localhost:80	<--	localhost:0	TCP	LISTENING	0/s	0/s
02/16 07:42:44	localhost:443	<--	localhost:0	TCP	LISTENING	0/s	0/s
02/16 07:42:44	localhost:49955	<--	localhost:0	TCP	LISTENING	0/s	0/s
02/16 07:42:45	192.168.1.100:48...	-->	68.0.89.232:38674	TCP	CONNECTED	0/s	0/s
02/16 08:04:58	localhost:49955	-->	8[illegible].79.15:9444	UDP	ACTIVE	3.48KB/s	3.82KB/s
02/16 07:42:44	192.168.1.100:48...	-->	ui.skype.com:80	TCP	CONNECTED	0/s	0/s
02/16 08:08:47	localhost:4852	-->	143.89.128.56:19568	TCP	CONNECTED	1/s	0/s
02/16 08:08:47	localhost:4853	-->	129.217.168.112:60785	TCP	CONNECTED	1/s	0/s
02/16 08:08:48	localhost:49955	-->	129.217.168.112:60785	UDP	ACTIVE	0/s	0/s
02/16 08:08:49	localhost:49955	-->	143.89.128.56:19568	UDP	ACTIVE	0/s	0/s
02/16 08:08:55	192.168.1.100:48...	-->	80.56.[illegible]227:23504	TCP	CONNECTED	0/s	0/s
02/16 08:08:57	localhost:49955	-->	80.5[illegible]27:23504	UDP	ACTIVE	0/s	0/s
02/16 08:18:48	localhost:49955	-->	193.220.84.29:45251	UDP	ACTIVE	3.71KB/s	3.75KB/s
02/16 08:21:35	localhost:4878	-->	193.220.84.29:45251	TCP	CONNECTED	0/s	0/s
02/16 08:27:37	192.168.1.100:48...	-->	213.22.165.160:58236	TCP	CONNECTED	0/s	0/s
02/16 08:27:38	localhost:49955	-->	213.22.165.160:58236	UDP	ACTIVE	0/s	0/s
02/16 08:38:33	localhost:49955	-->	2[illegible]9.161.239:31986	UDP	ACTIVE	3.74KB/s	3.82KB/s
02/16 08:38:33	192.168.1.100:48...	-->	81.8.140.100:60153	TCP	CONNECTED	0/s	0/s
02/16 08:38:33	192.168.1.100:48...	-->	137.224.229.232:53559	TCP	CONNECTED	0/s	0/s
02/16 08:38:34	localhost:49955	-->	137.224.229.232:53559	UDP	ACTIVE	0/s	0/s
02/16 08:38:34	localhost:49955	-->	81.8.140.100:60153	UDP	ACTIVE	0/s	0/s
02/16 08:40:01	192.168.1.100:48...	-->	221.163.103.120:48302	TCP	CONNECTED	0/s	0/s
02/16 08:40:02	192.168.1.100:48...	-->	24.202.254.78:36475	TCP	CONNECTED	0/s	0/s
02/16 08:40:02	localhost:49955	-->	221.163.103.120:48302	UDP	ACTIVE	0/s	0/s
02/16 08:40:02	localhost:49955	-->	24.202.254.78:36475	UDP	ACTIVE	0/s	0/s
02/16 08:44:01	localhost:49955	-->	[illegible].130:17746	UDP	ACTIVE	2.14KB/s	2.18KB/s
02/16 08:44:01	192.168.1.100:48...	-->	80[illegible]23.130:17746	TCP	CONNECTED	0/s	0/s
02/16 08:45:42	localhost:49955	-->	64.246.48.23:1080	UDP	ACTIVE	0/s	0/s
02/16 08:46:12	localhost:49955	-->	24.90.74.124:36078	UDP	ACTIVE	0/s	0/s
02/16 08:46:42	localhost:49955	-->	80.57.107.23:33179	UDP	ACTIVE	0/s	0/s
02/16 08:46:42	localhost:49955	-->	213.51.215.106:13823	UDP	ACTIVE	0/s	0/s
02/16 08:47:12	localhost:49955	-->	139.30.17.187:19513	UDP	ACTIVE	0/s	0/s

This is NetPeeker. It shows the four IP addresses of the parties I am hosting, along with the download and upload bandwidth. Look at the one near the bottom. That’s Bimbobo! He get a narrow band codec because he has just a small machine. (till next week)

This was a five party call. Total upload bandwidth 13.55 KBs or 108 Kilobits per second. My P4 Laptop PC was using about 70 percent of available power.

You can enter the IP Addresses in Ping Plotter and do a trace route, watch packet losses, latency changes etc.

The stars are added to hide the IP addresses so the paranoid can sleep at night.



Ahhh!!! Now I know why I can't have a good quality call with my business partner who is on dialup. You can see her download bandwidth hit zero. Her dialup connection on later analysis showed a latency of 3-4 seconds. I had exactly the same problem in a call to Russia today.

This is an error I got when a member of our conference call requested authorization for me to be added to his contact list during the conference call.