Dear reseller,

Please perform the following actions to get an idea about the quality / load of your customer's Internet connection and their local LAN.
We strongly doubt about some issues listed below and for you the task to find out what is going on.

Steps to perform:

-ping to **sip.goandcall.com -t** (for 5 minutes) ; analyze the fluctuations in latency / delays and let us know!
(send us the report with the results)

-calculate the NEEDED bandwidth for EACH of your customer locations using g729 codec:

--> http://www.newport-networks.com/pages/voip-bandwidth-calculator.html

MEASURE the available bandwidth + delays for VoIP usage from the customer's network:

You can use 2 websites for that:

1; Website in the UK (pretty similar speed to XeloQ's systems located in The Netherlands)
2. Website in the USA (good to see the number of total VoIP calls possible)

Website in the UK

http://myspeed.visualware.com/voip

Steps to take:

1. wait until the page loads completely (you need Java Run timer Environment on your PC)

2. Click the button 'Click to start MySpeed'

3. It will take around 1 minute; be patience

4. After it is finished you will see 4 TAB's in the little screen (which you can click) and the Summary screen shows 5 round buttons (green and red)

5. Click around and you will see the results of the Internet connection where you are at that moment

6. In the Summary + Advanced TAB you will see 2 links (below in the screen) that say 'Click here for detailed information' and 'View text'
   --> analyse the results yourself and send it to XeloQ also.

Website in the USA

--> http://www.talkswitch.com/voip/voip_test.asp

This is a test to a USA server but for the total numbers of g729 / g711 calls it is a good test.

Steps to take:

1. Determine your customer's Internet upload speed.

2. Press the button 'Click to start MySpeed'

3. It starts working; it will now measure your download + upload speed

When it's finished you will see the results; the Latency / delay (Round trip time) can be ignored because they test to a server in the US.
The ping results that you tested BEFORE (see above) are important and need to be known.

4. Go to step 2 on the screen --> 2. Enter your Upload Speed
Now manually TYPE the upload speed from the test in 'kbps' and fill it out in the 'Upload speed' box (and select kbps)

5. Now press CALCULATE

6. In the next screen you will see: 'VoIP Bandwidth Test Results'

You can make up to 10 simultaneous VoIP calls using your current Internet upload speed of 414kbps and the default settings on TalkSwitch.

Well now -->> This is ALSO the MAXIMUM number of calls when using the g729 codec on the XeloQ network (XeloQ's default)

In the little overwiew below that you will see this:

Detailed Results
Codec Estimated number of simultaneous VoIP conversations*

Using codec G.711, you can make up to 4 simultaneous VoIP calls

Using codec G.726, you can make up to 7 simultaneous VoIP calls

Using codec G.729, you can make up to 10 simultaneous VoIP calls --> this is the **most important** !

As the last mentioned is the MAXIMUM calls on your or your customer's Internet connection, you should never make MORE simultaneous calls on that internet connection.
If you ALSO use PC's for Internet browsing, this number of simultaneous calls will be LESS.

So test a minumum of 5 times in a NORMAL operational environment and use the LOWEST number of simultaneous VoIP calls you tested.

To be safe even use 1 or 2 LESS than the lowest tested maximum calls.

Then as stated in the Reseller Support Addendum we need ALL this information too:

-used endpoints (brand / type e.g. LinkSys PAP2)

-affected SIP accountnumber(s) - from which SIP accounts does the problem occur?

-describe the LAN Network infrastructure (how are the endpoints connected etc…)

-describe the WAN / Internet infrastructure/connection (type, upload, latency ; ping the sip servers to get this details)

-Traces / analyzed protocol information of what is happening during the problem

-numbers dialed (complete International numbers of the affected destinations)

-date & time when problem occurred

-describe the exact problem that you are facing (describe it as detailed as you can)

We strongly doubt the delays, stability and bandwidth of your customers used local networks / Internet connections.

So TEST all of the mentioned things **ONSITE at the customer's site** at the time the problems exist !

We suggest YOU as a reseller also perform this (read below):

1. Testing it from your own Internet connection at your office is good to have a reference for the customer's reported problems so perform all the tests also from your own office. Write the results down and compare them to the customer's sites that reported problematic calls.

2. Test the numbers that are reported faulty by your customers and DIAL them yourself; see + listen what happens at your office with these numbers and also test YOURSELF at the customer's site after that.

Write the results down and compare them to the customer's sites that reported problematic calls.

3. Use the latest firmware in the VoIP devices and tune the devices concerning packet size, codecs etc...

The LinkSys tuning mail sent out last week is good to have as a reference; use that and experience if it helps you!

After all this has been done, provide us with a detailed report of your findings and let us know if the problems are found and solved!

Good luck.
Support Team XeloQ Communications