IP telephony Questions and Answers

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Q. Could you explain latency and the effect it can have on my traffic?

If your provider is not on high bandwidth connections, you will suffer contention issues and experience latency problems. Latency is the delay in the data packets moving from one server (yours) to the endpoint of the call on VoIP, normally the provider's servers. You should be looking to keep that delay down to around 100ms.

Once you start to move above say 120ms people often find they get intermittent faults.

It would probably be best to look at using one of the Janet connected carriers as they peer with Janet at the tier 1 level in Telehouse/Telecity. This helps get around any QoS issues that you could experience and improves latency etc.

Q. Will my college's 100Mb line would be suitable to handle a SIP connection?

It is a little difficult to answer definitively as you would need to look at the existing traffic on your 100Mb link. A SIP trunk providing around 30-ish channels will occupy about 2Mb bandwidth so that should not pose not a problem. If, however, your link is being maxed out each day then you will suffer issues but you could get around that by dedicating say 4Mb of the 100 to voice only. That is more complex though and brings its own problems.

If you get some test numbers from the carrier you could try a ping test from the desktop to see what the delay is. Anything over 80ms end to end will cause you a problem with calls dropping. You can get an idea by checking what delay is set on your VoIP system parameters.

Links
[1] https://community.ja.net/system/files/222/FAQs%20-%20Telephony%20Purchasing%20Service%20-%20Jan%202015_0.pdf