Rise of the Stupid Network

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A review of the current state of the Stupid Network,

first described in David Isenberg's June 1997 article.

by

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Summary

In a business telephony market worth £1.9bn in 2009, delivering calls to non-geographic numbers the PSTN is still going strong. There has not been a large switch to VoIP calls, except for SKYPE, which carried 300 billion minutes of call traffic in 2011, half of it voice.

Take up of SIP trunks and SIP PBXs by businesses has provided the opportunity for hosted services and opened the market for IP based Intelligent Network (IN) services. The IN market in the UK is worth more that £300m, dominated at present by the large telephone network (PSTN) service providers.

The roll-out of fibre and 4G wireless networks, providing high speed broadband connectivity, is creating an opportunity for Internet Telephony Service Providers (ITSP) to offer alternative telephony services. The time may come when a PSTN based on copper wire will go the way of the telegraph.

Introduction

David Isenberg wrote about the relationship of the 'Stupid Network' and the Public Switched Telephone network (PSTN) in August 1997 [1].

He predicted the demise of the PSTN, superseded by voice communication over the Internet, facilitated by the ever growing power of the devices at the end points of the connection.

The model of single function, unintelligent, end points – telephones – linked by a dedicated circuit through complex switches would be replaced by intelligent end-points controlling their own communication streams, permanently connected by the Internet.

The concept of ‘connecting callers’ would cease to exist because the devices would be permanently connected.

If all call traffic is carried on the Internet would all calls, like SKYPE, be free?

Calls to geographic numbers are already a commodity and are often bundled as 'free' in a telephone service package.

Ten years after David Isenberg's article, in a 2007 report in the Parliamentary Office of Science and Technology [7] it was announced that the British Telecom (BT) IP core network was being rolled out and that it would replace the PSTN by 2012.

> 'BT’s planned roll-out of its £10bn ‘21st Century Network’ (21CN) by 2012 will make the UK the first country to replace its incumbent telephone network with an NGN.' [7]

This report will examine the current state of telephony and try to determine whether the Stupid Network has indeed 'risen'.

The current state of business telephony:

The Intelligent Network (IN) referred to by David Isenberg is all about business customers (and their customers):

- getting the caller to the most appropriate agent
- establishing a relationship with the customer so that your company is the first one called for a class of services/products
- providing easy access to simple information.

The distinction between 'simple' and 'complex' information, referred to above, has more to do with the nature of the question:

- a well defined question with a single answer can be appropriately answered by an automated system (IVR)
- an ill-defined question, with more than one potential 'best' answer is better routed to an expert agent.

Judging by a report from OFCOM [2], the PSTN is still in rude health and generating £1.9bn in revenues in 2009, connecting callers to non-geographic numbers.

The Intelligent Network (IN) services routing the calls to the non-geographic services are generating over £180m per year for number-rental (1.5m NGN services @ £10pm) and £120m per year for advanced feature usage (50k plans using 3 node types at BT Inbound Services prices [4]).

The market for IN services is changing as more business numbers, the termination points of IN services, can be reached via the Internet. As more companies switch their telephony trunks from PSTN to SIP trunks it opens the market for IN services over IP to offer the same services at the edge of the 'cloud' [3].

This provides as least one end of the end-to-end model of intelligent end-points permanently connected by a Stupid Network.

Most individuals still connect using fixed line (PSTN) or cell (GSM) networks, so the caller end of the end-to-end model is still not in place.

For the moment, the caller generated revenue streams to the incumbent Telecommunication Companies (telcos) remain intact.

With the roll-out of high-speed broadband services over optical-fibre and 4G wireless networks this may all change.

In an article in the Sunday Herald (Scotland) on 16 September 2012 [8]:

> Peter Cochrane, former chief technologist at BT, said: "The advent of 4G networks is a significant turning point, introducing the British public to an era of highly flexible, rapid and feature- rich mobile telecommunications that will ultimately result in the home telephone becoming extinct. Nobody under 24 today would consider it either necessary or practical to have a phone installed in their home, and as a concept it will certainly cease to exist within the next five or 10 years."

Current trends in the VoIP Telephony Market

VoIP Servers

A number of production tested VoIP servers are available on Open-Source licenses, as are production capable databases and operating systems. This significantly lowers the barriers to entry both for system developers and integrators, and service providers.

SER/OpenSIPS, sipXecs, and Asterisk are some of the available Open-Source servers.

The current set of VoIP servers has both the functionality and scalability to replace the IN services available on the PSTN.

The open-source VoIP servers mentioned in the previous section have all been deployed successfully in a commercial market. Most incoming calls arrive via gateways linking the PSTN and IP networks.

Client Applications

SIP client applications and devices, particularly in the home environment, have not been widely deployed. The reasons for this are largely to do with the price of the SIP phones and the complexity of configuring the

phone. At present, unless there are significant savings to be made on telephone calls, there is no compelling reason to change!

SIP phones generally have a standard numeric key-pad that does not lend itself to entering a SIP URI. In order to facilitate the use of 'telephone' numbers an ENUM lookup facility is available in order to translate the dialled-number into a URI. This is especially true when a standard telephone is plugged into a SIP adaptor.

Ideally the SIP phone or adaptor should be provided by the Internet Telephony Service Provider (ITSP), preconfigured with the information required to register itself so that incoming calls are immediately enabled. The ITSP will allocate a number to the customer and register the number and URI so that incoming calls can be directed to the customer's phone.

For outgoing calls the ITSP will perform the ENUM lookup to locate the destination and connect the call.

VoIP phones that are available now include:

- SIP desktop phones – with a traditional form but are more complex than PSTN phones to set-up
- Soft phones on a PC or laptop, for SIP or SKYPE, reasonably easy to set-up but have an easy-to-use keyboard – free for download from a number of sources.
- VoIP phone app on mobile and smart-phones, available for Android, iOS and Symbian operation systems. Examples include Sipdroid and FRING.

The number of SIP phones in service is a small percentage of all installed telephones and won't displace them until SIP phones become plug-and-play consumer devices and there are compelling reasons to make the change.

The new IN service providers

Advanced Telephony Service providers that will be best placed to serve the new business IP telephony market will be those that are able to provide the broadest integration of services. These services range from high-volume, high resilience services with limited additional (IN) functionality to complex services that include multi-application integration.

Businesses will be able to utilise customisable services that integrate functionality like speech-recognition, Text-to-Speech and database-connectivity.

The lower barriers to entry offered by the availability of public-domain servers like OpenSIPS, sipXecs, and Asterisk provide an opportunity for industry-specialist service providers to flourish.

It would appear that the biggest change to the Telephony Service market will be a move to smaller, market-niche focused, service providers away from the dominant telecoms companies.

BT's rollout of 21CN has been focused on broadband provision, enabling the BT Vision video product, but opening the door to other ITSPs. In 2007 there was concern that VoIP voice quality was not up to PSTN standard, but with video capable bandwidth voice quality is no longer an issue.

The new Value Proposition

The PSTN is treated as a finite resource and anyone using it to make a telephone call should pay for the capacity that they have used. With Internet connectivity beginning to be seen as an infinite resource [9] then the capacity used to make a call is essentially zero.

Since the cost of making the call is zero, only the utility functions of directing calls to and from the caller can be charged for.

The ITSP has a number of revenue opportunities; the sale of a phone, a monthly service charge, and potentially a 'connection' charge. Call duration should not be a chargeable item.

Revenue will be generated by the provision of ancillary services:

- VoIP phone registration – so that incoming calls can be connected
- ENUM lookup – return a SIP URI for a numeric phone number so that outgoing calls can be directed to the recipient
- IN services – advanced call routing and complex termination services.

With regard to premium rate services, a billing mechanism can be provided where the terminating service provider furnishes the ITSP with caller and duration details allowing the ITSP to bill its client on behalf of the service provider.

Direct access to the destination URI will bypass connection charges, but that means that the caller both knows the destination URI and has the means to enter an alpha-numeric string. Whilst it is true that an IP address and port is a valid URI, most consumer devices don't have a fixed address. Outgoing calls rely on a service provider to locate the destination (determine its URI) and also to direct incoming calls to the device's current address.

In evaluating the risk to call connection revenues accrued to the PSTN network providers, the question then becomes: How soon will VoIP supplant PSTN calls?

There are two parts to this question; what kind of VoIP call, and who are the callers calling?

Based on the availability of VoIP phones, VoIP traffic is dominated by SIP and SKYPE protocol calls, with SKYPE the dominant protocol.

Recently, active, concurrent SKYPE connections passed the 40 million mark [5]. The vast majority of those SKYPE users will make SKYPE-to-SKYPE calls, some to the PSTN via SKYPE-Out, and a very small number will terminate on a SIP agent.

Steve Ballmer of Microsoft reported that in 2011 SKYPE carried 300 billion minutes of call traffic, half of which were voice calls [6].

Steve Ballmer's report about the volume of SKYPE traffic suggests that the compelling reason to switch from PSTN and GSM calls to VoIP may the ability to make video calls. The cost of calls, except for international calls, is unlikely to be a compelling driver. Embedded 'call-me' links in web pages and applications will normalise the utility of VoIP calls, leading to the VoIP phone becoming the default telephony device.

It may be that a compelling reason to switch to VoIP telephony will not be required after all, simply that the VoIP phone app becomes the telephony default.

Conclusion

Whilst the Stupid Network is indeed on the rise, as a network carrying telephony traffic it still has not replaced the PSTN and GSM networks.

When a SIP-based IN service replaces a PSTN IN service, all that is left to the incumbent telcos is the initial call connection, connecting the call originator to the new Internet located service providers.

The incumbent telcos will still get the originating call charge but will have lost the additional revenues of providing advanced IN services.

While the originating network is still PSTN the existing call charges can be charged to the caller and revenues, worth £500m in 2009 [3], passed to the service providers for premium rate services.

If calls both originate and terminate on the Internet then this revenue model is at risk.

References:

1. Isenberg, David – Computer Telephony, *August 1997, pg 16-26*. : http://www.isen.com/stupid.html - http://www.hyperorg.com/misc/stupidnet.html

2. OFCOM - Simplifying Non-Geographic Numbers : http://stakeholders.ofcom.org.uk/binaries/consultations/simplifying-non-geo-numbers/summary/non-geo.pdf : p3

3. OFCOM – Communications Market Report 2012 : http://stakeholders.ofcom.org.uk/binaries/research/cmr/cmr12/CMR_UK_2012.pdf : p321

4. British Telecom (BT) :

- Pricing Homepage - http://www.bt.com/pricing/homepage.html
- Advanced Services - http://www.bt.com/pricing/current/Call_Charges_boo/0020_d0e101276.htm#0020-d0e101276

5. Jean Mercier – Skype Numerology - http://skypenumerology.blogspot.co.uk/
6. Steve Ballmer – CES 2012 (CES keynote) - http://skypenumerology.blogspot.co.uk/2012/01/skype-calling-minutes-in-2011.html
7. Parliamentary Office of Science and Technology - http://www.parliament.uk/briefing-papers/POST-PN-296.pdf :p1
8. Heraldscotland – How 4G is set to kill off the landline - http://www.heraldscotland.com/mobile/news/home-news/how-4g-is-set-to-kill-off-the-landline.18888974
9. Peter Cochrane – The bandwidth charging sham: http://archive.cochrane.org.uk/opinion/articles/silicon/2011/28-06-2011.php

Useful Links:

sipdroid - http://sipdroid.org/

Asterisk - http://www1.digium.com/en/products/asterisk

OpenSIPS - http://www.opensips.org/

sipXecs - http://en.wikipedia.org/wiki/SipX

- https://www.ezuce.com/home
- http://download.sipfoundry.org/pub/sipXecs/

Comparisons:

- Asterisk/OpenSIPS - http://www.voiptoday.org/index.php?option=com_content&view=article&id=231:asterisk-vs-opensips&catid=55:pbx-comparison&Itemid=103
- Voip Clients and Servers - http://en.wikipedia.org/wiki/Comparison_of_VoIP_software

Glossary of terms

NGN	**Non Geographic Number** : e.g. 0800, 0845, 087 etc. or **Next Generation Network** The NGN is that one network transports all information and services (voice, data, and all sorts of media such as video), generally utilising Internet Protocol (IP).
IVR	**Interactive Voice Response** A prompt-and-answer process used to collect caller choice in choosing between call routing options.
IP	**Internet Protocol** The Internet Protocol is responsible for addressing hosts and routing datagrams (packets) from a source host to the destination host across one or more IP networks.
GSM	**Global System for Mobile Communications** (originally ***Groupe Spécial Mobile***) is a standard set of protocols for second generation (2G) digital cellular networks used by mobile phones.
VoIP	**Voice over IP** also known as Internet telephony
SIP	**Session Initiation Protocol** A protocol used to connect a VoIP caller to a VoIP destination.
URI	**Uniform Resource Identifier** In ENUM systems the URI returned will usually be a URL (Uniform Resource Locator) that provides the address of the resource and the protocol used in communication. e.g. sip://your_extension@the_voip_provider

ENUM	**E.164 Telephone Number Mapping** A system of mapping and retrieving e.164 encoded telephone numbers and SIP URI's. Other termination options for a telephone number are also possible (e.g. PSTN, e-mail)
ITSP	**Internet Telephony Service Provider**
ISP	**Internet Service Provider**