10 years of SIP

And some lessons along the way

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- Work for ProVu Communications Ltd
  - A distributor of SIP equipment
  - A provider of hosted phone deployment and management services
  - Who doesn't sell any connectivity
- 10 years experience in deploying SIP phones for open standards VoIP
- My experience is through our reseller base who primarily support small businesses.

Intro

- Small business open SIP
  - VoIP using open standards
- Not big corporates
- And not carrier (like BT Virgin gamma)

Basics

- Get a SIP account from an ITSP
  - Or install Asterisk on a server
- Buy some phones
  - Snom, Grandstream, Cisco, Yealink
- Plug it all in and make phone calls
  - You'll need some internet too

We've come a long way

The UK

- Has a very fluid market with hundreds of hosted SIP providers
- Good wholesale number provision
  - And porting
- Good DSL provision
  - 800k upstream is good for 8 calls
- Open regulation framework
  - Voip seems to be banned in many countries

The market

- Two main routes to market
- SIP trunks with a PBX
  - The PBX itself might be hosted on onsite
- Hosted System
  - The size of these are growing as bandwidth more reliable and cost effective.

Why SIP Voip?

- Usually to add a degree of flexibility to a phone system
- People want clever business features which they can code themselves relatively easily, at significant cost savings to the traditional guys.
- People want to move location
- Never really about cheap call costs
  - But you can save a bit on line rentals

Some Examples

- Logging calls to MySQL for flexible reporting
- Clever call routing
- Pair offices together into one large virtual office
- Easy integration with home brew CRM systems
  - Lots of easy hooks to control phones
- Home workers
  - Bad weather planning

It works

- Using a hosted SIP provider has for 6 years or so been a viable option for a small business in the UK
- Call quality is perfect
  - If you put it in properly

When it all worked

- Probably around 2006 when we had the first SIP phones we could hand on heart say worked well enough to deploy
- Before then, everything was a bit buggy
  - Or at least more hard work

What holds the market back

- Bit of a shaky start in beginning
  - Reputation of dodgy calls from poor infrastructure and bad practitioners
- Lack of peripherals
  - Door entry, tanoys ……
  - But this is largely sorted with a range of products on stream
- Availability of bandwidth
  - Uneconomical if you have a large office outside of an FTTC area

Platforms

- The UK market is largely Asterisk based
- Larger providers tend to use Broadsoft
- Actually, people do carry the Audio through the boxes
  - To keep track of calls
  - And to go through NATs
- Providing good service is about the glueware of numbers, platforms, phones and support

What goes wrong?

- The mystical SIP ALG
- People with duff routers
  - Not enough packets per second
- Infrastructure problems
  - Faulty Lan cabling
  - Some old router, switch ….

SIP ALGs

- SIP `helpers` in consumer routers
- I've no idea why people put them in
- They almost always do more harm than good
- Just disable or run your SIP services on different ports

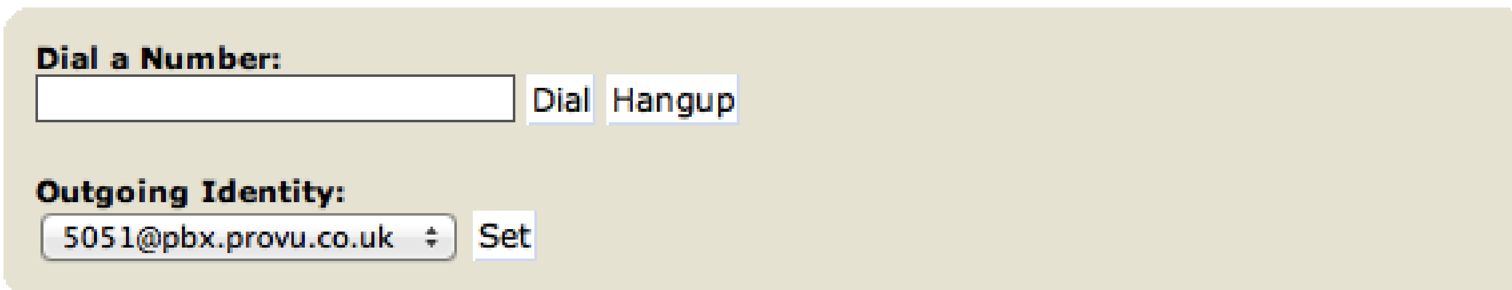
Why Nat is evil

- Port starvation
  - Some consumer routers seem to wimp out at 800 ish sessions
  - Others seem to randomly lose nat state table entries when under load
  - Some streaming services (sky) open and close lots of ports
  - Seems to be a bigger problem on FTTC
  - Symptoms are calls with audio missing in one direction – each call uses another UDP port

Security

- Historically a lot of dial through fraud in telecoms
- Many asterisk PBXs setup around the world provided an easy target for dictionary attacks
- Clear commercial drivers to rip people off big time

Easy click to dial



Hacks

- Scanning for SIP servers and then brute forcing them
- Scanning for SIP phones and extracting SIP passwords from the phones
- Remotely controlling the phone to dial
- Scanning for provisioning servers
  - And yes, we have seen people following redirects in manufacturers redirection servers

Anti Fraud

- All the SIP providers have decent anti fraud
  - They would go bust pretty quickly otherwise
- ISDN providers are usually reliant on downloading billing records from BT
  - Can be days before a problem is noticed
  - Easy to get done for ££££££££
- My view that even the most basic asterisk distributions should have call velocity checks by default

Phone Call Security

- If you can tap the network, easy to listen in
  - Wireshark does this
- For many years phones have supported SIPS and SRTP
- Some phones even have unique client certificates installed at the factory
- But very low usage of these by service providers

Provisioning

- Phones can load settings files using HTTP
  - Most manufactures have a redirection server
- If you have a lot of phones have them talk to a central server
  - essential to keep the firmware up to date for security
  - Consistent settings saves a lot of support
- But, use HTTPS with client certificates
  - Delete the passwords off the server asap

IPv6 on SIP phones

- Nobody does it well enough
- In theory IPv6 helps solve the nat issue.
- Gigaset – works, but single stack only
  - Only on the desk phones, no Dect support
- snom – working on it, but a long way to go. Agree that dual stack is the way forwards
- Yealink – claim support, but can't talk to a router so just one subnet.

IPv6 implications

- Longer SIP/SDP packets
  - So more chance of block fragments
- More likely to upset a SIP ALG
- More overhead if the Voice goes over IPv6
- Just not enough real world experience
- Harder to find a phone on the lan
  - I think you need DHCP with RFC1918 address

Audio Codecs

- In the early days, everybody was about low rate for more calls in the bandwidth available.
- Actually, with overhead, it doesn't save much
- I'd always take the quality option
  - g.711a codec at 64 kbit/s + overhead
- Recently towards about HD Audio
  - G7.22 codec at 64 kbit/s + overhead
  - Improved the quality of their handsets
  - Again – not that much take up by ITSPs

Qos

- A year ago, I would have said packet prioritization was the way to go
- Now I know the answer is just to get rid of bufferbloat – let TCP back off
  - Ok, doesn't help abusive network streams but is fine for most people
  - Just drop packets rather than queuing
- Decent phones do have adaptive jitter buffers
  - Latency is the killer.

ISDN or Not ISDN

- Traditionally, ISDN30 seen as the post reliable type of phone line.
- On fibre, they might be. On copper they fail.
- FTTC or ADSL provide a much cheaper and (in practice) more reliable service in the daytime.

Is video next?

- For business calls, people will not pay a price premium.
  - A lot of people using video on webcam, separate to the phone call
  - Maybe MS Lync will drive this area
- For business meetings, people will use if on a very nice system in a professional video suite.
  - Booking the meeting is the key here

Video Maybe

- Traditionally the SIP videophones on the market
  - Didn't have very good audio
  - Made rubbish business voice phones
  - Way too expensive
- Now starting to see better devices appear
- Not really a SIP carrier (I know about) that does video well
  - I'd like to see somebody launch a service

Lync

- Microsoft's new communication platform
  - Evolving and getting some momentum
- Gaining ground in the enterprising and corporate world
  - Driven by instant messaging and desktop sharing rather than voice
- Little evidence in the small business world
  - Some hosted providers, but these don't seem to have any voice offering
  - Office365 doesn't support voice without onsite servers

Lync phones

- Most of the phones run MS software on third party hardware.
  - Some USB and some ethernet direct
- snom have developed an independent lync firmware for their phones
  - Can run SIP and talk to lync at the same time
  - Good for staged deployment or future proofing

What ProVu looks for in a phone

- Secure web interface
  - Can't get the password out
- Provisioning support for central management
  - Redirection server
- Unique HTTPS/SIPS client certificate in each phone
- Good SIP interoperability
- Audio Quality
- Commercials

My wishes for the future

- More security in the ecosystem
  - Providers supporting TLS and SRTP
  - More proactive vendor security audits in house.
  - Phones delivered without open access to web interface
- More IPv6 support

Any Questions

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