VoIP

Arron Bowley
UKERNA
Projects Engineer

What is VoIP?

- Is it Cisco’s Call Manager Platform?
- Is it a user using “Netmeeting” across the internet?
- Is it two PBX’s connected via a VoIP trunk?
- Is it ISP’s and Telco's interconnecting voice traffic using IP?
- What about H323?

Comparing Voice and Data Networks

- Generally in a voice network the intelligence is located in the core of the network and in data networks at the edge.
- Carriers add services to their telephony networks if they enhance revenues, whereas users add applications where they feel they will be useful.

Early VoIP Deployments

- PBX vendors IP-enable the PBX with VOIP trunks across intranets
- Quality of service is under the enterprise's control, limited low cost installation.
- Low risk of disaster.
- Generally vendor centric.

VoIP LAN Technologies

- Adapters can be used to connect analogue handsets onto IP infrastructures
- IP Soft Switch (such as Cisco Call Manger) switches IP packets rather than streams of voice bits.
- Tend to be very proprietary and expensive to deploy.

Circuit Switched vs Packet Switched

- If the traffic levels are low, channels are idle therefore bandwidth is wasted.
- If traffic levels are high then calls are “blocked” or “congested”.
- In a Packet Switched environment if the traffic is low then users get more bandwidth.
- If traffic is high is unlikely that users will get no bandwidth, just less.

Why Merge Technologies?

- Most organisations have separate voice and data networks.
- In general voice activity tends to be low and therefore a lot of capacity is wasted. Many channels are idle outside of busy periods.
- Potential administrative savings on one network rather than two?

Disadvantages of VoIP?

- Single point of failure, phones and computers.
- Reliability of traditional voice systems is generally better than data systems (currently). IP is a best efforts medium, TDM guarantees a level of service
- Who supports the voice network?
- Call Quality can be a problem across the “unmanaged” internet.

ITU and IETF

- The ITU has traditionally been driven by incumbent telcos, with a strong TDM background
- The IETF has traditionally been a more open forum, with a data networking background
- ITU produced a standard for delivering multimedia across packet switched networks called H323.
- IETF have produced a standard for the initiation of voice traffic across IP packet switched networks called SIP.

Internet VoIP

- Stateful systems such as “Skype” and “MSN Messenger” allow users with a sound card and a microphone to subscribe to voice services and to call other “subscribed” parties across the Internet, using their PC.
- Call quality can sometimes be unpredictable and be ready to receive some unsolicited calls.

Internet Telephony Service Providers

- ITSPs use their own private IP network to route calls thus bypassing the PSTN.
- VoIP calls to the PSTN are routed to a gateway close to the destination of the source of the call.
- There are now many ITSPs worldwide offering similar services and benefits.

So what have we done?

- Commissioned a small Call Manager platform running version 3.2
- Tested PBX interconnection using an analogue line
- Created a corporate directory accessible from the phone, created other services accessible from the phone's display such as weather information and network status.
- Upgraded the Call Manager to V4 and introduced a SIP trunk to an open source SIP server to allow SIP/Call Manager interoperation

What have we done?

- Extended the directory services to include SIP handsets and H323 devices.
- Introduced SIP phones to open source platform and tested call / feature interoperability between Call Manager handsets, SIP handsets and IP Wireless devices.
- Integrated an open source voice mail system for use by both SIP and Call Manager.
- Opened system for limited PSTN traffic using a SIP trunk to a BRI based PSTN gateway

Things to think about?

- Stateless SIP – Registrar, Location and Redirector services to be used by the community.
- Directory Schema, EMOL, E164 numbering plan?
- SIP interoperability between different vendors' SIP platforms
- Interconnecting existing Call Manager platforms?
- Testing voice codecs across the WAN and offering Centrex solutions to smaller sites?
- Interoperability testing of cheaper less well known SIP handsets such as SNOM and Grandstream

Circuit Switched Networks.

- Reliable
- Predictable
- Easy to manage
- Does What it's told
- Ease of Ownership

Packet Switched Network

- Unpredictable
- Tried to train a cat?
- Independent
- You don’t own a cat!