Getting Started With VOIP

Dave Saville

- What we are *Not* going to cover!
- Skype
  - Propriety protocol
  - Bandwidth constraints
    - Google "skype bandwidth"

VOIP

- **V**oice **O**ver **I**nternet **P**rotocol
  - TCP and UDP
    - **T**ransmission **C**ontrol **P**rotocol
    - **U**ser **D**atagram **P**rotocol
- VOIP <–> VOIP or VOIP <–> PSTN
  - **P**ublic **S**witched **T**elephone **N**etwork
  - Or POTS
    - **P**lain **O**ld **T**elephone **S**ystem

SIP

- **S**ession **I**nitiation **P**rotocol
- Open Standard
- Most VOIP Providers support it
- Most equipment supports it

Equipment - Softphones

- No software for ECS (as far as I know)
  - Question asked on comp.os.os2.apps last March
  - Nothing then
  - Nobody seems interested in porting a Java one
- The PC needs to be on to make a call
  - All the time if expecting incoming calls

Equipment - ATAs

- **A**nalog **T**elephone **A**dapter
- Allows use of “normal” PSTN handset(s)
- Plugs into Router
- Usually configured by Web Browser
- No need for PC to be on
- Usually support multiple SIP Accounts
- Often support more than one handset
- May support more than one call at once

Equipment – DECT Phones

- Normal DECT phone used as just handset
  - Only advantage is it is cordless
- DECT Base Station that connects to PSTN and Ethernet
  - Allows calls over PSTN or VOIP from same handset
  - Often allows multiple calls
  - Rudimentary dial plan
  - Browser configurable

Equipment – Asterisk

- Asterisk is the world's leading open source PBX, telephony engine, and telephony applications toolkit.

Equipment - Routers I

- The Firewall problem
  - Need to open TCP/IP ports 5060/5061
  - Need to open a range of UDP ports
    - Typically 10,000 to 20,000

Equipment - Routers II

- The NAT problem
  - SIP device's IP is buried in the transport stream
  - NAT does not “see” it
  - Some routers have a SIP ALG in their firmware
    - **A**pplication **L**ayer **G**ateway
  - Otherwise you need a SIP proxy
  - Or a STUN server
    - **S**imple **T**raversal of **U**ser datagram protocol through **N**etwork address translators

Service Providers

- Give you a SIP account
- Allows outbound calls
- May give/sell you an incoming number
  - That can be dialed from a PSTN phone
  - Can be local or in a different Country

Regulation

- Some Countries will not allow it – eg India
  - Does not stop you calling India
- Some are completely deregulated – eg UK

Making a Call

- "Dial" a direct IP address
  - 81*187*184*102
- Put IP address in phone book
- 123456@voiptalk.org In phone book
- Dial a PSTN number that *is* a PSTN number
- Dial a PSTN number that is mapped to an IP by VOIP provider
- Dial a PSTN number that is mapped to an IP by e164

Useful Links

- www.websitepulse.com/kb/SIP_status_codes.html
- www.mysipswitch.com
  - Handle multiple SIP accounts with only one defined in your hardware
- www.sipbroker.com
  - Provides 'short code' dialing between SIP networks
- www.e164.org/index.php
  - Provides translation of PSTN numbers to VOIP
- http://www.asterisk.org/
  - For your own PBX