VoIP/IP Telephony

Switching Technologies

WHY VoIP/IP TELEPHONY

Deploying a VoIP or IP Telephony System can have many benefits:

- Reduced Costs – using a VoIP system can reduce telephony and related expenses by 30-50% - sometimes more.
- Greatly Enhanced Features – at a lower price per seat, typical VoIP systems have features not found on most traditional phone systems. Some feature benefits are:
  - Greater Mobility – you can easily move your phone within the LAN or WAN – anywhere you have a broadband connection
  - Unified Messaging – integrating your phone services and your desktop applications
  - 4 Digit Dialing between offices – reducing costs and increasing operating ease
  - Centralized voice mail for easy administration
  - Centralized Telephony Servers
  - Use Soft phones on lap top

WHY VoIP/IP TELEPHONY

- Greater Internal Controls – Traditional phone systems require "adds, moves and changes" to be made by a telephony expert. With most VoIP systems, the control is moved to internal staff who can do most everything using easy GUI interfaces. This reduces the cost of ownership and the speed of making changes.
- Central Configuration - for an enterprise with remote locations an IP Telephony System will allow you to have your IP telephony server and voice mail at your HQ and just deploy routers, switches and phones, therefore reducing the cost of costly PBX equipment at each remote site.
- Grows with your Network - when a IP Telephony System is deployed it will offer seamless communications across the entire network.
- ROI – a return on investment can be quickly realized as most companies will reduce their per-seat costs from 50-70%.

IP TELEPHONY

IP Telephony is a System where the phone systems run over a LAN/WAN. This is a departure from traditional phone systems which were required to be more fully integrated with the telephone company (i.e. SBC or AT&T).

- It is a Server based system. Using Raid 1 or Raid 5 Servers (explained later) The OS can be Windows based or Linux based.
- An IP Telephony System routes at Layer 2 (MAC Address) and Layer 3 (IP Address), which facilitates calls being routed from the main server to the proper phone extension.

SWITCHING

Definition: The technique of connecting two or more network end-points network together which involves switching paths, traversing networks and routing decisions. Switching can be Non-Connected oriented protocol or Connected oriented protocol.

Three types of switching which are important to understand include:

- Circuit (CKT)
- Frame Relay (VoFR)
- Internet Protocol (IP)

SWITCHING
Circuit (CKT)

CKT Switching connects two endpoints (of a call) together.

Once a call is established (whether it's a voice call or a data call) there is a continuous path from one end of the call to the other. This path is dedicated solely to that call.

With CKT based switching there is no delay budget -- no delay in the transmission of the Data/Voice.

SWITCHING
Voice over Frame Relay (VoFR)

Definition: Frame Relay is a switching Technology which moves Frame Packets from one Network to another -- connecting networks together across large geographical areas.

Frame Relay was most popular from 1985 to 2003 and was a great alternative to costly leased lines. However, due to changes in technology and improved bandwidth options, Frame Relay is being replaced with Private IP Networks and even the Internet.

SWITCHING
Voice over Frame Relay (VoFR)

Frame Relay is a non-connected packet switched technology, which means there is no direct path between Frame Relay Locations

A Frame Relay connection is comprised of two components:

- Port Speed (Physical Layer/Wire Speed) – this is the total available speed though not all must be used. Port speeds can vary, for example, being 64K, 1.5 MB (T-1), up to 45 MB (DS-3).
- CIR Throughput (Committed Information Rate) – this is the guaranteed bandwidth and is most likely configured at half the Port Speed.

SWITCHING
Voice over Frame Relay (VoFR)

- Frame Relay has a number of mechanisms to minimize delay.
- The ones that are most commonly used are, Frame Relay Fragmentation, Prioritization, and Traffic Shaping.
- Frame locations are connected by DLCI (Data Link Connection Interface). The DLCI identifies the local port on the Frame Relay Switch. Once the DLCI is assigned to the Port or location, an auto discovery protocol discovers all of the DLCI in the Private Frame Relay Network.

SWITCHING
Internet Protocol (IP)

To understand how voice is transported over an IP Network, it is first necessary to understand the nature of IP Networks:

The Internet, the grandest IP Network of all, promised only “Best Effort” delivery. This means the network layer of the TCP/IP stack cannot guarantee delivery of packets in sequence and without delay

This presents a problem for voice-based communication which cannot permit delays in sending signals (makes phone calls sound like video-phone calls from Iraq).

SWITCHING
Internet Protocol (IP)

It is not easy to express an analog signal as a digital stream (ones and zeros). Digitizing voice, which has been in use since the 1970's, works well for VoIP because the Analog Voice Signal is already in Data form.

To do so requires three steps required to "digitize" an analog signal:

- SAMPLING Bandwidth is narrowed to 300-4,000 hz and the analog signal is sampled at twice the rate of 8,000 times per second
- QUANTIZATION A PAM or pulse amplitude modulation is created from the sample
- ENCODING a 8 bit code is assign to express the PAM Signal and 1's & 0's are transmitted across the T-1.

Once on the other side the steps are reversed and an analog signal is presented to the end caller.

SWITCHING
Internet Protocol (IP)

In IP Networks, Transmission Control Protocol or (TCP) can resolve packet ordering problems by using sequence and acknowledgement numbers.

Voice should be carried using User Datagram Protocol or (UDP). It is necessary to use UDP as the transport protocol for VoIP.

IP Addresses are used to identify devices or IP Phones connected to the Network. The address is 32 Bits and is expressed in decimal such as: 10.1.1.12.

SWITCHING
Internet Protocol (IP)

MPLS or Multiprotocol Label Switching

MPLS combines the best aspect of high-speed switching with intelligence of routing using labels to simplify packet forwarding and adds a level of control without impacting performance.

MPLS label forwarding occurs when the Label Switch Router (LSR) performs a label look-up on an incoming packet, swaps the incoming label and forwards the packet to the next LSR along the label switched path.

SWITCHING
Internet Protocol (IP)

- MPLS Can Perform:
  - RSVP- Reserve Bandwidth for a session
  - Policy-based routing
  - DSCP-Differentiated Service Code Point (Priority)

- All Packets group into the same LSR receive similar treatment along the LSP path

IP TELEPHONY
Session Initiated Protocol (SIP)

- The ITU-T Standard for call control and system set-up is SIP (Session Initiation Protocol) an open standard.
- SIP has emerged as a lightweight and extensible alternative to H.323 H.323 is an older ITU-T (International Telecommunications Union) standard designed to control Real-Time Applications across a LAN or WAN. It is used for: Call-Control, Call-Set-up,... This first applications to use H323 were “NetMeeting” and some VoIP.
- SIP components:
  - SIP Server (Network)
  - SIP User Agent (Client)
  - SIP Trunks (Voice Channels)

IP TELEPHONY
Session Initiated Protocol (SIP)

- SIP location Server
  - Tracks which IP address a client (phone) is currently using.
  - SIP Messages:

Invite -	Join a Session/call
ACK -	To Accept the Invitation
Options -	Determine the Capabilities of the Server
Register -	Register with the SIP Server
Cancel -	Cancels a Issued request
BYE -	End Call

IP TELEPHONY
Session Initiated Protocol (SIP)

- A SIP Server should be a RAID 1 or a RAID 5
  - RAID 1 servers have mirrored hard drives
  - RAID 5 servers have at least 3 mirrored hard drives with failover and hot-swappable with 2 independent power supplies.

- The SIP Server can act as a:
  - DHCP- A pool of IP addresses which are parsed-out to the IP phones. This saves time because you don't have to enter in all of the IP addresses for the devices on your IP telephony network.
  - TFTP- This where you would use option 66, so the each network device (i.e. IP phones) can get its configuration from the SIP server.

IP TELEPHONY
Session Initiated Protocol (SIP)

- One Important standard for IP telephony is RTP (Real Time Protocol)
- RTP is a protocol designed specifically to handle the needs of real-time communications.
- RTP has embedded protocols to help with:
  - Sequence Number Incremented for each RTP Packet
  - Time-Stamp used to record the sample rate and therefore playback rate.

IP TELEPHONY
Session Initiated Protocol (SIP)

- Once a RTP Stream is connected between Ethernet Ports or connections to the PSTN the SIP server is done for that session. The SIP server will be employed again when the session needs to request options, such as transfer, on-hold or BYE end call.

QUALITY OF SERVICE (QOS)

- Voice Quality is based on users experiences and expectations. In general, today business telephony systems have virtually no noise, echo or delay.

- In an IP telephony system, noise, echo or delay can be easily addressed. To reduce noise, and echo you can deploy the following – which can be done in the Layer 3 routers, Layer 2 switches or perhaps in the devices (phones).
  - ECHO CANCELLERS
  - JITTER BUFFERS

QUALITY OF SERVICE (QOS)

- In a Local Area Network latency is not a problem. Even with only a 10 Mbps a LAN takes only 51 microseconds to transmit a 64 byte packet and most LAN’s today are running at 100 Mbps or even giga-bit speeds.

QUALITY OF SERVICE (QOS)

Prioritizing voice traffic on the network is important.

- Creating VLANS on the network provides the most flexible QOS. On the same Ethernet port 2 or more VLANS can be created on the Layer 3 routers or switches – one for voice and the other for data. For example you can create 2 VLANs VLAN 101 (voice) and VLAN 105 (data).

- Each VLAN will have it’s own subnet address.

QUALITY OF SERVICE (QOS)

- So, you can have all of your voice traffic on VLAN 101 and within this VLAN you will give priority to the voice packets.
- For example for all voice packet traversing VLAN 101 you can set the TOS (Type of Service) in the IP header to the number 5, which is the highest priority.
- VLAN 105 will be used for all other data traffic.

QUALITY OF SERVICE (QOS)

- Next we will address what QOS can be implemented when the voice packet needs to traverse a Wide Area Network.
- Because your WAN will have limited bandwidth you must use compression when the voice packet reaches the edge of the network. For this we can use G.729a an 8K CODEC.
- When your voice packet reaches the gateway or edge router of the WAN you will need to transcode from G.711 (PCM 64K) to G.729a. DSP or Digital Signaling Processing will be used for the transcoding of the 2 CODECS.

QUALITY OF SERVICE (QOS)

- The standard CODEC within the LAN is G.711 (PCM 64K)
- The Standard CODEC within a WAN is G.729a (8K Compression)
- The ITU-T Speed standard for voice packets is 150ms one way or 300ms round-trip. Anything over this standard will cause the voice quality to suffer.

CONCLUSION

- IP Telephony is ready for “prime-time”
- Quality of calls is now on par with traditional phone systems
- Integration with other communication tools and unified messaging systems, important for businesses, is simplified using data-driven systems.
- Existing market for re-sale of current phone systems will be disappearing as new technology becomes more pervasive
- Benefits include:
  - Greatly increased features
  - Greater flexibility of use (anywhere there’s a broadband connection)
  - Easily deployed over an existing network (LAN or WAN)
  - Scalable with your network growth
  - Cost Reductions of 50-70% per installed seat
  - More flexible and lower cost calling plans
- Availability of reasonable cost bandwidth is now more commonplace
- Cost of systems has come down

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