Technical Training and Consultancy to the NEW Communications Era

VoIP and 'Quality' of Service (QoS)

What is Quality of Service (QoS)?

Quality of Service (QoS) is a descriptive for defining the 'Quality' of a telephone conversation. QoS is not a standard or protocol, but simply a generic industry term for outlining technologies, standards and strategies to provide for 'Quality' telephone / telephony conversation.

Defining QoS 'Quality'

With traditional telephony telephone conversations, we almost always experience 64kbit, G.711, PCM encoded voice. Scored using methodology such as the Mean Opinion Score (MOS), this generally provides for good 'quality' voice conversations (MOS score of 4 or more), and is often termed 'toll quality' voice.

With IP Telephony, we are looking to provide similar 'quality' levels and provide for consistent user conversation experiences.

Measuring QoS and 'Quality' telephony

As mentioned above we can measure the quality of a telephony conversation using a measurement called Mean Opinion Score (MOS). MOS is generally a 'subjective' of quality, and relies on the end persons providing a score to a conversation.

We could and do measure VoIP telephony, using MOS in the same way. However there is an alternate 'objective' method which is a standard called G.107, which produces an 'R' value from measurements to the actual impairments to the conversation such as jitter, latency and loss (using RTP & RTCP)

What are the main effecting QoS issues & impairments?

Latency *:* The BIGGEST effect to impairment and quality. The delay / duration of time a conversation takes from person to person. Example latency effecting factors could be the phone, voice encoding / codec, the network (switch, routers, gateways and encryption). Latency should always be kept to a minimum, complaints start to occur when one way latency exceeds 75ms (150ms 2 way) – (ITU recommends <100ms 1 way)

Jitter *:* 'Data' networks can experience different loading conditions over short spaces of time. This results in small variations of time from when traffic is sent to it being received (the receive 'rate' is not constant). Jitter has little effect to time tolerant traffic, but for voice, we apply a 'buffer' to soak up these differences and produce a flat line condition so that we receive and process voice in a constant manner. The best buffers are dynamic and adapt their buffering to the network conditions, generally allowing for buffering between 0 and 50ms (buffering is seen when network jitter is poor, with little buffering when jitter is not experienced) . Note: any form of buffering adds to latency.

QoS issues & impairments (Cont)

Loss *:* Loss normally occurs in networks when buffers and queues are exceeded and there is no place for traffic to go (too much traffic). Voice cannot be subjected to too much loss (pieces of conversation would be lost as result). Recommendations are that <2% of loss can be experienced without loss of perceptible quality.

Bandwidth : Clearly the throughput capabilities must be matched to expected loading (eg. you cannot make 100 calls over a modem dial-up link)

User Complaint	Chief Causes
I keep "talking over" the far end	Network delay, network jitter
I (or far end) hear(s) echo	Impedance mismatch, poor physical design of handsets, low quality echo cancellation techniques
Voice is too soft or too loud	PSTN loss, digital loss, automatic gain control, conference loss plan, etc.
Clicks, pops or stutters in the call	Packet loss, digital loss, clock drift, jitter, false DTMF detection, silence suppression algorithms, etc.
Caller sounds muffled, distorted, noisy, etc.	CODECs, transducers, physical design of handsets/speakerphones, environment, analog design, etc.

Impairments seen in VoIP Calls

The QoS Strategy (how to approach QoS)

There are two basic approaches to creating a QoS strategy:

- ***Integrated Services (IntServ)*** - Is an approach where the endpoints and application can 'work' with the network infrastructure to provide required resources and conditions required to produce a 'quality' telephony conversation. Example – Resource Reservation Protocol (RSVP)
- ***Differentiated Services (DiffServ)*** - Is an approach where the network devices (switches, routers and gateways), provide pre-defined resources and conditions when ever a packet or traffic type is encountered. Each device is configured in such a way to prioritize voice and other time dependant traffic over non time dependant traffic. Often termed as a per hop behaviour (PHB) Class of Service (CoS) approach. Example L2 - VLAN & 802.1p, L3 – TOS, DiffServ, DSCP and Expedite Forwarding, L4 – port priority, L5 upwards – application priority

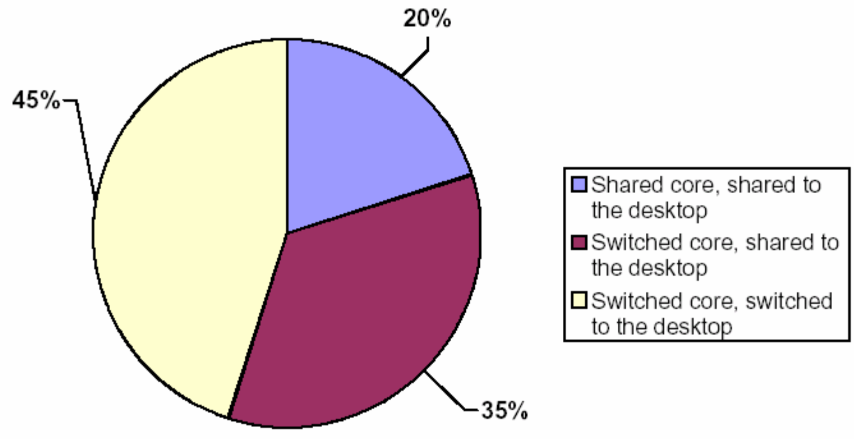
Which QoS approach to choose?

With expensive / limited bandwidth availability, the favoured approach to date has been on a device by device basis, and simple DiffServ PHB (classification, marking and queuing) approach.

IntServ QoS strategies have not been looked upon favourably within the LAN particularly, as they have been difficult to implement and maintain, and also see network control (traffic shaping etc…) being taken away and placed in the hands of the applications themselves.

Is your network Ready?

The majority of networks will support IP telephony, but the majority of networks will not support successful 'quality' IP telephony as the chart opposite shows. This chart shows the status of enterprise migration towards 'switched' networks away from 'shared' networks. A starting point always when discussing network readiness is whether the network is switched, or not or partly?



Network Readiness & Network Requirements

Early implementations of IP telephony solutions were not particularly successful in producing 'quality' telephony results. Not considering and addressing (perhaps not even understanding) the 'key' network requirements to minimize impairments, showed us just how important this task is in making the network 'Ready' for supporting 'quality' telephony.

The below identifies a number of important network considerations and requirements for minimizing telephony impairment and maximizing telephony 'quality'.

Switched Network - As we have seen, we MUST realize the network topology, and that the local delivery to the desktop is completely 'switched'. Without this, we are in real danger of competition between voice and other traffic - with collisions inevitable, we will see impact to latency and knock on to jitter as voice gets though 'when it can'. Little can be done at Layer 1 to prioritize one traffic type over another, some vendors have been seen to 'cheat' standards such as Ethernet CSMA/CD to 'grab' access to the network and gain control, and provide some form of priority – however, going forward with switched networks will negate this issue.

VLAN segmentation – Recommended, placing telephony into a separate network VLAN, keeps administration simple, gives access to possible Layer 2 QoS parameters (802.1p), protects endpoints from excessive network broadcasts and also helps with security for access and also possible Denial of Service attacks.

Network device support – Whichever QoS strategy or approach you make (IntServ or DiffServ), the relative support of this strategy MUST be followed on a network device by device basis (switch / router). This means that each and every device must be capable and configured correctly – missing a single device can undo a lot of hard work, sometimes re-classifying traffic and creating no QoS/CoS parameters / metrics from that point on, or simply not agreeing terms for possible selection / use.

Network 'uplinks' – These are the network 'bottlenecks' between network switches (eg. link from workgroup level switch to distribution switch). Often relatively small (in consideration to potential throughput demand). Address by adding capacity to the uplink or by implementing QoS strategy, VLAN and 802.1p prioritization (as above).

Router 'interfaces' (to the WAN) – BIG potential 'bottleneck' problem area – WAN network capacity is relatively expensive, so adding capacity is financially painful – consider a L3 QoS strategy with DiffServ or IntServ approach, but remember you cant magic up throughput, if the link is already well utilized you may have to manage it with QoS and with the purchase of additional capacity (traffic volume dependant).

Network Readiness & Network Requirements (cont)

Queuing approach, types and technologies – Following on from the last two considerations (uplinks and interfaces), there will always be possibility for queuing (clearly the less network utilization the lower the queuing probability). In real life, networks are finely balanced between cost and service demand – queuing is almost always necessary.

Simple First in – First out (FIFO) queuing is often used at peak network demand and periods and is fine for most time tolerant traffic. More intelligent and sophisticated queuing is beginning to be implemented to realise potential across network links and interfaces as well as providing a great means for prioritization of different traffic flows (eg. IP Telephony). Examples of queuing used for IP Telephony are Priority Queuing (queuing non priority traffic and allowing through priority traffic) and Class Based Queuing (CBQ), which is very sophisticated and assigns 'classes' to traffic, again setting priority, allowing bandwidth allocation and even 'borrowing' across interfaces / links.

Policy Management – Defining a QoS strategy is extremely important as we have seen. Static (device by device) implementation of a QoS strategy (and the upkeep of) can often prove extremely difficult. Vendors realise this, and are producing strategy and policy management software tools to take some of this pain away by centralising management and propagating decisions / changes.

Telephony Solution – The chosen VoIP / IP Telephony solution can have an impact to the quality and to the overall user 'experience'. Whilst there is keen promotion of standards within the vendors and the solutions they provide, many standards are simply not ready for use and adoption, or don't provide the necessary scope to be used successfully. Almost all available IP telephony solutions that exist today are hybrid standards and proprietary based, which provide for differences in overall telephony 'quality' and user experiences. We are encouraged that the overall market trend is towards standardisation, but believe that there will always be differences between vendor solutions, providing healthy competition and selective differentiators.