Quality of Service Challenges for Voice over Internet Protocol (VoIP) within the Wireless Environment

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Abstract

Voice over Internet Protocol (VoIP) is a rapidly growing user technology. Uptake in VoIP technology has significantly increased, especially more recently with the home user moving VoIP away from business dominance. The general population have become accustom to the excellent Quality of Service (QoS) that is now expected from the telecommunications industry both within the landline and mobile infrastructures. QoS within VoIP poses many challenges to be able to compete with the traditional telecom infrastructures. Therefore protocols and standards used to manage today's evolving networks must change to cope with the present challenges of new technologies. This is especially true if you wish to evolve VoIP practically within the wireless domain. This paper investigates current protocols supporting VoIP services, then identifies current networking technologies to improve their QoS, analysing their viability as a solution

Overview

VoIP is a technology that provides voice communication over the Internet with the aim to equal or improve on the traditional services provided via the telecommunication industry, but also enabling a consistent Quality of service (QoS) globally. From the business perspective VoIP is being rapidly adopted mainly due to the financial attraction of reducing call costs by routing communications over IP based networks. Some telecommunication companies are migrating voice and data networks to converge IP-based networks therefore creating architectures that will provide a platform that converges all communication services [1],[15]. But not all countries have such sophisticated IP network infrastructures, nor welcome such communication advancements, that could provide equable QoS. For instance developing countries still pose infrastructure challenges (see figure 1).

"It is now no longer a question of whether VoIP will wipe out traditional telephony, but a question of how quickly it will do so. People in the industry are already talking about the day, perhaps only five years away, when telephony will be a free service offered as part of a bundle of services as an incentive to buy other things such as broadband access or pay-TV services." [2] With the

propagation of broadband Internet access and the emergence of integrated real-time applications, VoIP evolved from the ability to stream real-time multimedia over the Internet rather than the traditional telecommunications infrastructure. Though the ability to transmit voice over the Internet has been available since the 1960's [3] the constraints of the technology at that time meant that it was never pursued as a viable application until the last few years. Even with the development of applications such as Vocaltec [3] in 1973 VoIP was still not a viable alternative to the telephone, possibly due to the accessibility of the equipment required such as computing devices and Internet connections etc. VoIP QoS requirements depend on which codec is used.

Currently established networking architectures were never devised with such technologies in mind. They are passive entities that provide no extra facilities other than to move data from sender to receiver. VoIP remains sensitive to network performance degradation and therefore needs adequate QoS protocols to facilitate it within the network architecture. It is not only the users that suffer when transmitted voice quality is poor, but business also, resulting in lost revenue. This is especially true if the business transactions are phone based e.g. Sales. In this scenario good QoS transmission is crucial.

Quality of Service

Cisco defines Quality of Service (QoS) as the "*capability of a network to provide better service to selected network traffic over various technologies*" [5]. Satisfactory levels of performance should be maintained especially if application requirement demands a guaranteed level of available bandwidth. If the network bandwidth is limited or has high congestion, a QoS mechanism should ensure the propagation of priority data. VoIP has brought QoS issues to the fore, due mainly to the direct comparative user expectations of equal quality to the PSTN infrastructure. Therefore it is not only important to monitor current network quality but also to develop a means to predict expected network quality.

As VoIP is a real time application, it demands a guaranteed amount of bandwidth with low latency to operate efficiently. QoS mechanisms become a crucial factor in the VoIP environment when network conditions deteriorate, for instance if network traffic increases to a level that voice quality degrades to such an extent that it becomes unintelligible. Before this happens a QoS mechanism should step in and give the VoIP traffic priority. Alas, neither QoS nor security factors can be controlled as reliably as in the traditional POTS services [2]. Not all security challenges come in the form of typical network security issues such as denial of service attacks, viruses and unauthorised access to data. Personal security/safety raises concerns as well, these include dial up access to emergency services via 999 or 911 etc; this is currently not available with VoIP technology.

VoIP technologies can utilize the User Datagram Protocol (UDP) or the Transmission Control Protocol (TCP) for transmission. The majority of services use UDP as this reduces the amount of latency on the network and suits real-time traffic, but at a cost. UDP does not provide any assurance of delivery therefore reducing reliability and the order that data is received, which is part of the QoS matrix

The Internet is a network of networks consequently this involves a 'wild' zone in which the data is transported, which is beyond the bounds of control for any single entity. The Internet provides no guarantees or facilities to ensure a consistent data transfer, therefore there is total reliance on the correct implementation of standards and protocols to provide a best-effort policy. This adds further complexity to the challenges of QoS faced by VoIP

If QoS encompassed prioritisation of traffic should all VoIP calls be of the same priority? Who should determine the priority, the carrier or the user? What traffic prioritisation, or level of priority should VoIP be given and by who? For example should social communications via VoIP be given the same priority as emergency calls? The bigger question is, will there be provision for emergency calls, especially as an excellent QoS would be demanded? VoIP may be the key factor to stimulate service convergence, QoS becoming a priority in itself not only to provide a good quality of service for the user but also more importantly ensuring a more efficient use of the available bandwidth for both the user and provider.

Quality of Service Challenges

There are many differing factors that contribute to the degradation of a communication when using VoIP, these encompass latency, jitter echo and packet loss constraints:

Latency –the delay is the time the packets take from being sent from the source to arriving at their destination. This can be affected by, physical distance, the number of router hops, encryption and other overheads and general network conditions. The affect of latency becomes significantly noticeable when round trip times are reaching 250ms. The International Telecommunications Union (ITU) recommends that latency never exceed 150 ms one way from speaker to listener, though the human ear can tolerate as much as 250ms of latency [6]

Jitter - This is the variation in time of packets arriving at their destination. This means that the interval timing of packets exceeds an acceptable level and affects the quality of the voice data. The recommended time for throughput across a communication median is 100ms [16], fast timings can be resolved via a buffer. The affect of jitter can results in white noise interference or substandard audio.

Echo – A consequence of delay is echo on the communication medium. This is a reflection of the original wave to such an extent that this becomes a separate wave to the original on the communication medium and represents itself as an echo to the human ear. This is the case when the delay is more than 10ms [16]. The delay should not be over 65ms nor 30db of attenuation [16].

Packet Loss – Using UDP packet loss is always going to be a potential issue. The level of packet loss affecting the quality of voice transmission can depend on the codec used, e.g it is estimated that as little as a 1% packet loss can seriously affect the quality of the transmission, but if the codec compression rate is high than the effect of a 1% packet loss is much greater [16].

The key elements to providing good QoS in VoIP are in the network design and configuration, the call admission and control (CAC), and Voice Quality monitoring areas of transmission.

Network Design and Configuration.

The challenge is, to ensure enough network capacity is available e.g. bandwidth, to handle Voice calls. Network design and configuration must be optimally designed to encompass good QoS as well as other business constraints [17]. Network monitoring is required to ascertain performance and recovery status. This includes ensuring guarantee bandwidth is available when required, as with the user requirements of VoIP. As different data types need differing requirement it follows that priorities are set for transmission needs and this is policed to the edge of the network. Multi-Protocol Label Switching (MPLS) could be utilised for traffic and recovery services. To enhance the network must incorporate traffic signalling, softswitch and gateway sizing and placement, placement messaging [17].

Call Admissions Control (CAC).

Call Admissions Control (CAC) is provided via the session management/call session control function or gateway. This will provide greater controls at times of significant traffic loads that cause major congestion. During congestion there are greater instances of dropped packets and longer delays. A circuit switched network has full control of the network, if it's busy then the call is blocked, to enable the calls in progress maintain good QoS on dedicated circuits. If packet loss is significant then the call becomes unintelligible to the human ear therefore requiring a request to set-up another call. Recall Rate stands at approx. 80%, which in itself creates switch processor overloads that in the worst-case scenario could bring down the network [17]. Houck and Meempat (2002) recommends a bandwidth management approach that is scalable and efficient while Houck et al endorses a caller to receiver measurement approach providing good QoS to resolve this.

Voice Quality standards

IP-based networks have many elements that can result in degraded voice quality on the communication medium. An end-to-end call consists of the following stages, Call set-up time, call blocking rate and call tear downtime. After the call is set-up it is the quality of the transmission that is paramount for the whole duration of the call. The elements that can affect voice quality is the codec, delay and packet loss. Some of these can be caused by the configuration of the network equipment. This in itself will impact on the overall performance of the network. As networks are dynamic entities there is a need to continually monitor performance.

Means Opinion Scores

Means Opinion Scores (MOS) is the subjective measure of the user perception of intelligible speech that really determines quality. ITU specifies P.800 as the recommended measure of voice quality. This collects the mean scores for voice quality. Human listeners categorises voice quality as excellent, good, fair, poor and bad. Though this is subjective it is the mean score that is recorded, which is generated under controlled lab conditions.

Perceptual Model

This is an objective measure of voice quality. Since the 1990's the ITU have worked to standardize objective speech quality measures. Currently there are two measures, the e-model and perceptual models. The perceptual model compares the signal that was sent to that which was received, this determines the quality, focuses on one-way speech distortion only. The constraint of this system is that a sample of the sent signal needs to be sent to the receiver to analyse whether the original signal is distorted. This puts additional strain on the network. This model only determines the quality of the voice transmission and not why the transmission could be distorted. Therefore there is no mechanism to rectify the fault. This model is suited to artificial, test calls in the lab rather than real-time calls.

E-Model

This is a specified standard G.107 from the ITU. It utilizes transmission parameters to predict voice quality of an average user. This is predominately the best estimation of quality, taking into account the causes of network interference. The E-model then calculates a transmission rating R. This is then converted to MOS. As it utilises the actual elements that cause interference, it is able to inform the network management system of the problem. This is a scalable model unlike the perceptual model.

$$R = R_o - I_s - I_d - I_e + A \qquad (1)$$

To calculate R, take the basic signal to noise ration (R_o) from the end-to-end transmission and subtract the sum of real-time transmission (I_s), delay (I_d) and equipment interference (I_e) and then add the advantage factor of new services (A) (see equation 1). The values for these variables are predetermined within the ITU standard in lookup tables. This helps to maintain consistency and simplicity of the model.

The E-Model can thus be utilized to plan or evolve VoIP networks by using the E-model (see fig 3), delay and packet loss can be estimated. Theoretically implementations of networks can be investigated to determine whether that configuration should be used to deliver the required performance level of the voice quality.

Voice quality Monitoring

The Voice quality can be effectively maintained by utilizing both the e-model and the network management system. The Network management system could poll the routes to forward packet loss, which some router's monitor e.g. Cisco SAA. The E-model focuses on the hardware utilization measures rather than software such as the correct functions of the codecs, whereas the perceptual evaluation of speech quality (PESQ) could test for quality issues within the codec but this has limitation with the accuracy needed to evaluate service level agreements. The draw back of utilising this system to its full extent, is that the data is only available post call for the system. To be efficient real-time monitoring is needed to ensure the quality of the actual call and prevent the need for a recall.

Current Protocol Implementation for VoIP

There are currently several protocols being used to run VolP applications over the network, this section will focus on the two most common protocols by, discussing there QoS techniques.

H.323

The aim of the H.323 protocol is to cope with the demands of multimedia communications systems over an IP-based network. This encompasses both voice and video transmissions. This protocol is ratified by the ITU-T their challenge is to implement a protocol that can achieve the robustness and interoperability that is found on the PSTN. This specification explicitly addresses the use of VoIP and Video conferencing.

The H.323 protocol’s reliability strengths are in its means of recovering from connection failures thus enhancing the protocols ability to scale. As each call end-point report’s to the gatekeeper that enables each communications to be tracked individually. For the VoIP environment this means that such tracking of communications can provide details for call charging. H.323 incorporates procedures to cope with the challenges of lip synchronization.

Though UDP is used for VoIP H.323 increases the reliability by using TCP for signalling to ensure delivery via the acknowledgement element of the TCP protocol. H.323 can utilized the usual media topologies unicast, multicast, star and centralized it also offers encryption and authentication.

Session Initiation Protocol (SIP)

The Session Initiation Protocol (SIP) is fundamentally a session protocol to create communication or a “session” between two points. It has been standardized and governed primarily by the Internet Engineering Task Force (IETF). It was not set up primarily for multimedia transmissions but vendors now utilize its capability within the session to do so. Interoperability problems plague this protocol they have been overcome in some instances by utilising a PSTN gateway. Another challenge that SIPS face is its inability to recover from failure, causing increased delay or packet loss. The delay is compounded by how SIPS use ASCII code for massage encoding rather than binary. Though this is suitable for humans it is extremely complex for IP-based networks. It results in large messages consuming large amounts of bandwidth that further contribute to network delay. Especially when utilizing the WAN as it is highly likely that the message could exceed the MTU size compounding network delay and packet loss resulting in an attempt to utilize binary for encoding to resolve the excessive delay issue. SIPS uses a PSTN gateway to help with load balancing and call time outs as it doesn’t have built in provisions for load balancing it practices a “trail and Error” system.

For address- resolution protocol SIP’s incorporates user agents to route its INVITW message via a proxy server. SIPS processing requirements at this stage are far higher than with H.323, as SIP’s invokes at least 3 messages between the device, proxy and the next hop. Tracking of VoIP communications can only be achieved if the connection is set up and maintained for the duration of the communication this imposes further overheads on the system. As communications are achieved via sessions, SIPS do not provide services via a web browser through XML or SOAP. SIPS only provide limited capabilities for video conferencing, as there is no mechanism for lip synchronization. Neither has it any mechanism to recover from packet loss in stream media such as the ‘video fast update’ command built into H.323.

Though SIPS do not rely on a proxy as it can make direct connections between 2 devices. Though proxy’s are used for registration, address resolution and call routing. Transmission and signalling are conducted via UDP and is therefore unreliable. Though it can provide authentication via SSL, PGP and S/MIME. To be able to provide conferencing capability SIP has to closely couple with other protocols such as SDP and RTP. SDP describes the media content of the session this encompasses the ports and the streaming codec while it is RTP that carries the actual voice content. SIPs make full use of gateway support of RSVP to ensure QoS by resource management. It’s this resource management that reserves sufficient bandwidth and bounds on packet loss and the resulting jitter. Therefore no connection is setup if the required bandwidth cannot be guarantee. A buffer is used to reduce the effect of jitter on the transmission though if this is implemented incorrectly e.g. greater or less than required this method can be ineffective.

Dynamic Transfer Mode (DTM)

Dynamic Transfer Mode **(**DTM) is a networking technology, standardized by ETSI. DTM has been designed to work with video streaming techniques, and is equally applicable for VolP. It works by allocating a capacity to channels by assigning a number of time slots. Similar to Time Division Multiplexing (TDM), it assigns time slots for bit streams to use on a communication channel basically taking turns on that channel. DTM provides a complete networking architecture that needs setting up to provide support with good provision for QoS. The utilization of time slots provides a mechanism to reserve bandwidth for the duration of the transmission, ensuring a static bit rate to avoid traffic congestion.

The issue with this approach is that there is only a finite amount of simultaneous time slots available therefore to manage this, requests for new data streams are dropped or allocated bandwidth is reduced on the current connections.

Path switching

Path switching aims to facilitate QoS for all networking technologies. Path Switching in essence is the ability for a packet to have multiple paths to travel to its destination. The decision to travel a certain pathway is defined by:

1.Measuring and Predicting network performance of various paths,
2.Map measured/predicted network performance into application estimates,
3.Make sensible path switching decisions based on the estimated application quality.

The route taken is determined by comparing the best path available based on path performance measuring. The path selection then depends on the data type of the transmission, which depends on the: codec, buffer size, and error controls.

This decision procedure is carried out by a application driven path-switching (APS) gateway. The APS gateway is assigned by the VoIP application at either end with Sender and Receiver. The gateway then performs transparently and defines the movement of networking traffic (see fig. 4).

By briefly looking into Path Switching and Dynamic Transfer Mode this paper has established two methods of providing extended Quality of Service tools to be used in conjunction with VoIP and other multimedia transmission technologies. Path Switching was the less invasive option of the two and is currently implemented in Skype [4]. DTM is a more invasive option requiring physical set-up within network architecture. This would likely need to be installed on backbone connections to provide a complete end-to-end QoS.

Conclusions

The current QoS standards set by the H.323 and SIP protocols are not ideal, they are generic features that rely on the correct set-up of the technologies involved to allow for seamless VoIP transmissions end-to-end. Initially these were designed to compliment the traditional data transmissions of Ethernet. As broadband facilities propagate to the home user, the evolution of VoIP will gain more support. DTM and Path switching standards aim to provide good QoS and guaranteed transmission but are not without constraints that is mainly the finite availability of required bandwidth.

In conclusion, an adaptation of DTM integrated with path switching based on the performance measurement algorithms provided by the E-Model would enhance the support needed for VoIP. With this in mind, VoIP still has many hurdles to overcome, including the problem with emergency calls, and the fact landlines are detached from the electricity circuit and still work with power cuts acting as a life line. Current VoIP technologies do not, and this has hindered total reliance on mobile technology. Certainly the technologies outlined in this paper will improve QoS when making important calls but the constraints of battery power and QoS continues to limit the potential of VoIP.

Futurework would be to test the combination of the protocols in the mobile domain and provide enhancements as required. Todays mobile phones has inbuilt wireless access therefore, apart from the limitations of battery life, a total mobile VoIP infrastructure integrating with the fixed networks could challenge the traditional telecommunications industry. This may drive a total convergence approach to complete communication networks. This may instigate translations of simple telephone numbers into IP addresses that could possibly provide geographical representation within the IP based system.

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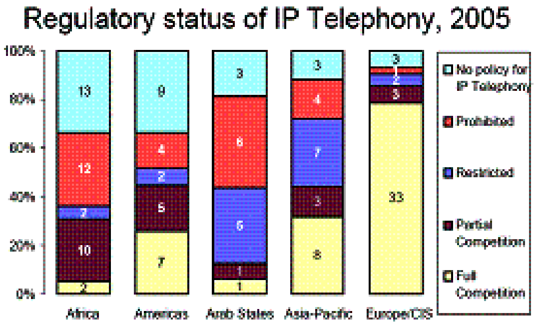
Figures

Figure 1 Regulatory Status of IP Telephony, 2005[1]



Based on responses from 149 economies.
"**Prohibited**"= no service is possible.
"**Restricted**"= only licensed PTOs can offer service.
"**Partial Competition**"= non-licensed PTOs may use either IP networks or public Internet.
"**Full Competition**"= anyone can use or offer the service.
(ITU World Telecommunication Regulatory Database 2005 questionnaire).

Figure 2 - A complete, sophisticated protocol stack [14]

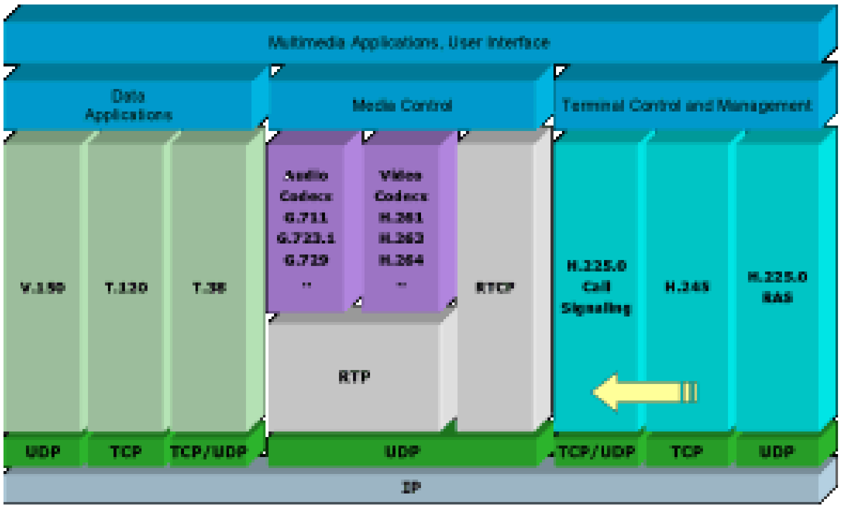


Figure 3 Voice quality compared to delay for different Codec Choices using e-model values [18]

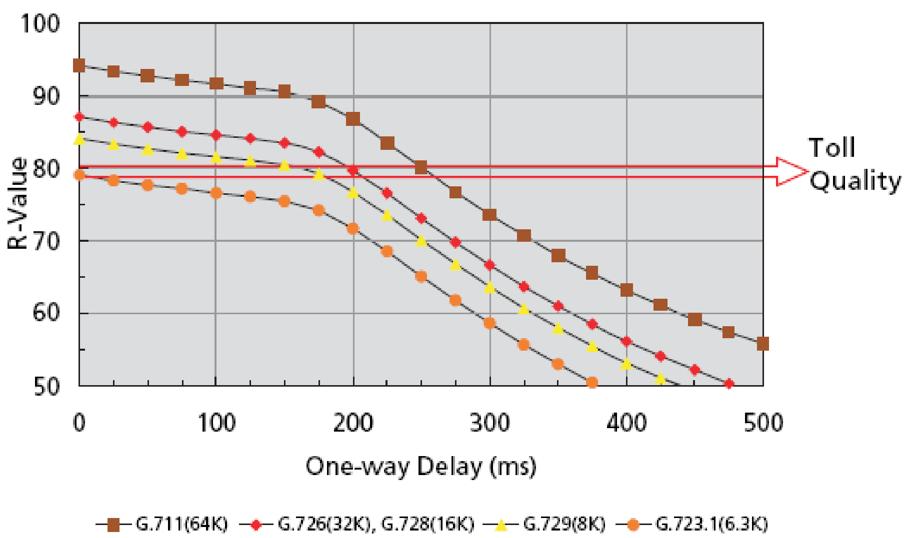


Figure 4. Multiple Pathway Connection