VoIP QoS

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Executive Summary

This white paper discusses QoS (Quality of Service) in VoIP networks. It starts with discussing termination network and terminating partners. It discusses the multilayer terminating partner network currently developing. It introduces basic signaling protocols for VoIP like H.323 and SIP.

In then introduces different parameters that affect Quality of Service over a VoIP network. It introduces terms like Latency, Jitter, Packet Loss, PDD (Post dial Delay) etc. and explains them in QoS background.

It then introduces the media transfer protocols like RTP and RTCP and mentions how they can be used to monitor ongoing QoS.

At the end it introduces how can the concept of "Cost" of delivery of call be generalized to take into account the Quality of Service in it as well as the financial cost of the call.

In the end it mentions strategies for VoIP providers to improve upon their LCR (Least Cost Routing) mechanisms to consider full cost of a call including its Quality of Service. It also discusses what support you should have from your billing system to monitor QoS and do improved LCR.

Introduction

QoS is a short term for **Quality of Service**. The success of any product/service is directly proportional to the quality it retains. With reference to the current scenario in Telecom world, *Quality* and *Cost* are two major factors that can affect the attractiveness of any service. This whitepaper focuses on the "Quality" side of the discussion.

In telecommunications services, emerging use of VoIP (Voice over Internet Protocol) and other services (like VOD, IPTV etc.) has made QoS monitoring an essential element for high-quality service. It is used to monitor the quality of a network in terms of transmission, error rate and other characteristics that can be maintained to improve the quality.

Quality of any service depends on the traffic flow as well as the network of terminating partners. A **VoIP terminator** is one who takes VoIP calls off internet and delivers them to PSTN phones. Therefore, selection of a terminating partner that can transmit your calls to their destinations with better quality is also vitally important. While selecting a terminator, following different issues should be considered to provide better-quality service.

- Number of calls managed simultaneously by the network
- The alternate way to transfer the call to it desired destination in case of any fault/failure occurred in the network
- Supported CODECs for coding and encoding purposes
- Overall setup of the network
- The protocol used by the termination network

Most commonly used signaling protocols are H.323 and SIP (Session Initiation Protocol) and can be used in the same network. Both these protocols are used in VoIP (Voice over IP) and Video Conferencing. H.323 provides compatibility between VoIP equipment and equipment from different manufacturers. SIP is introduced after H.323 but is now much popular for VoIP services. It is specifically designed to attain simplicity and scalability.

H.323

H.323 is an international multimedia conferencing protocol, developed by ITU-T (International Telecommunications Union) in 1996 for communication over Packet Switched Networks (LAN, WAN and Internet). H.323 is extensively used in VoIP (Voice over IP), Video Conferencing and Data Communication over the Internet.

H.323 can manage failure of NEs (Network Entities) like Gatekeepers and endpoints. It also supports recovery of connection failures. H.323 performs coding in binary format that is appropriate for narrow and broad band connections.

SIP

SIP stands for Session Initiation Protocol developed by IETF (Internet Engineering Task Force) in 1999. It allows establishment of different sessions that can be used for communication over the Internet. SIP has no procedures defined to handle or manage failure of Network Entities.

Quality Measures

To establish a network that offers highest level of quality of service, telecom operators experience few challenges such as:

- Latency

  It is the amount of time required to transmit data from source to the destination. It is an end-to-end delay that occurs in information exchange between two nodes. Simply, it can be referred as the speed of the network that can affect the overall quality of the service. Delays in data packets can be reduced by reducing overall packet size.

- Jitter

  Information is transferred from source to destinations via small messages called packets. Such packets experience certain delays to reach their required destinations. The variation in these delays is known as jitter and it adversely affect quality of the service provided. It makes certain sounds due to packet loss but can be managed via jitter buffers.

- Packet Loss

  Data packets can be dropped due to congestion in the network or limited buffer size at the other end. Once these packets are lost, they cannot be recovered unless retransmitted by the sender. Thus affect the speed and finally the quality of the network.

  To reduce data loss, QoS monitoring ensures congestion and queue management via various tools like Priority Queuing (PQ), Custom Queuing (CQ) etc. Queue management prevents queues from filling and provides space for high priority packets.

- Post Dial Delay (PDD)

  On dialing phone number, either there is a ring or busy tone that tells us that whether the called party is available or not. The time elapsed between dialing a number and hearing a tone is referred to as Post Dial Delay (PDD).

- Bandwidth

  Bandwidth is the total capacity of a transmission medium to transfer data. Bandwidth and Latency both can affect the quality of service in terms of speed and capacity of the network. Greater the bandwidth more is the ability of the network to transmit data. Capacity of a network to transfer information decreases if the network is oversubscribed with users.

Media Transfer Protocols

Communication between two nodes is not possible without certain protocols. Correspondingly, well known protocols for media transfer are RTP (Real Time Transport Protocol) and RTCP (Real Time Control Protocol).

RTP is used in transferring real time data like audio, video or simulation data and provides end-to-end network transport functionality for applications communicating in a real time scenario.

RTP ensures link efficiency by reducing large data packets into smaller manageable chunks. Mostly, the payload (actual data) is less than the overload (additional bits in the header) that extends the packet size but the RTP header also known as Compressed Real Time Protocol Header decreases the header size and ensures on-time packet delivery that conclusively effects the quality of the network.

Data transferred via RTP also needs to be controlled by some mechanism. For this purpose, RTCP (Real Time Control Protocol) is used. It improves data transfer and provides data monitoring.

RTP and RTCP provide multicasting, time shaping, sequencing and delivery monitoring. RTP is responsible for media transmission while RTCP is responsible for end-to-end monitoring, data delivery and QoS Monitoring. Both of these protocols are independent of the basic transport and network layers.

Least Cost Routing

VoIP providers mostly offer **Least Cost Routing (LCR)** to ensure higher quality of service. LCR efficiently and successfully transfers your calls at a reasonable cost. It saves time and effort for routing international calls by using most cost effective method for transferring traffic. Thus improving the overall quality of the service provided.

In order to route a call effectively, certain issues should be considered and improved accordingly such as:

Average Call Duration (ACD)

It is the total amount of time taken by the call. In case of lower ACD, it is expected that the quality of the connection is not good enough for the subscriber to continue the call.

Post Dial Delay (PDD)

On dialing phone number, either there is a ring or busy tone that tells us that whether the called party is available or not. The time elapsed between dialing a number and hearing a tone is referred to as Post Dial Delay (PDD). In case of higher PDD, it is expected that there is no dial tone for the subscriber to initiate a call.

Answer-Seize Ratio (ASR)

It is the ratio between the successful calls and the attempted calls that cannot be answered for any reason. In case of lower ASR, it is expected that the route provided to the call is choked-up for the subscribers to make phone calls.

LCR itself is a part/feature of a billing system that is responsible for identifying the appropriate route for the calls originated/terminated.

Billing system has the list of different rates assigned to various destinations. To route calls, these rates are compared and total cost of each route is calculated and eventually the optimum route is selected for every call. The billing engine should manage all the stated issues efficiently to ensure qualitative service to the subscribers.

AdvancedVoIP offers a billing solution such as AdvancedVoIP Billing Solution that is proficient enough to cater to all the stated issues to provide accurate billing.

Summary

QoS is a short term for **Quality of Service**. The success of any product/service is directly proportional to the quality it retains. Quality of any service depends on the traffic flow as well as the network of terminating partners.

A VoIP terminator is one who takes VoIP calls off internet and delivers them to PSTN phones. Therefore, selection of a terminating partner that can transmit your calls to their destinations with better quality is also vitally important.

Most commonly used protocols used by the termination network are H.323 and SIP. H.323 is extensively used in VoIP (Voice over IP), Video Conferencing and Data Communication over the Internet. SIP stands for Session Initiation Protocol, it allows establishment of different sessions that can be used for communication over the Internet.

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In order to route a call effectively, certain issues should be considered and improved accordingly such as Average Call Duration (ACD), Post Dial Delay (PDD) and Answer-Seize Ratio (ASR).

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Contact Information

In case of any ambiguity regarding the concept, explained in the whitepaper, please feel free to contact us at support@advancedvoip.com or please, visit http://www.advancedvoip.com/voip_contact.html

For further information please, visit www.advancedvoip.com

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