Quality of Converged Services

Abstract

To many businesses, the term Converged Services (CS) is associated with the notion of poor service quality. This perception is mostly due to unsatisfactory experiences with consumer-oriented services such as Skype or Vonage, or with traditional ISPs delivering voice as yet another IP application running on top of a network not designed to carry voice traffic. These services, as attractive as they may be given their low cost, are characterized by levels of quality not suitable for mission-critical business use.

This view of Converged Services is, however, misplaced. In fact, large enterprises and carriers have been using Converged networks for quite sometime to their complete satisfaction. Today. even companies with less than 100 employees can benefit from the cost-savings and increased flexibility afforded by CS by relying on vendors providing end-to-end Quality of Service (QoS) management. Adequate QoS management can in fact provide levels of Mean Opinion Score (MOS) that are as good as those measured on traditional phone systems.

Introduction

Historically businesses have purchased their voice and data services over separate networks. A Business paid monthly for a set amount of phone lines and also paid monthly for a set amount of data services. These resources were available all day and all night regardless of if they were being used or not. While this design has been in place for decades, it also potentially leaves resources unused and wasted. An average office uses less than half of their phone lines during the majority of the business day. This connectivity out of the building sits idle and unused while employees may struggle with an internet connection which is overloaded and hurting productivity.

Instead of two separate connections one larger connection is provisioned and capacity is shared between voice, video and data. The telecommunications industry is rapidly moving towards a system where voice, video and data are being placed over the same physical infrastructure. voice calls, rather than being placed on dedicated lines are being chopped into thousands of fragments called packets and sent in a continuous stream of data. This idea is not new, Voice over ATM (VoATM) and Voice over Frame Relay (VoFR) have been around years though never widely deployed. This has changed in recent years due to the rapid adoption of the Internet Protocol (IP) and technologies such as Session Initiation Protocol (SIP) which was defined in 1999. After years of testing and refinement VoIP utilizing SIP is the accepted the standard and is direction the industry is moving towards.

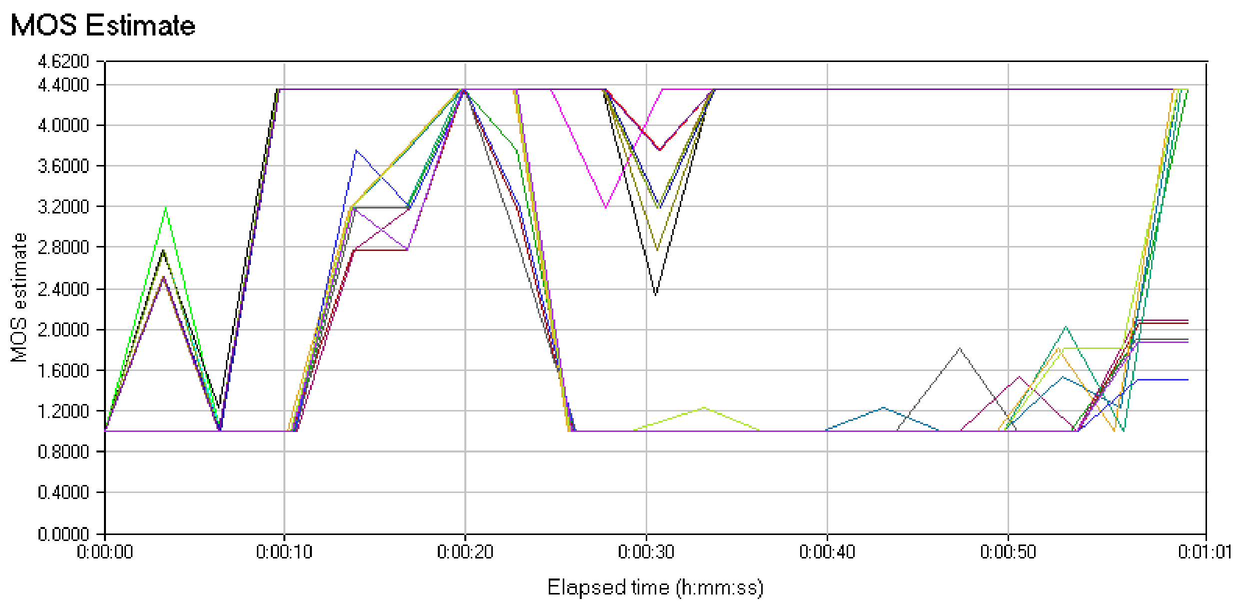
This converged design means that fewer resources are wasted and business are getting more for their telecommunications dollar. Not all communications are equal however, with voice, video, and data sharing the same connection there is the potential for overcrowding of the system. Without some mechanism to differentiate between these different types of services the network will drop a little of the data flows, a little of the voice calls and a little of the video to make room for everything. Data is resilient and will resend the lost bits or packets of information. Voice however cannot be retransmitted as by the time it has been noticed as missing the lost packets should have already been played to the end callers. This leads to an overall lower quality call and poor call satisfaction.

Measuring Quality

In the past telephone companies seeking to rate their service paid a large pool of people to listen to calls and rate them on a scale from one to five. The average of that score is called a Mean Opinion Score or MOS and is calculated today based on mathematical analysis characteristics observed during the call. A score of 4.0 or higher is considered high quality while a score of 3.6 and below is considered to be unsatisfactory and will leave the majority of users dissatisfied with the call quality.

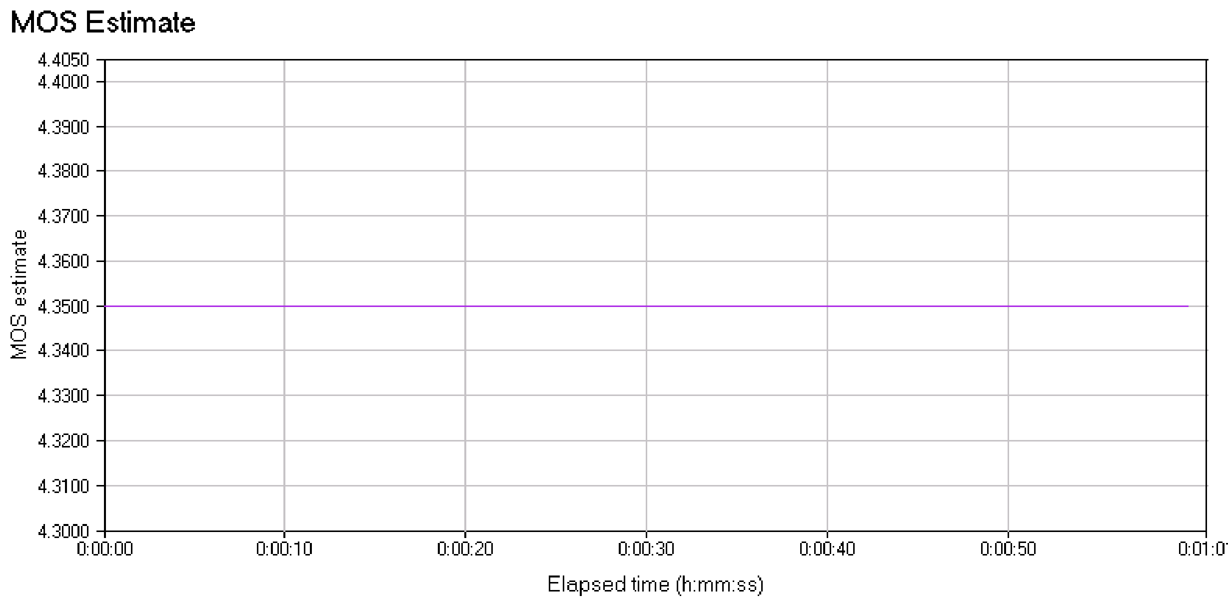
A mechanism called Quality of Service (QoS or Class of Service) already exists to provide this differentiation thus keeping the MOS above 4.0 and leaving all users very satisfied with their calling experience.

Below is an actual test result showing ten concurrent voice calls crossing over a T1 (1.544 Mbps) connection without any quality of service enabled.



As you can see the call quality ranges from outstanding quality of 3.5 to completely unacceptable at 1.0. This is happening because there is other data (ftp, email, web traffic) traveling along the same connection with the ten voice calls. Without QoS the networking equipment drops packets from both the data and the voice when there is congestion.

When we enable QoS on this same connection we tell the networking equipment to give preference to the voice calls and only drop the data packets if congestion occurs.



What happens is all ten calls experience outstanding quality of 4.35 with no degradation throughout the entirety of the test. The results appear as one solid line on the graph. Data continues to flow over the connection alongside the voice but only uses what is left

However not all networking equipment is capable of providing QoS, especially on a large scale with hundreds or even thousands of individual connections. This level of service is dependent on the hardware your provider uses and their technical ability to deploy it.