Marketing Systems Group

PRO-T-S

Centralized Dialers

When implementing a centralized dialing solution across a large geography, the primary concern will be reducing latency in the delivery of the various packets on the network. A network with infinite bandwidth would be ideal but is not realistic. So, various methods of packet compression and prioritizing are utilized to help minimize the delay in getting packets to where they need to be across the entire network infrastructure. Also, network hardware and software tools allow the monitoring and proactive reporting of the networks performance. MSG can work with your network facilitators to help identify the best solutions for your environment, but we do not provide these type of network tools.

The use of Multiprotocol Label Switching (MPLS) circuits are becoming a popular way to help facilitate the necessary quality of service (QOS) required for time sensitive packet traffic and provides an invaluable tool in providing true QOS across connected networks. The support for the tagging of packets from end-to-end in a network provides the best possible solution and is the only way most carriers will "guarantee" a certain service level for prioritized packet traffic. Utilizing the QOS tagging available in a MPLS connected network allows the best opportunity to reduce packet latency of every type.

The use of compression on the WAN (G.729 codec) for VOIP traffic is a popular way to try and save bandwidth on a WAN but should be used only when absolutely necessary. That is because there is a separate penalty of "on-the-fly" compression/decompression that can cause the delay of packet delivery which can be another source of pauses and audio artifacts on calls. Typically, the pauses that are the result for this type of issue are most noticeable at the start of a call. The use of G.711 everywhere possible simply provides the best possible call quality.

The use of hardware based IP phones instead of software based IP phones is also essential in maintaining good QOS and call quality. The software based IP phones are too subject to interruptions by the host CPU on which they run, so voice packets can be delayed at the host machine's discretion causing unwanted pauses and audio artifacts. The hardware based IP phones are not prone to this type of packet delay since their hardware is dedicated to the VOIP process. And as with any phone, a good quality headset with echo cancellation will help provide the best results when they must be used.