DLS INTERNET SERVICES

Network Performance Requirements

Internet Connection

DLS Hosted PBX Phones require an Internet connection to make and receive calls.

Bandwidth

Each call on DLS Hosted PBX phone requires anywhere from 28 to 88kbps of bandwidth depending on CODEC chosen by the Hosted PBX Administrator for both upload and download. DLS Requirements Analyzer helps you determine your actual bandwidth requirements based on your choice of CODEC (standard or high quality audio) and a number of concurrent telephone calls.

For Example: if you have a T-1 connection (1.5Mbps) you can make up to 16 concurrent calls at any given time at high quality (G.711). However in many cases your computers and DLS phones will share Internet bandwidth effectively reducing amount of bandwidth available for your telephone calls.

VoIP QoS Requirements

QoS (Quality of Service) is a major issue in VOIP implementations. The issue is how to guarantee that packet traffic for a voice or other media connection will not be delayed or dropped due interference from other lower priority traffic.

Things to consider are

- **Latency:** Delay for packet delivery
- **Jitter:** Variations in delay of packet delivery
- **Packet loss:** Too much traffic in the network causes the network to drop packets
- **Burstiness of Loss and Jitter:** Loss and Discards (due to jitter) tend to occur in bursts

For the end user, large delays are burdensome and can cause bad echoes. It's hard to have a working conversation with too large delays. You keep interrupting each other. Jitter causes strange sound effects, but can be handled to some degree with "jitter buffers" in the software. Packet loss causes interrupts. Some degree of packet loss won't be noticeable, but lots of packet loss will make sound lousy.

Latency

Callers usually notice roundtrip voice delays of 250ms or more. ITU-T G.114 recommends a maximum of a 150 ms one-way latency. Since this includes the entire voice path, part of which may be on the public Internet, your own network should have transit latencies of considerably less than 150 ms.

Most network SLAs specify maximum latency on their backbone however the total latency for a VOIP call may also include additional latency in the VOIP provider's and the user's local ISP networks.

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Packet Loss

VOIP is not tolerant of packet loss. Even 1% packet loss can "significantly degrade" a VOIP call using a G.711 codec and other more compressing CODECs can tolerate even less packet loss. The default G.729 codec requires packet loss far less than 1 percent to avoid audible errors. Ideally, there should be no packet loss for VoIP

Jitter

Jitter can be measured in several ways. There are jitter measurement calculations defined in:

- IETF RFC 3550 RTP: A Transport Protocol for Real-Time Applications
- IETF RFC 3611 RTP Control Protocol Extended Reports (RTCP XR)

But, equipment and network vendors often don't detail exactly how they are calculating the values they report for measured jitter. Most VOIP endpoint devices (e.g. VOIP phones and ATAs) have jitter buffers to compensate for network jitter. Jitter buffers (used to compensate for varying delay) further add to the end-to-end delay, and are usually only effective on delay variations less than 100 ms. Jitter must therefore be minimized.

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