



## Quality of Service issues when deploying Voice Over IP Networks

The public Internet, in its current form, is a resilient network designed to pass information end-to-end by a series of packets, each independently routed and carried across the network. Lost data packets are merely resent by the source to ensure reliable end-to-end data transfer. This results in variable transmission delay, increasing rapidly if the network is congested.

Two-way interactive communication for voice requires very short delays, with speech carried in a large number of small packets sent at frequent intervals.

Because of the transit time across a network, it is not possible to wait for data retransmission and lost or out of order packets have to be discarded.

Lost packets and transmission delays are typically due to Internet congestion, insufficient band-width and/or poor band-width utilisation (ie. QoS, codec selection).

### Internet QoS Issues

The current public Internet is a “best endeavours” network. During periods of congestion it may not be able to deliver on time, each and every packet in a session (or call). However, there are several issues which, if addressed appropriately will provide the user with a very high Quality of Service (QoS) on almost all occasions.

## Quality of Service End-to-End Operation

The issues which effect call quality are numerous. At the customer level consideration of several key factors can improve the call quality considerably.

### Band-width

Insufficient band-width will impact the quality of voice traffic. Jitter, or even complete packet loss, results from a variance in the time stamps that are placed on both the sending and receiving ends of the packet. This variance can be corrected by adding buffers to store the packets and “feed” them in a steady stream to the application.

As a general rule of thumb we would recommend that the customer have band-width equal to twice the anticipated load. Remember that whilst you pay for a high speed connection, there is no guarantee that your I.S.P. will supply you with the stated band-width.

### Compression

Compression and decompression (CODEC) of digital signals is a means of reducing the required bandwidth or transmission bit rate.

Each CODEC produces a marginally different speech quality. Selecting a high band-width codec may produce a superior speech quality but it will increase the chances of transmission problems during times of internet congestion.

The quality of transmitted speech is a subjective response of the listener. A common benchmark used to determine the quality of sound produced by specific CODECs is the mean opinion score (MOS). MOS is a subjective measure determined by a wide range of listeners testing the quality of a voice sample on a scale of 1 (bad) to 5 (excellent). The scores are averaged to provide the MOS for that sample. Telco-grade voice refers to MOS scores of approximately 4 or above.

The table below shows the relationship between CODECs and MOS scores.

Table – Compression Methods and MOS Scores

Compression Method	Bit Rate (Kbps)	Framing Size (ms)	MOS Score
G711	64	1.25	4.1
G729	8	10	3.92
G729 x 2 encodings	8	10	3.27
G729 x 3 encodings	8	10	2.68

### Packet Prioritisation

The existing public internet currently has no means of differentiating packets containing time critical “Voice” data from regular data traffic. However, at the LAN and WAN level the priority of data leaving a network can be controlled with QoS enabled hardware.

Quality of service enabled switches and routers prioritise voice packets in order to minimise packet loss and transport delay on networks that share the same connection for both voice and data. On a shared voice and data connection traffic prioritisation is critical to avoid lost or delayed packets and ensure high voice quality.