## Network Assessment

All IP trunks and telephone extensions connect to the system via the customers data network. It is therefore absolutely imperative that the customer network is assessed and reconfigured if necessary to meet the needs of VoIP traffic.

! WARNING: A Network Assessment is Mandatory

When installing IP phones on any IP Office system, it is assumed by Avaya that a network assessment has been performed. If a support issue is escalated to Avaya, Avaya may request to see the results of a recent network assessment and may refuse to provide support if a network assessment with satisfactory results has not been performed.

Current technology allows optimally configured networks to deliver VoIP services with voice quality that matches that of the public phone network. However, few networks are optimally configured and so care should be taken to assess the VoIP quality achievable within a customer network.
Not every network is able to carry voice transmissions. Some data networks have insufficient capacity for voice traffic or have data peaks that will occasionally impact voice traffic. In addition, the usual history of growing and developing a network by integrating products from many vendors makes it necessary to test all the network components for compatibility with VoIP traffic.

A network assessment should include a determination of the following:

A network audit to review existing equipment and evaluate its capabilities, including its ability to meet both current and planned voice and data needs.

A determination of network objectives, including the dominant traffic type, choice of technologies and setting voice quality objectives.

The assessment should leave you confident that the network will have the capacity for the foreseen data and voice traffic.

## Network Assessment Targets
The network assessment targets are:

Latency:
Less than 180ms for good quality. Less than 80ms for toll quality.
This is the measurement of packet transfer time in one direction. The range 80ms to 180ms is generally acceptable. Note that the different audio codecs used each impose a fixed delay caused by the codec conversion as follows:

G.711:
20ms.

G.722:
40ms.

G.729:
40ms.

Packet Loss:
Less than 3% for good quality. Less than 1% for toll quality.
Excessive packet loss will be audible as clipped words and may also cause call setup delays.

Jitter:
Less than 20ms.
Jitter is a measure of the variance in the time for different packets in the same call to reach their destination.
Excessive jitter will become audible as echo.

Duration:
Monitor statistics once every minute for a full week.
The network assessment must include normal hours of business operation.