Introduction

Packet voice systems accept "analog" voice signals from telephone handsets, digitize and compress the signal, placing the resulting series of bits into a short packet, send the packet over a network and then decode and reconstruct the signal at the remote end.

The packet network may be IP based, ATM based, Frame Relay based .... leading to a variety of "Voice over" technologies including VoIP, VoATM, VoDSL, VoCable, VoP etc. We will use the term VoIP to generally refer to all of these.

IP phones perform the digitization, compression and packetization process directly within the phone and send the resulting stream of packets over an Ethernet connection.

IP gateways interface to analog or digital phones or to TDM trunk systems and convert each voice signal to a VoIP packet stream. An IP PBX or Enterprise Gateway is often used by a company to interface IP Phones to the conventional telephone network. A Trunking Gateway is used within a phone company network to convert bulk telephone traffic to Voice over IP.

CODECs and Vocoders

The terms CODEC (literally Coder, Decoder) and Vocoder are used interchangeably to refer to the device within a VoIP phone or gateway that provides the digitization, compression and packetization process. There are a variety of different CODECs that are in use, for example:-

G.711 - which is the widely used PCM standard, encoding 8 bit speech samples at 8000 samples per second, giving 64 kilobits per second of digital speech data

G.729/ G.729A - which is an 8 kilobit per second encoding standard

CODECs, notably those that run at lower bit rates, can introduce some speech distortion.

Although the published MOS score for a low bit rate CODEC may be close to "toll quality" very low bit rate CODECs do cause audible distortion. Of more concern is that there is some speculation that CODECs such as G.729A *may* lead to stress when used continuously, for example in call center applications.

Call Quality

Call quality can be measured using subjective testing, intrusive testing or non-intrusive monitoring. Typically call quality is described in terms of MOS scores or R Factors.

Packet Loss

Packets may be lost during transmission, in which case there may be gaps or audible problems in the decoded voice signal. CODECs often incorporate Packet Loss Concealment, which helps to mask the effects of lost or discarded packets.

In depth discussion of packet loss distribution

Jitter and Jitter Buffers

The transit time of packets can vary considerably. This variation in delay is termed Jitter or Packet Delay Variation. A Voice over IP phone or gateway incorporates a Jitter Buffer that introduces a small amount of delay in order to smooth out these timing variations. If packets arrive too late then they may still be discarded - hence a jitter buffer translates jitter into additional delay and packet loss.

>> In depth discussion of jitter

Delay

If the transit delay for packets exceeds 100 milliseconds then users start to notice the delay. If delay exceeds 200 milliseconds then the users can experience conversational difficulties due to the breakdown in the usual conversational "protocol".

Predeployment Testing

Predeployment testing is used to verify that a network will be able to support Voice over IP.

Quick Links

- Call Quality
- Packet Loss
- Jitter
- Delay
- Predeployment testing