G.726

Band	Narowband
Sampling frequency	6kHz
Bitrate	40 kbit/s - 32 kbit/s
MOS (Codec)	4,2 - 3,85

G.726 is an **ADPCM**-based narrowband codec (50 to 4000 Hz) of the International Telecommunication Union (ITU-T) for compressing speech into digital telephone signals.

Technical Specifications

The method is based on Adaptive Differential Pulse Code Modulation (**ADPCM**).

The codec supports bit rates of 16, 24, 32 and 40 kbit/s.

G.726 achieves a **Mean Opinion Score (MOS)** of about 4.2 for the 40 kbit/s variant and about 3.85 for the 32 kbit/s variant.

Usableness

G.726 is also used for IP telephony.

With **DECT** telephones, the 32 kbit/s variant is used for narrowband telephony. The **DECT** standard is specially adapted to G.726-32, so a **DECT** radio channel can transmit exactly 32 kbps. The decision for **G.726** was also made because **ADPCM** is relatively insensitive to bit errors, which is of particular interest for radio applications. The 32 kbit/s variant also has the advantage that two interconnected channels produce 64 kbit/s, which makes it possible to transmit exactly one G.722 data stream (64 kbit/s) with two channels and thus also realize HD telephony with **ADPCM** via **DECT** .

The codec is also used for international telephone network connections of the fixed and mobile network infrastructure. The multiplexing method used is usually DCME (Digital Circuit Multiplication Equipment) implemented according to **G.763** and uses the **G.726** codecs with 16, 24, 32 and 40 kbit/s depending on the utilization of international voice traffic. These compressors are also used internationally in some access networks for the connection of private branch exchanges.

Data volume and delay times

For example, voice compression to 32 kbit/s in one minute produces a data volume of 240 kB; a one-hour **VoIP** call results in 14.4 MB of voice data. Not included here are the protocol data for communication in IP networks, which require up to 50% additional bandwidth depending on the number of data packet rates and the protocol. In circuit-switched networks, the protocol data are part of a separate signalling channel.

Delay times in IP networks depend on the transmission time (transmission delay), the necessary buffering with **Jitter** (Jitterbuffering), the number of interconnected nodes and their transmission rates (transmission delay, unless they are cut-through switches) as well as the encoding and decoding (packetization time) of the voice using the **G.726** codec used here with the corresponding packet rate. In circuit-switched networks there is only a delay due to transmission time, encoding and decoding.

Deployments

Other Hardware

SNOM supports both MIME-type **G726** and AAL2-G726.

Some implementations, such as old versions of AVM Fritz!box, enabled the user to set the bit order himself. With the checkbox "Provider supports G726 according to RFC 3551" the bit sequence could be switched by the user. The background for this is that AVM used the MIME-type G726 instead of AAL2-G726 for Big Endian for a very long time and therefore a transitional regulation towards standards-compliant behaviour was necessary.

Source: https://de.wikipedia.org/wiki/G.726

Related Links:

- ADPCM - Adaptive Differential Pulse Code Modulation
- Codec

- Do snom phones support packet loss concealment
- G.711
- G.722
- G.726
- G.729
- GSM - Global System for Mobile Communications
- How can I obtain a license for using the AMR Wideband Codec
- Jitter
- MOS - Mean Opinion Score
- Opus
- VoIP - Codecs