GRANDSTREAM NETWORKS
FAQ – CODEC

Q1: How much overhead does Ethernet add to RTP packets?

For voice over IP over Ethenet, an RTP packet contains 54 bytes (or 432 bits) header. These 54 bytes consist of 14 bytes Ethernet header, 20 bytes IP header, 8 bytes UDP header and 12 bytes RTP header.

Q2: What types of voice codec do Grandstream VoIP products support?

Each codec has its uniqueness for certain application. Grandstream BudgeTone series VoIP phones and HandyTone series analog telephone adaptors support G.711-uLaw, G.711-aLaw, G.722, G.723, G.726, G.728 and G.729 and iLBC. The wideband codec G.722 is only supported by BudgeTone VoIP phones. It has the same bit rate as G.711 but with twice sampling rate (16KHz vs. 8KHz) and better sound effect.

Q3: What is the frame rate and bit rate for each codec?

G.711 has 10ms frame length with 64kbps bit rate;
G.722 has 10ms frame length with 64kbps bit rate;
G.726-32 (also referred as G.721) has 10ms frame length with 32kbps bit rate;
G.728 has 2.5ms frame length with 16kbps bit rate;
G.729 has 10ms frame length with 10kbps bit rate;
G.723 has 30ms frame length with either 5.3kbps or 6.4kbps bit rate.
iLBC has 20ms or 30ms frame length with 15.2kbps or 13.3kbps bit rate;

Q4: What is the total bit rate(including Ethernet and IP header and UDP and RTP over head) or bandwidth for each codec?

G.711 -- 107.2 kbps bit rate;
G.722 -- 107.2 kbps bit rate;
G.726-32 (also referred as G.721) -- 75.2 kbps bit rate;
G.728 -- 188.8 kbps bit rate;
G.729 -- 53.2 kbps bit rate;
G.723 -- either 19.7 kbps (for 5.3 frame bit rate) or 20.8 (for 6.4kbps frame bit rate).
iLBC – either 36.8 Kbps (for 20ms frame length) or 27.7 Kbps (for 30ms frame length
NOTE: Above are based on Q3 frame interval.

Q5: What is "Voice_Frames_Per_TX" and how does it relate to Ethernet traffics?

To reduce the overall Ethernet/IP/RTP overhead introduced by the 54 bytes header, multiple voice frames can be packed into single Ethernet frame to transmit. Of course, this would increase the voice delay. In case that network bandwidth is constrained, increasing this count may improve the overall voice quality.

If RTP packets are sent every 2.5ms (for G.728), the total Ethernet/IP/RTP overhead is 0.432*400 = 172.8kbps. This won't work well over public Internet. However, if RTP packets are sent every 10ms, the total Ethernet/IP/RTP overhead is down to

0.432*100=43.2kbps. If RTP packets are sent every 20ms, the total Ethernet overhead may be further down to 0.432*50=21.6kbps.

Grandstream suggests 30ms packet rate for G.723, 10ms for G.728 and 20ms for the rest codecs. Voice_Frames_Per_TX is then set to 1 for G.723, 4 for G.728 and 2 for the rest.

Q6: Why is G.723 the best option for narrow bandwidth IP communication?
For G.723, the frame rate is at 30ms and the codec bit rate is 5.3kbps (20 bytes per 30ms) or 6.4kbps (24 bytes per 30ms). The total bit rate are 5.3 + 0.432*33.3 = 19.7kbps, or 6.4 + 0.432*33.3 = 20.8kbps. This low bit rate is ideal to transmit over 28.8kbps dialup modem connection. With other technologies like data link layer compression, silence suppression and comfortable noise generation, the total bandwidth could be even further lowered.

Q7: What codec should I select?
Generally speaking, all codecs provide good voice quality. However, lower bit rate codec may have poor quality for music. DTMF tones and fax signals on audio channel may not be decoded on remote premises. If the bandwidth allows, G.711 is the default option, G.722 give even better sound quality.