VOICE OVER IP AND JITTER AVOIDANCE ON LOW SPEED LINKS

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Abstract: This paper explains the VoIP network design for good voice quality on low speed links and describes the quality measurement results. Many of issues, such as compression of the speech frame and delay variation, are inherent to VoIP. With careful planning and solid network design these effects on VoIP networks can be minimized.

1 Introduction

With the aim of reducing communication costs, efforts of integrating voice and data networks have been a rising priority for many companies. Organizations have been working on the solutions which would make them use the excess capacity on broadband networks for voice and data transmission, as well as utilize the Internet and company Intranets as alternatives to expensive systems. At the same time, more and more companies are seeing the value of transporting voice over IP networks to reduce telephone and facsimile costs and to set the stage for advanced multimedia applications. Providing high quality telephony over IP networks is one of the key steps in the convergence of voice, fax, video, and data communications services.

VoIP delay or latency is characterized as the amount of time it takes for speech to exit the speaker and reach the listener. The ITU-T recommendation G.114 specifies that for good voice quality, no more than 150 ms of one-way, end-to-end delay should occur. In an unmanaged, congested network, queing delay can add up to two seconds of delay. This lenghty period of delay is unacceptable in almost any voice network.

On the other side is no less important the type of using voice coding, decoding. Codecs are developed and tuned based on subjective measurements of voice quality. Standard objective quality measurements, such as total harmonic distortion and signal-to-noise ratios, do not correlate well to a human's perception of voice quality, which in the end is usually goal of most voice compression techniques. A common subjective benchmark for quantifying the performance of the speech codec is the *Mean Opinion Score (MOS)*. MOS tests are given to a group of listeners. Although MOS scoring is a subjective method of determining voice quality, it is not the only method for doing so. The ITU-T put forth recommendation P.861, which covers ways objectively determining voice quality using *Perceptual Speech Quality Measurements (PSQM)*. [2]

Compression technique relation and MOS speech quality parameter is presented in Figure 1.

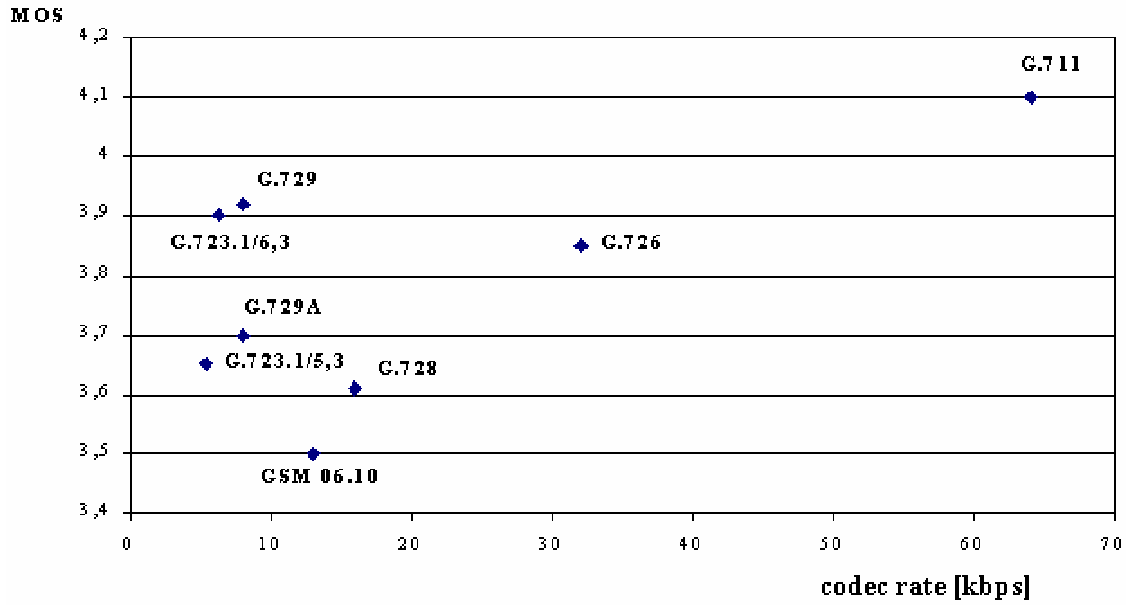

Figure 1. : Compression Technique and MOS.

2 Low Latency Queuing

Queuing delay is only one component of end-to-end delay. A packet-based networks generate various delay for various reasons. LLQ is a feature that provides a strict Priority Queue (PQ) to Class-Based Weighted Fair Queuing (CBWFQ). LLQ enables a single strict PQ within CBWFQ at the class level. With LLQ, delay-sensitive data (in the PQ) is dequeued and sent first. In a VoIP with LLQ implementation, voice traffic is placed in the strict PQ. The PQ is policed to ensure that the fair queues are not starved of bandwidth. When you configure the PQ, you specify the maximum amount of bandwidth available to the PQ. When the interface is congested, the PQ is serviced until the load reaches the configured bandwith value in the priority statement.

Voice traffic packet is recognized by:

- Access-lists for UDP port range, hosts addresses,
- IP header ToS fields, IP Precedence, DSCP, and more
- IP RTP port range
- IP ToS (Type of Service) Fields: DCSP and/or IP Precedence
- Protocols and Input Interfaces
- All valid match criteria used in CBWFQ

3 Link Fragmentation and Interleaving (LFI)

While 1500 bytes is a common size for data packets, a typical VoIP packet (carrying G.729 voice frames) can be around 66 bytes (20 bytes voice payload, 12 bytes RTP, 8 bytes UDP header, 20 bytes IP header and 6 bytes layer-L2 PPP header). [1]

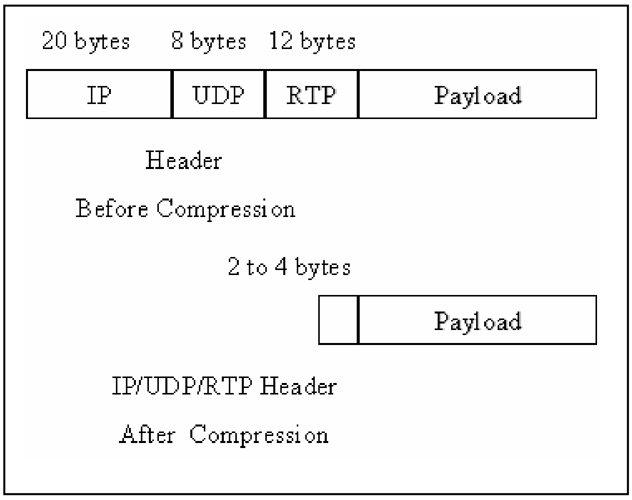

Figure 2. : RTP and cRTP packet.

If a voice packet is ready to be serialized just when a data 1500 bytes packet starts being transmitted over the 64 kbps link, then there is a significant delay. The delay-sensitive voice packet will have to wait 187 msec before being transmitted.

The large data packets can adversely delay delivery of small voice packets, reducing speech quality. Fragmenting these large data packets into smaller ones and interleaving voice packets among the fragments reduces jitter and delay. LFI feature helps satisfy the real-time delivery requirements of VoIP. [1], [3]

4 Experiment on the Low Speed Link

The most used codec G.729 is used in experiment, fragment size in configuration has the recommended value 20 ms. The link between routers is realized by two synchronous modems on 64 kbps. The routers interconnect 100 Mbps LAN and on the both sides are connected computers, which enable data transfer, by using FTP protocol is reached the traffic saturation on the 64 kbps link. The situation is presented in the figure 3.

Configuration , router Cisco 1751:

```
interface Multilink1
 bandwidth 64
 ppp multilink
 ppp multilink fragment-delay 20        /* 20 ms is max. transmission time
 ppp multilink interleave               /* switch packet interleaving is ON
 multilink-group 1
 ip rtp priority 16384 16383 48         /* RTP priority is based on the range UDP
                                        destination port 16384 – (16384+16383), 48 kbps
                                        is maximum allowed bandwith in priority queue
```

dial-peer voice 113 voip

destination-pattern 42069699....	/* dialing pattern
session target ras	/* RAS signaling to Gatekeeper
no vad	/* switch Voice Activity Detection is OFF

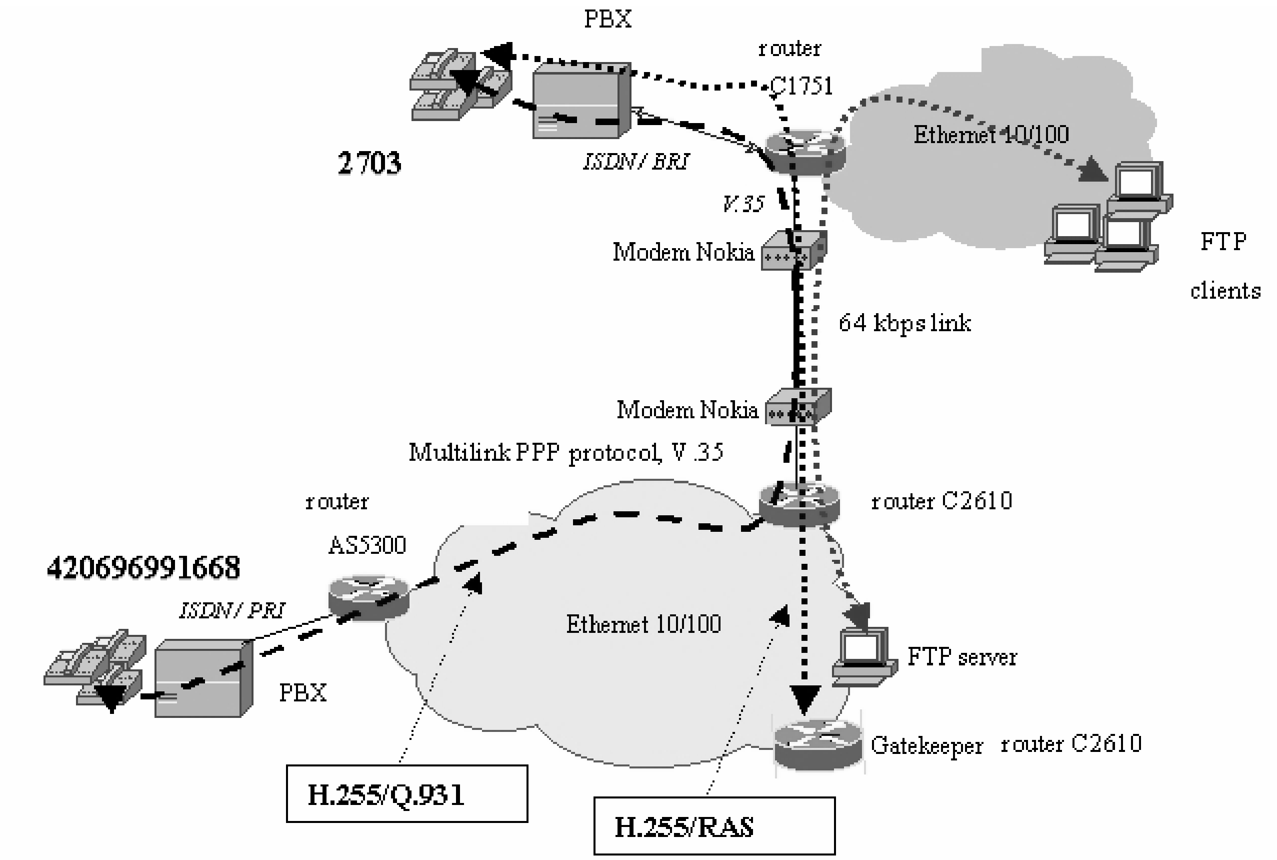

Figure 2. : RTP and cRTP packet.

5 Measurement on the Low Speed Link

Measurement device:

LAN Analyzer *Performer* (RADCOM)
Ethereal (Open source SW, http://www.ethereal.com)

Call setup:

Codec:	G.729 / 8 kbps	
Call setup attempts:	2 connection, successfull	OK
Setup time (SETUP – RINGING) :	743 ms	OK
Listening Quality (MOS):	3,72	OK

Delay variation (jitter):

Simply stated, *jitter* is the variation of packet interarrival time. Jitter is one issue that exists in packet-based networks. While in a packet voice environment, the sender is expected to reliably transmit voice packets at a regular interval. These voice packets can be delayed throughout the packet network and not arrive at that same regular interval at the receiving station. The difference between when the packet is expected and when it is actually received is *jitter*. It is cause, why is necessary *jitter bufer*. The jitter buffer is considered a dynamic queue.

Jitter buffer value in experiment: 60 ms

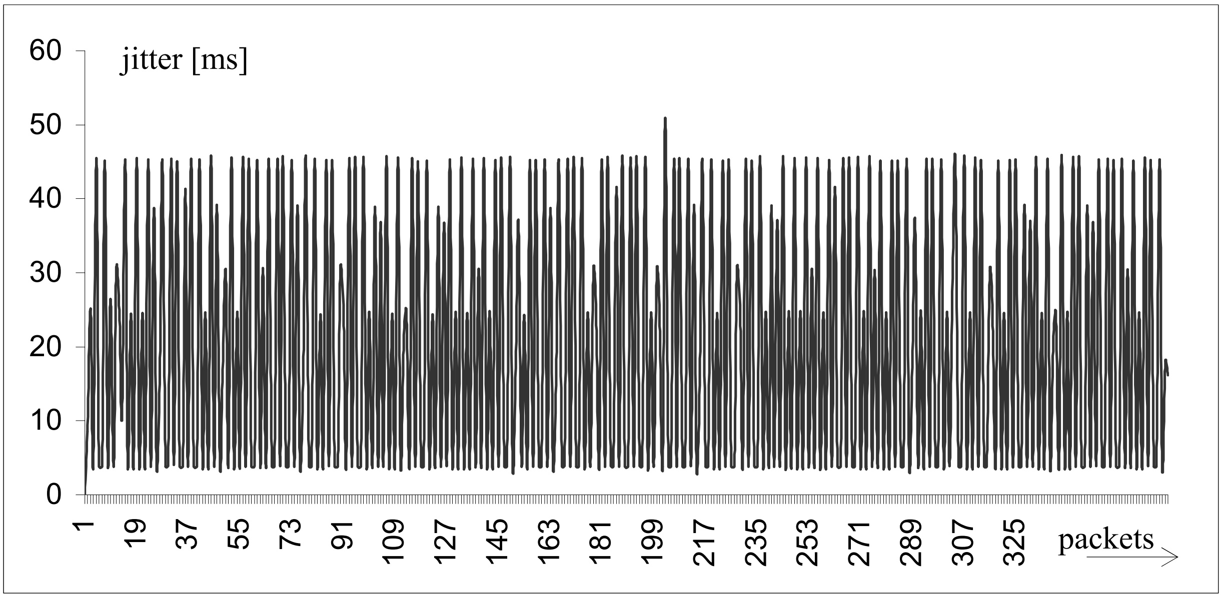

Figure 3. : Measured jitter of the voice packets.

Advantages:

- voice traffic is matched and directed to the strict priority queuing PQ
- good voice quality on the low speed link

6 Conclusion

Many skeptics do not believe IP can give the proper QoS for such a real-time application, but with the proper network design and the right tools, it is possible. Each network is different and requires not only attention to detail but also a knowledgeable administrator who knows how to tune the network to provide optimal QoS. QoS can help solve some of these problems, namely packet loss, jitter and queuing delay. Some of problems QoS cannot solve are propagation delay, codec delay and sampling delay.

References

[1] Schulzrinne H., Casner S., Frederick R., Jacobson V. , *RTP: A transport protocol for real-time aplication,* RFC 1889, 1996

[2] Peters J., Davidson J. , *Voice over IP Fundamentals,* Cisco Press, Indianopolis USA, 2000, ISBN 1-57870-168-6

[3] Voznak M.:The Voice Over IP Technology,COFAX,Bratislava1999,ISBN 80-233-0429-1.