VoIP QoS on low speed links

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Abstract: Quality of Service providing in Internet-like computer networks has been tested for low-speed links and VoIP service in the operational CARNet network. QoS techniques, traffic selection and prioritization were deployed and tested for three voice codecs G.711, G.728, and G.723.1, and for four low-speed links of 56, 64 and 128 kbps and 2 Mbps. VoIP QoS in presence of TCP and UDP traffic was acceptable at 2 Mbps only. With VoIP traffic only, low speed codecs performed well on 56-128 kbps links. G.723.1 at 5.3 kbps has shown very good results. Main reason for lower VoIP QoS was IP packet delay variation. Results of Mean Opinion Score and subjective quality estimated through E-model from packet QoS parameters were comparable. This approach can be used as a model for any service quality testing.

1. INTRODUCTION

Voice over IP has become common part of the integration process of communication and data services, using the Internet as a technical basis for the deployment. Internet-based networks do not initially provide quality of service, or care for specific services characteristics, which is not enough for commercialized usage.

Two approaches for quality assurance could be taken - implementing QoS mechanisms in the network core and at the edge of the network. Quality assurance in the network core can be obtained by reserving the needed bandwidth. At the edge of the network, QoS is assured by prioritizing one class of traffic over the others. Traffic classification is done in both cases, either by traffic type (regarding service characteristics or quality parameters), user (source and/or destination), or service (port numbers). Priority is enforced through queuing, whereas knowing traffic behavior as well as parameter tuning can significantly improve the overall service quality.

1.1 VoIP network elements

Passing from the source to the destination, voice goes through three network elements: end-user device (speaker and listener), edge network devices and telecommunication channels, Figure 1.

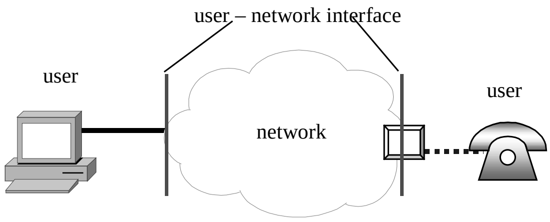

Figure 1 - VoIP network elements

In VoIP, user equipment handles voice coding and decoding and traffic from multiple applications that are running in parallel. Codecs with low processor and memory requirements might be of interest.

At network edge traffic scheduling and forwarding take place, and can sometimes include coding/decoding. In both cases, the amount of extra delay that is introduced to a certain service is of concern, either through (de)coding that implies that less requiring codec should be used, or buffering which is then impacted by applied queuing technique and packets' size. Playback buffer size must be optimized regarding jitter and interactivity.

In edge enforced QoS, network core should manage as much traffic as possible, switching it as fast as possible to the destination. It should not be concerned with the type of service, nor the prioritization. In the case of heavy traffic, the only solution is to increase the bandwidth.

1.2 Previous work

VoIP deployment was accompanied with activity to discover the impact of the current best effort Internet to the QoS of voice flows, to understand the mechanisms involved and to possibly develop best strategies for QoS assurance. Measurement activities were mainly performed on simulation testbeds, while measurements on live networks were less often because of their complexity and cost. However, in early phase real network measurements resulted in ITU-T E-model [1] suitable to evaluate subjective QoS scoring from voice flow packet traces. Using E-model, expensive MOS measurements could be avoided.

In later phase, lot of live measurements was done in the field of backbone QoS assurance. The mechanisms of real network in the presence of sufficient link capacity were investigated. Numerous algorithms in the fields of coding and speech activity detection, influence and concealment of packet losses, and to eliminate impact of delay variation were proposed. Useful review is given in [2].

However, impact of access networks, often characterized with slow links, seem to be somewhat neglected. In this paper, the results of the real network measurements on low capacity links are presented. The paper is organized as follows: in Chapter two the QoS framework is described. Chapter three deals with testing environment. Results are presented in Chapter four, and conclusions in Chapter five.

2. QoS FOR VoIP OVER LOW-SPEED LINKS

In this paper, the term "low-speed link" is defined in regard to existing connections in CARNet network. The weakest points in CARNet network at the moment are links for modem access: 56 kbps or one or two channel ISDN, i.e. 64 kbps, and 128 kbps. Next slowest speed is 2 Mbps. Those speeds were therefore considered in this paper.

Quality of service in packet data networks depends upon available bandwidth, packet delay from source to destination, packet delay variation also called jitter, and packet loss rate during transmission. Depending upon the service, some of those parameters can be more or less important, while depending upon the path that a packet takes, some parameters can be more or less emphasized.

Parameter values whose compliance assures voice communication quality are given in Table 1, according to [3] for delay, and [4] for jitter and packet loss.

Packet delay	Packet delay variation (jitter)	Packet loss
≤ 150 ms	≤ 25 ms	$\leq 10^{-4}$

Table 1 - VoIP quality assurance parameter values

Quality of service can be influenced using different coding algorithms, since it determines packet size and required speed. Three ITU-T codecs have been used in measurements - G.711 at 64 kbps, G.728 at 16 kbps and G.723.1 at 5,3 kbps. Codec characteristic comparison is given in [5].

In every point in the network, packet scheduling and the order at which packets are served has direct effect on packet delay. Queuing algorithms choose which packets will be served first and which will be dropped. Four techniques have been observed - First In First Out (FIFO), Weighted Fair Queuing (WFQ), Weighted Random Early Detect (WRED) and Low Latency Queuing (LLQ), also called Classed Based Weighted Fair Queuing with Priority Queuing (CBWFQ PQ).

3. TESTING ENVIRONMENT

Measurement scenario for each queue-codec-speed combination is shown on Figure 2. During the first 20 seconds, VoIP traffic is the only introduced traffic on the link. After 20 seconds TCP traffic starts in addition to VoIP, and UDP traffic is added during the last 20 seconds. Although VoIP also uses UDP on transport layer, "UDP traffic" here relates to any other UDP-using traffic that was not VoIP. Other traffic observed could not have been eliminated, but could have been influenced on by different QoS mechanisms.

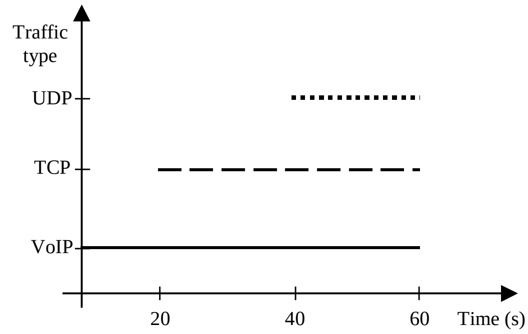

Figure 2 - Observed traffic

TCP and additional UDP traffic were generated using IP Traffic packet generator [6], and collected and analyzed using Ethereal [7]. Generated additional traffic had data portion of 1460 octets in both TCP and UDP packets.

Testing was performed between two locations in the operating CARNet network - the CARNet regional center Split and University of Split, as shown on Figure 3.

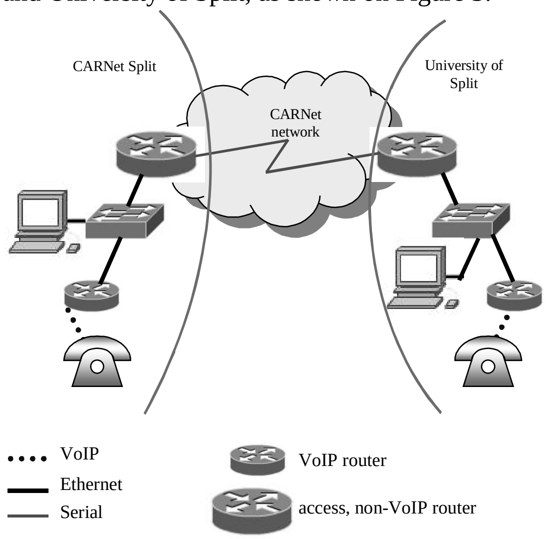

Figure 3 - Network topology used in measurement

As VoIP routers, brand name routers with voice modules were used. Access routers did not have VoIP support, and were not available for configuration changes, which corresponds to a situation where QoS can be influenced only at customer side of the network. Computers on the Figure 3 were used as source and destination devices for TCP and UDP traffic generated by IP Traffic.

Configuration is based on recommendations given in [8], measurement techniques are in conformance with [9], and results are calculated as in [10].

The following limitations were set for the experiment:

- only one VoIP conversation during the measurement,
- no codec changing during the measurement,
- quality was observed from the recipient's point of view, so packets were analyzed at the destination,
- tested codecs are available in all voice-coding devices,
- for queuing other than FIFO, VoIP traffic was marked with DSCP EF (IP precedence 5) and given priority over other TCP and UDP traffic,
- for all other parameters, default values were used.

This way, quality of service was observed for one voice conversation between two points that also had traffic other than VoIP, both adaptable (TCP) and inadaptable (UDP), with available bandwidth, codecs and queuing techniques as variable and varied values.

3.1 Subjective quality perception: MOS and E-model

Subjective quality was tested through modified MOS (Mean Opinion Score) measurements for the first phase when VoIP traffic was the only introduced traffic. Two participants of the conversation gave subjective judgment and mean score was taken as a mark for each case. Subjective testing for second and third phase was skipped because of weak QoS achievements.

According to E-model [1], corresponding scores can be obtained from packet characteristic. Packet delay was used as the only varying parameter and the values obtained through E-model R parameter were compared with MOS values.

3.2 Relevance of trace statistics

Single measurement was performed per scenario, each one in three phases of 20-second duration. The flow parameters for three codecs are presented in Table 2.

codec	band-width [kbps]	payload [octets]	packet size [octets]	time-stamp [ms]	no of pkts in 20 sec
G.711	64	160	200	20	1000
G.723.1	5,3	20	60	30	667
G.728	16	40	80	20	1000

Table 2 – Flow characteristics for selected codecs

Delay, jitter and packet loss calculations are based on significant number of packets per trace.

4. RESULTS

There are several aspects in which results have been analyzed and conclusions were searched for:

1. the impact that VoIP traffic and applied QoS mechanisms can have on other (TCP and UDP) users' traffic and vice versa,
2. finding the lowest speed that can provide satisfying quality for all three packet traces,
3. choosing the best codec,
4. finding a queuing algorithm with best results,
5. determining which of the QoS parameters was near to the limit of acceptance and thus the most important to take care of,
6. comparing results of MOS and E-model with main reasons for poor VoIP service quality.

4.1 TCP and UDP behavior in presence of VoIP

With enough available bandwidth, 2 Mbps in this case, there is no need for transmission dynamics changes for any of the three observed services traffic, Figure 4.

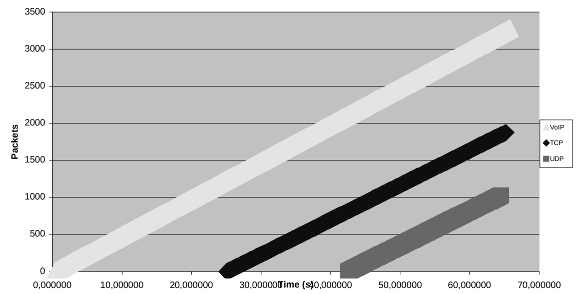

Figure 4 - Traces with available bandwidth

On slower speed links, e.g. 56 kbps, the nature of TCP and UDP traffic has shown on the first glance - TCP traffic with the feedback from the network has slowed back, Figure 5, or even stopped sending, Figure 6, leaving the channel to UDP traffic.

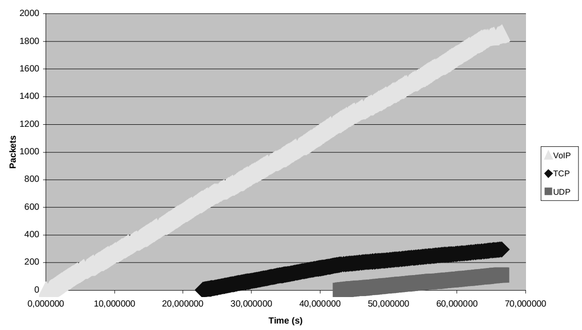

Figure 5 - Adaptability of TCP flow

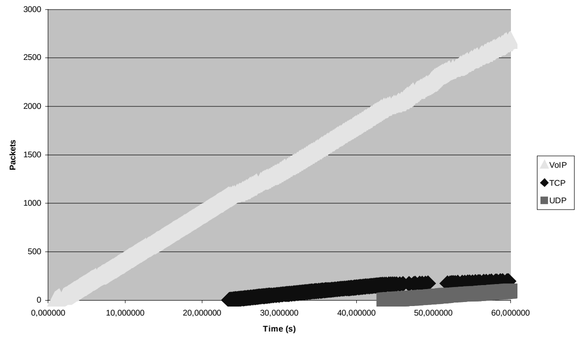

Figure 6 - Retreat of TCP flow

Therefore, it is obvious that forcing persistent traffic (e.g. VoIP) can lead to denial of other services.

4.2 VoIP QoS with additional traffic

For all observed codecs and queuing algorithms, three types of traffic coexisted successfully on 2 Mbps only.

On speeds lower than 2 Mbps, packet delay variation was the main non complying parameter, as shown for phase 2 in Tables 3 and 4. Results for phase 3 are similar. Jitter results in packet losses at receiver, where late packets are discarded.

JITTER	2 Mbps			128 kbps		
	711	**723**	**728**	**711**	**723**	**728**
FIFO	0,77	3,10	0,53	36,70	40,35	31,04
WRED	0,46	4,41	0,58	28,57	40,97	30,94
WFQ	0,23	3,41	0,45	28,47	48,34	31,07
LLQ	0,34	2,82	0,37	37,86	41,55	31,20

Table 3 – Jitter (ms) for 2nd phase at 2 Mbps and 128 kbps

JITTER	64 kbps			56 kbps		
	711	**723**	**728**	**711**	**723**	**728**
FIFO	82,70	53,64	34,52	89,51	53,09	33,36
WRED	41,83	51,37	34,76	54,58	52,25	34,99
WFQ	47,27	53,64	34,84	49,84	53,21	34,86
LLQ	70,62	23,70	37,54	83,53	54,12	49,45

Table 4 – Jitter (ms) for 2nd phase at 64 kbps and 56 kbps

In all measurements, jitter was most frequently exceeding the allowed limit of 25 ms. The reason for this is that during the transmission of already accepted lengthy TCP or UDP packet, VoIP packet has to wait until resources are available again, even if it is prioritized over all other traffic. Priority is important only at the moment when choice which packet should be forwarded is done.

Therefore, it is necessary to provide mechanisms that can interrupt transmission of low priority packet (if long) and serve another with higher priority, and then possibly continue with transmission of the interrupted one. During the testing, techniques like LFI (Link Fragment Interleaving) were not available.

When sufficient bandwidth is available, transmission time of data packet is not significant, and it is not necessary to perform additional processing through traffic selection and prioritization since it can cause additional delay.

4.3 Codecs comparison

G.711 requires 64 kbps and as such cannot be a choice for links slower than 128 kbps because of packet headers. If only VoIP traffic is used, 128 kbps is good choice. 2 Mbps is needed if additional traffic is to be used.

G.728 has shown good performance, but the subjective perception was much worse than for other two codecs. The silence suppression sounded unnatural.

G.723.1 with small packet sizes, despite the relative big overhead of VoIP header regarding the whole G.723.1 VoIP packet, has shown very good results in all measurements. It is, therefore, recommended codec for any future usage.

4.4 Queuing

WFQ has shown the best results at 2 Mbps and 128 kbps. At 2 Mbps, LLQ was also correct, while some packet loss was observed with FIFO and WRED. For lower speeds and with additional traffic, regardless of applied queuing algorithm, satisfying QoS was not obtained.

4.5 Voice quality

MOS results for 2 Mbps and 128 kbps are given in Table 4, and for 64 and 56 kbps in Table 5. E-model results (not shown) are in accordance with MOS scores.

MOS	2 Mbps			128 kbps		
	711	**723**	**728**	**711**	**723**	**728**
FIFO	5	4,82	4,825	4,95	4,9	4,92
WRED	5	5	4,9	4,75	4,78	4,9
WFQ	4,95	4,9	4,78	4,65	4,9	4,85
LLQ	4,92	5	5	4,9	4,8	5

Table 4 - MOS for 1st phase at 2 Mbps and 128 kbps

MOS	64 kbps			56 kbps		
	711	**723**	**728**	**711**	**723**	**728**
FIFO	1,85	4,85	4,88	1,08	4,9	4,85
WRED	1,62	4,95	4,72	1,55	4,9	4,85
WFQ	1,62	4,85	4,85	1,1	4,8	4,78
LLQ	1	4,72	4,72	1	4,9	4,88

Table 5 - MOS for 1st phase at 64 kbps and 56 kbps

In the first phase, packet delay variation for all measurements is shown in Table 6 for 2 Mbps and 128 kbps and Table 7 for 64 and 56 kbps. It can be seen that G.711 codec used at 64 kbps and 56 kbps does not satisfy jitter requirement of less then 25 ms, regardless of queuing algorithm applied. For all other combinations for this first phase of sole VoIP traffic, QoS requirements were satisfied.

Jitter (ms)	2 Mbps			128 kbps		
	711	**723**	**728**	**711**	**723**	**728**
FIFO	0,93	0,71	0,04	0,32	0,68	0,30
WRED	0,05	0,53	0,04	0,24	0,38	0,06
WFQ	0,05	0,78	0,46	0,32	0,49	0,21
LLQ	0,23	0,47	0,03	0,64	0,54	0,15

Table 6 - Jitter for 1st phase at 2 Mbps and 128 kbps

Jitter (ms)	64 kbps			56 kbps		
	711	**723**	**728**	**711**	**723**	**728**
FIFO	39,38	0,56	0,11	44,96	0,67	0,04
WRED	43,05	0,36	0,03	45,51	0,62	0,05
WFQ	42,58	0,50	0,03	45,22	0,66	0,03
LLQ	124,09	0,51	0,04	145,98	0,37	0,03

Table 7 - Jitter for 1st phase at 64 kbps and 56 kbps

In the graph at Figure 7, for all measurements, a dot presents a pair of MOS-jitter values

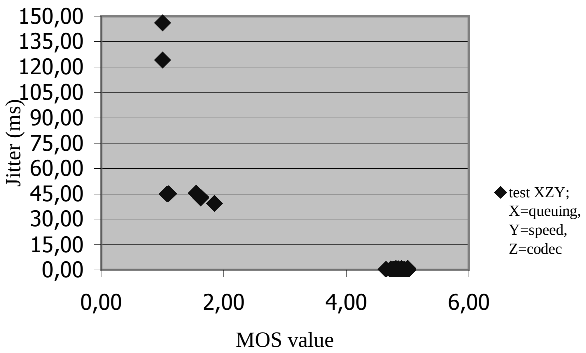

Figure 7 - MOS-Jitter correlation

The correlation of jitter and MOS is strong, in the manner that the consequence of high jitter are low MOS scores.

5. CONCLUSIONS

In this paper, a set of measurement results is presented in order to examine the VoIP traffic behavior when confronted with other traffic on relatively low speed links. All measurements were performed for one VoIP conversation between two points. As additional traffic, one TCP flow was present in second, and one TCP and UDP flows each were present in the third phase. Jitter and MOS results are given for the first phase of sole VoIP traffic, and jitter for the second phase.

On 2 Mbps three traffic traces successfully coexisted in all three phases, for all codecs and queuing algorithms. Such scenario of limited number of flows is recognized as SOHO implementation.

For lower speeds, VoIP traffic required prioritization over other traffic with lower speed codecs. Otherwise, VoIP should have been the only traffic. Subsequently, VoIP coded with G.711 could not have been used on 64 kbps and 56 kbps because of header overhead.

WFQ can be a good choice for traffic selection and scheduling, for all observed speeds, regardless of applied codecs.

Among tested codecs, G.723.1 codec with low rate of 5,3 kbps has shown very good quality and is highly recommended for further usage.

The main reason of QoS degradation is jitter caused by lengthy TCP packets and their significant transmission time on low speed links. Usage of techniques that allow long packet interruption and restoration for higher priority traffic is unavoidable.

The approach taken here can be extended to any other multimedia or real-time service.

Possible future work includes observing several VoIP flows simultaneously, using equipment from different vendors, examining other real-time services (i.e. videoconferencing) and network media (e.g. wireless), and/or extending QoS testing to IPv6 platform.

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ACKNOWLEDGEMENTS

All measurements described in this paper are done as a project of Croatian Academic and Research Network. First author would like to thank Nevenko Bartolinčić for his useful comments and suggestions.