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Performance Analysis of Packet-based Voice Networks

Petru Serafin[1]

Abstract – The article presents some investigations about performance for transporting compressed voice over packet-based networks. The scope is to determine optimal values for major parameters of the voice networks (packet size and dejittering delay) to meet the requirements of providing an acceptable Quality of Service (QoS). It is demonstrated by analyzing the behavior of two identical compressed voice networks that it is possible for voice over packet (IP or ATM) to meet QoS requirements if the networks are properly managed and provided with optimal parameters. The analysis is performed for Voice over IP networks with reference to Voice over ATM networks.

Keywords: QoS, VoIP, VoATM, M2E delay, codec delay, dejittering delay

I. INTRODUCTION

The purpose of this work is to investigate the performance of real-time transport of voice over packet to determine its optimal configuration to meet the quality limitations as they were presented in a previous paper [4]. The two types of voice over packet networks that are being analyzed are: Voice over IP (VoIP) and Voice over ATM (VoATM).

VoIP has the main issue to demonstrate that its Quality of Service (QoS) can be obtained and guaranteed without over-dimensioning the network. VoATM with AAL-1 (ATM Adaptation Layer-1) is limited by the standards to 64 kb/s circuit emulation.

The mathematical model used in this paper is only valid for VoIP and VoATM using AAL-1. The performance analysis of the packet-based voice network will therefore be presented in relation to VoIP and a comparison with VoATM using AAL-1 will be presented for information.

For the reason to simplify calculation we will assume some working hypothesis that are restricting the area of use of the mathematical model determined: we suppose no packet loss inside the network, we neglect any other delays than codec delays and the dejittering delays, all codecs that operate on the network are supposed to be of the same type. We consider all codecs of the same type, this means that the codecs operate at the same bit rate and they may all use or all do not use the Voice Activity Detection (detailed in the following section of the paper).

With these assumptions we can define the configuration for the packet-based network that will be used for performance analysis.

The stream of bits produced by the voice encoder is grouped in packets (all packets are considered of the same size, variable in VoIP or fixed in VoATM). The voice packets are then transported from source to destination over the packet-based network. Because all the packets in the network are transporting voice information they are very sensitive to delays and there must not be any priority mechanism in the network.

The packet-based network with the configuration such described above has the following remaining degrees of freedom: size of voice packets (valid for VoIP network only), working network load, value of dejittering delays, bit rate at which codecs operate, use or not use of VAD.

Note that VoIP has one degree of freedom more than VoATM using AAL-1: the packet size for VoIP is variable and can be chose for optimal performance, but for VoATM the packet sizes are fixed to 53 bytes.

One of the main results will be to determine the optimal value for the packet size for VoIP networks, under the considered assumptions. Evidently the packet size of the voice packets transported in the network has an impact on the network performance: larger packets need more time to fill and therefore occupy longer the servers and the queues in the network, smaller packets are not efficient since every packet does have a header that must be used to routing to the correct destination and in this case the network will be used mainly to transport routing headers instead of useful voice information.

Another result is to determine the optimal network load. Network load has the following impact on the network performance: larger network can support more voice connections but if the network is too large or over-dimensioned the delays introduces will be out of the limits of QoS.

The dejittering delay impacts the QoS because if it is too large then all packets arrive in time but the M2E delay becomes too large, if it is too small

[1] Alcatel- R&D Technical Center Romania, Timişoara
petru.serafin@alcatel.fr

the contribution to the M2E delay is negligible, but many of the packets will arrive too late and will be effectively lost, increasing the packet loss ratio.

We want to determine the optimal packet size, network load and dejittering delay. Study will be performed on two specific codecs and the conclusions of the comparison will determine the basic notions for proper network management.

II. DELAYS IN PACKET-BASED NETWORKS

By the definition of a codec, a voice encoder inputs the voice signal and outputs a stream of bits of compressed voice and a voice decoder inputs the stream of bits of compressed voice and outputs the voice signal. Both these operations, the encoding at the sender and the decoding at the receiver, need time to execute and this introduces delays that are called with the generic name of *codec delays.* The bits produced by the voice encoder are grouped together into a packet (called IP datagram in the case of VoIP). The time that is needed to fill the packets is called the *packetization delay.* Then the packets are routed over the packet-based network from source to destination. We assume the network transports only voice packets *(pure voice)* all packets must have the same priority. In the network nodes the packets compete for the available resources. Some packets need to be queued before they can be processed. This introduces *queueing delay.* If a queue overflows some packets may be lost but we assume that packet loss in the network is negligible to simplify mathematical model. The queueing delay is stochastic in nature and because of this the packets do not leave the network with a constant interpacket time, even if they entered the network with a constant interpacket time (during a period of active voice).

Because the decoder needs a Constant Bit Rate (CBR) stream (to reproduce a period of active voice) it is necessary to introduce a dejittering buffer: the first arriving voice packet is delayed over a fixed amount of time called the *dejittering delay* and from that moment on all packets are read out from the dejittering buffer at a constant bit rate. Packets that arrive too late are effectively lost. The dejittering delay should be selected so that only a few packets arrive too late.

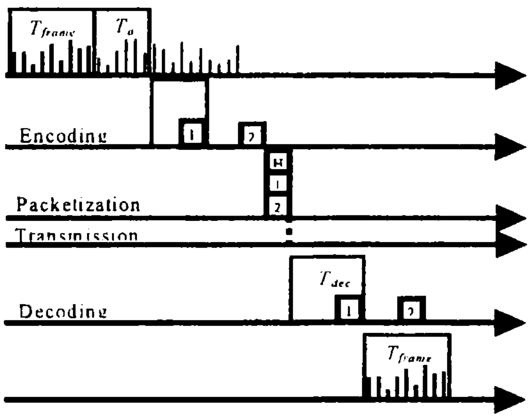
Codecs may use or not use Voice Activity Detection. If the codec uses VAD than it produces a constant bit rate (CBR) stream during periods of active voice and will produce nothing during periods of silence. If there is active voice, the encoder calculates a code word of bits per voice frame. If there is no active voice, the encoder calculates a Silence Insertion Descriptor (SID) that describes the background noise.

After the necessary samples are collected, the encoder decides whether there is active voice or not. During a period of active voice an integer number N_{frame} of consecutive code words are grouped together in one packet. Since the SID are sent very rarely, for example one SID for each period of silence, or require so few bits, it is assumed that during periods of silence the codec produces no packets.

III. CODEC DELAY

The majority of the standardized voice codecs are frame-based which means that consecutive voice samples are encoded together in a frame. We use the notation T_{frame} the delay of these voice frames. To be more effective some codecs can *look ahead* for frames, we using the notation T_a for this delay.

As input the encoder has a sampled voice signal that will be encoded and then packetized into compressed voice streams. As output, the voice signal is reconstructed by the decoder from the packets of stream of compressed voice. The processing delays are respectively T_{enc} for the encoder and T_{dec} for the decoder. These processing delays have to be smaller than the frame delay T_{frame}, or otherwise the codec



could not work in real time.

Fig. 1 represents the operation delays introduced by the codec exclusively:

Fig. 1. Codec delays

The total delay introduced by the codec as presented in Fig. 1 can be expressed by the following equation:

$$T_{codec} = T_{frame} + T_a + T_{enc} + T_{dec} \quad (1)$$

By definition the codec operational rate (or bit rate) is in relation with the frame delay T_{frame} and the number of bits per frame B_{frame} :

$$R_{codec} = \frac{B_{frame}}{T_{frame}} \quad (2)$$

In IP the packet size (datagram size) is always rounded to an integer number of bytes.

When using frame-based codec not all datagram sizes are possible because code words are normally not split over IP datagrams. A datagram always contains an integer number of frames produced by the codec. We consider a generic codec for which we neglect the frame granularity.

The number of bits per voice frame B_{frame}, the number of voice frames N_{frame} per packet and the size of the packet header S_{header} determine the packet size S_{packet} expressed in bytes, by the formula:

$$S_{packet} = \frac{N_{frame} B_{frame}}{8} + S_{header} \quad (3)$$

Note that the minimum packet size S_{packet} consists of the header only and this means a minimum of S_{header} bytes.

IV. PACKETIZATION DELAY

With these definitions the delay introduced by the packetizer can be rewritten as:

$$T_{packet} = \frac{8\,(S_{packet} - S_{header})}{R_{codec}} \quad (4)$$

The packetization delay increases as the codec bit rate decreases and as the packet size increases. The codec delay depends on the type of the codec and its hardware implementation, but has to be less than the frame delay in order for the codec to be operational:

$$T_{enc} + T_{dec} \le 2\,T_{frame} \quad (5)$$

In articles [5] and [6] are presented some parameters of the standardized codecs that will be used in this paper.

With the definitions in the previous section, during periods of active voice a source produces packets of size S_{packet} at a constant packet rate R_{packet}, given by the expressions:

$$R_{packet} = \frac{1}{N_{frame} T_{frame}} \quad (6)$$

$$R_{packet} = \frac{R_{codec}}{8(S_{packet} - S_{header})} \quad (7)$$

The packets of this source compete in the network nodes for the available resources with packets of other sources. In a node of capacity R_{node} the packet needs a service time $T_{service}$ of:

$$T_{service} = \frac{8\,S_{packet}}{R_{node}} \quad (8)$$

We define the activity index $P_{activity}$ as the fraction of the time the codec detects active voice. Notice that if the codec does not use VAD, $P_{activity}=1$, and that if VAD is used, $P_{activity} \approx 0.5$.

So the source places a load by the formula:

$$\rho_{source} = P_{activity} R_{packet} T_{service} \quad (9)$$

When replacing the terms we obtain:

$$\rho_{source} = \frac{P_{activity} R_{codec}}{\phi\, R_{node}} \quad (10)$$

We can define the number of connections of bit rate R_{codec} that can be transported over a node of capacity R_{node} if Synchronous Transfer Mode (STM) and assuming sufficient hierarchy of bit rates, using the expression:

$$N_{STM} = \frac{R_{node}}{R_{codec}} \quad (11)$$

Equation (10) can be then rewritten:

$$\rho_{source} = \frac{P_{activity}}{\phi\, N_{STM}} \quad (12)$$

where the filling factor of the packetizer is:

$$\phi = \frac{(S_{packet} - S_{header})}{S_{packet}} \quad (13)$$

The total load on a node where N_{packet} connections are compete, is then:

$$\rho = N_{packet} \rho_{source} \quad (14)$$

When replacing equation (12) in equation (14) we obtain:

$$\rho = \frac{N_{packet} P_{activity}}{\phi\, N_{STM}} \quad (15)$$

Because the number of connections is supposed to be large we can consider the arrival process to be Poisson and in theory we will use the queue model M/D/1, that is a particular case of model M/G/1 with the serving law being deterministic. Since all packets have the same priority (the service discipline is FIFO) and they all need the same service time $T_{service}$, we model the queues as M/D/1 queues.

From the queueing theory [1], [2] we know for the queue model M/D/1 the Probability Density Function (PDF) of the waiting time is expressed in the form of its Laplace transform $\hat{W}(s)$:

$$\hat{W}(s) = \frac{1-\rho}{1-\rho\,\hat{B}(s)} \quad (16)$$

where $\hat{B}(s)$ is the Laplace transform of the serving time, expressed as :

$$\hat{B}(s) = \frac{1-\exp(-sT_{service})}{s\,T_{service}} \quad (17)$$

The average waiting time is:

$$\mu = A(\rho) T_{service} \quad (18)$$

where

$$A(\rho) = \frac{\rho}{2(1-\rho)} \quad (19)$$

and the standard deviation of waiting time :

$$\sigma = V(\rho) T_{service} \quad (20)$$

where

$$V(\rho) = \sqrt{\left(\frac{\rho}{2(1-\rho)}\right)^2 + \frac{\rho}{3(1-\rho)}} \quad (21)$$

If the load ρ is high enough, the PDF of the waiting time in one node has an exponential tail determined by the dominant pole of the Laplace transform of equation (16).

V. DEJITTERING DELAY

During a period of active voice the decoder needs a constant stream of voice frames. Since in the packet-based network a stochastic delay is encountered a dejittering buffer is necessary to compensate for the difference in delays. A typical PDF of the delays introduced up to the dejittering buffer is presented in Fig. 2.

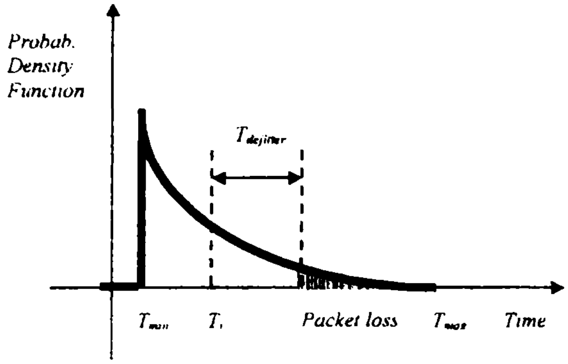


Fig. 2. Probability Density Function

Even the fastest packets experience a delay of T_{min} consisting of the coding delay, the packetization delay and the service time in network nodes, so that the minimum delay can be written:

$$T_{\min} = T_{codec} + T_{packet} + T_{network} \qquad (22)$$

where

$$T_{packet} = \frac{8(S_{packet} - S_{header})}{R_{codec}} \qquad (23)$$

$$T_{network} = \frac{8nS_{packet}}{R_{node}} \qquad (24)$$

Slower packets, which experienced a waiting time in at least one network node, are delayed more. The jitter ΔT introduced is only caused by the waiting time in the network nodes:

$$\Delta T = T_{\max} - T_{\min} = W_{\max} \qquad (25)$$

The following method is normally used to compensate for the jitter. The dejittering buffer delays the first arriving packet over a dejittering delay $T_{dejitter}$ and presents it to the decoder. All following packets are then read out from the dejittering buffer and presented to the decoder at the [illegible]. The delay of the first packet of a certain source is not known, i.e. it is not known whether the first packet of a source is a fast or slow packet. Suppose the first arriving packet of the source experienced a delay T_i. This delay T_i is a stochastic variable with the pdf of Fig. 2. The M2E delay experienced by every consecutive packet of this source is:

$$T_{M2E} = T_i + T_{dejitter} \qquad (26)$$

The worst case M2E delay is experienced by all packets of a source whose first packet was a slow one. Since T_{max} is not a strict upper limit but a quintile, there is always a small probability P_{BC} that a connection has a M2E delay larger than $T_{max}+T_{dejitter}$. We call these connections bad connections. The probability P_{BC} typically takes the value 10^{-5}, refer to document [3].

Once the dejittering delay $T_{dejitter}$ is chosen, the time the n-th packet is to be read out of the buffer is fixed, i.e. n interdatagram times ($nN_{frame}T_{frame}$) after the first packet. The n-th packet was transmitted n interdatagram times after the first packet and experienced a delay T_n. If this delay T_n exceeds $T_i - T_{dejitter}$ the n-th packet arrives too late, after the instant it had to be read out from the buffer, and is effectively lost. So the packet loss associated with the dejittering delay $T_{dejitter}$ is characterized by:

$$P_{loss} = prob\{T > T_i + T_{dejitter}\} \qquad (27)$$

This is illustrated as shaded area in Fig. 2. The worst case packet loss is experienced by a source whose first packet is a fast one (i.e. has a delay of T_{min}). The cell loss probability P_{loss} may take values in the range $[10^{-5}, 10^{-2}]$ depending on the codec used. For codecs that are designed to be robust against packet loss the relatively large value of 10^{-2} may be chosen.

By carefully choosing the dejittering delay $T_{dejitter}$ the M2E delay experienced by a source can be traded off against packet loss experienced by that source. Consider a connection that traverses n nodes. We want to guarantee that all, except a very small fraction P_{BC} of the so-called bad connections, experience a M2E delay smaller than a certain delay bound T_{M2E} and that all connections have a packet loss smaller than a certain tolerated packet loss P_{loss}. From the reasoning in the previous section we know that those two restrictions translate to :

$$prob\{T > T_{\min} + T_{dejitter}\} < P_{loss} \qquad (28)$$

and

$$T_{\max} + T_{dejitter} < T_{M2E} \qquad (29)$$

This first restriction can be rewritten as:

$$T_{dejitter}R_{codec} - \frac{8D(\rho, n, P_{loss})}{N_{STM}}S_{packet} > 0 \qquad (30)$$

with

$$D(\rho, n, P) = nA(\rho) + C_n(P)\sqrt{n}V(\rho) \qquad (31)$$

The second restriction can be rewritten as:

$$8\left\{1 + \frac{[n + D(\rho, n, P_{BC})]}{N_{STM}}\right\}S_{packet} + T_{dejitter}R_{codec} < (T_{M2E} - T_{codec})R_{codec} + 8S_{header} \qquad (32)$$

These expressions represent in fact restrictions on the packet size S_{packet} and the dejittering delay $T_{dejitter}$. Fig. 3 illustrates these restrictions.

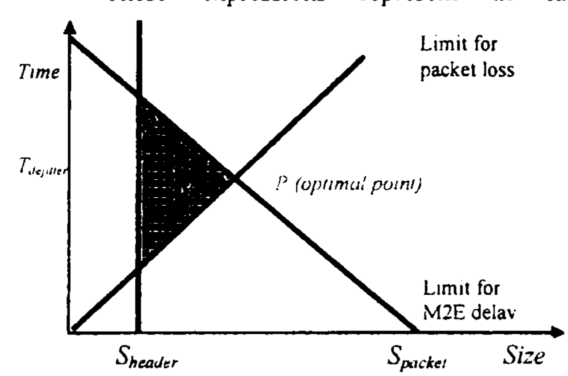


Fig. 3. Packet size and dejittering delay

Using equation (15) we see that the number of connections supported on a network node under load ρ can be expressed as:

$$N_{connections} = \frac{\rho\phi}{P_{activity}}N_{STM} \qquad (33)$$

It is seen that the number of connections depends on the filling factor ϕ, which in turn is determined by the packet size S_{packet}.

The shaded area in Fig. 3 is the triangle with the allowed pairs $(S_{packet}, T_{dejitter})$. This triangle changes as the load ρ changes. Remark that the triangle corresponding to a higher load is completely included in the triangle corresponding with a lower load: as the load ρ increases there are fewer paires allowed. Also note that the triangle of allowed pairs for a connection that runs through $n+1$ nodes is a subset of the triangle of allowed pairs for a connection that runs through n nodes. Consequently, if we choose the pair based on connections that run through $n+1$ nodes, the QoS limitations for connections that run through n nodes are also respected.

The following deductions are made to determine the optimal packet size in an IP environment. At given networks load ρ the maximum number of connections n_{max} is reached if the filling factor ϕ is as high as possible. The maximal ϕ is reached in the rightmost point of the triangle of allowed pairs $(S_{packet}, T_{dejitter})$ that is represented as point P in Fig. 3. This point establishes the optimal values for the packet size S_{packet} and the optimal values for the dejittering delay $T_{dejitter}$. With these values we determine the maximal number of connections in the network. It can be observed that at low loads the number of connections increases almost linearly. At higher loads this factor becomes important. So there exists an optimal network load. It is difficult to determine the optimal load analytically, but numerically this constitutes no problem. Note that the optimal value of the network load ρ does not depend on the codec bit rate R_{codec}.

In our reasoning we neglected the fact that the packet size S_{packet} can only take certain values. As explained earlier, in an IP environment an integer number of voice frames N_{frame} is put in an IP datagram that are always rounded to an integer number of bytes (multiple of 8). The rounding to the nearest byte has no significant influence on the obtained results.

VI. NETWORK LOAD

In this section we report some dimensioning results for an IP network that was analyzed with the following QoS requirements assumed:

$$\begin{cases} T_{M2E} < 200ms \\ P_{loss} < 10^{-5} \\ P_{BC} < 10^{-5} \\ T_{codec} \approx 0 \end{cases} \tag{34}$$

Other values for these limits can also be considered. The codec delay T_{codec} is taken to be negligible or it is supposed to be subtracted from the M2E delay.

It is assumed that the voice information is transported in IP datagrams (ex. 20 byte header in IPv4) using UDP (ex. 8 byte header) and RTP (ex. 12 byte header). So, the IP overhead is S_{header}=40. In this section the codecs do not use VAD, $P_{activity}$=1.

The analysis is done for two standardized codecs : G.723.1 (cf. [7]) and G.729 (cf. [8]).

The number of connections that can be accepted on a network consisting of one node without violating the QoS limitations is given as a function of the load ρ, and with the codec bit rate R_{codec} and the node capacities R_{node} as parameters.

Table 1 shows results of the optimal load $\rho_{optimal}$ for different node capacities and the maximum number of VoIP connections for these capacities. For comparison it is indicated the maximum number of connections supported by an STM node of the same capacity with 8kb/s. To make the comparison with STM more correctly we choose capacities close to Plesiochronous Digital Hierarchy (PDH) and Synchronous Digital Hierarchy (SDH). The link capacity of 2 Mb/s corresponds more or less to a PDH E1 system. A transport capacity of 2.048 Mb/s (=32x64 kb/s) was taken. The link capacity of 34 Mb/s corresponds to a PDH E3 system with a net transport capacity of 33.92 Mbs (=530x64 kb/s). The achievable multiplexing gain is defined as the ratio of the number of VoIP connections to the number of 64 kb/s STM connections. As the capacity R_{link} of the node increases, the optimal load increases.

Table 1: Optimal load and maximum number of 8 kb/s connections for different node capacities.

R_{link} [Mb/s]	$\rho_{optimal}$	n_{max} IP [8kb/s]	n_{max} STM [8kb/s]
2	0.905	175	256
34	0.975	3368	4240

The optimal load decreases as the number of nodes increases. Table 2 gives the optimal load $\rho_{optimal}$ for networks of different capacity R_{node} as a function of the number of nodes n. For a network capacity of 34 Mbs the optimal load hardly changes when the number of stages n increases. For a network of capacity of 2 Mbs a larger impact is observed. Table 2 also shows the number of VoIP connections compressed to 8 kbs that can be supported at that optimal load.

Table 2: Optimal load and maximum number of 8 kbs connections for VoIP network of capacity R_{link}

R_{link}	2 Mb/s		34 Mb/s	
N nodes	$\rho_{optimal}$	n_{max} IP [8kb/s]	$\rho_{optimal}$	n_{max} IP [8kb/s]
1	0.905	175	0.975	3368
2	0.89	171	0.97	3351

To reach the maximum number of connections at the optimal load the size of the IP datagram S_{packet} and the dejittering delay $T_{dejitter}$ have to be chosen optimally.

VII. OPTIMAL PACKET SIZE AND OPTIMAL DEJITTERING DELAY

The optimal size of the IP datagram S_{packet} and the optimal dejittering delay $T_{dejitter}$ are presented as a function of the codec bit rate R_{codec}.

The following figures (Fig. 4 and Fig. 5) represent the optimal packet size and optimal dejittering delays for networks of capacity 2Mb/s for different codec bit rates with 1 and respectively 2 stages:

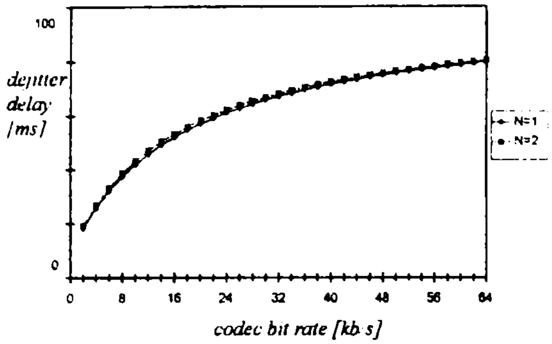


Fig. 4. Optimal packet size [bytes] for different codes bit rates [kb/s] for a network of 2 Mb/s

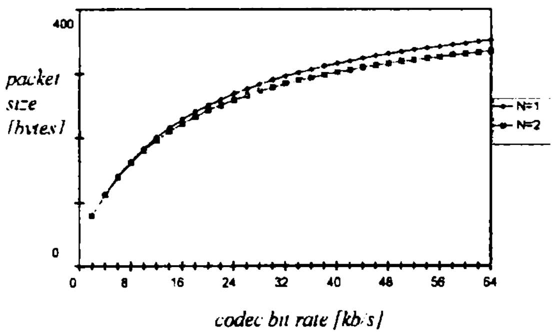


Fig. 5. Optimal dejittering delay [ms] for different codes bit rates [kb/s] for a network of 2 Mb/s

The optimal packet size increases if the codec bit rate increases because as the codec bit rate R_{codec} increases, the packetization delay becomes less important. Also because the queueing delay is proportional to the packet size, the optimal packet size decreases with increasing number of stages.

The dejittering delay is chosen to compensate for the variability of the M2E delay. This variation of the delay is introduced in the queues during the transport. All other delays are deterministic in nature. If the codec bit rate increases, then the packetization delay decreases. Since the M2E delay remains the same, the reduction in packetization delay means that the time spent in the queues can increase and so the variability of the delay increases. Therefore, the optimal dejittering delay increases as the codec bit rate increases.

VIII. CONCLUSIONS

In this article we investigate performance for transporting voice over packet-based networks. The main assumption was that only real-time traffic is transported on the network of pure voice with no data traffic in the network so there would be no need for priority mechanism in the queues. Another supposition was that all codecs operate at the same bit rate and also the sizes of the packets produced (containing compressed voice) is the same for all codecs operating in the network.. To correctly dimension the packet-based voice network, the design elements are packet size, network load and dejittering delay.

In a voice network, packets of different sources compete for the available resources and so is introduced a stochastic delay depending on the network load. At the receiver the jitter introduced in the queues of the network has to be compensated. The purpose of this report is to indicate how to choose these design parameters to obtain maximum number of connections, with the respect of some limitations of QoS: M2E delay < 200 ms and packet loss < 10^{-5}.

The optimal load at which a network has to operate to allow the most connections was determined first. This optimal load increases with the capacity of the network and decreases as the number of nodes traversed increases. But the optimal load is independent from the codec bit rate. At loads below this optimal load the network is not efficiently used. At loads above this optimal load the QoS limitations cannot be met for all connections. On a network consisting of different nodes, the connections should be routed such that the least number of nodes are passed through. It is for connections that run through the most nodes that the QoS limitations are most difficult to meet.

Corresponding to this optimal load and the number of stages, it was determined the optimal values for the packet size and the dejittering delay for two specific codecs. Choosing the packet size too small means that the network is not optimally used. Choosing the packet too large QoS requirements cannot be met. By decreasing the dejittering delay on some connections there is an excess of packet loss. By increasing the dejittering delay some connections experience an M2E delay larger than tolerated limit.

The main conclusion is that VoIP and VoATM with QoS is possible if the IP or ATM network is properly managed to respect the following objectives:

- the maximum number of nodes a connection traverses should be limited (the worst case being connections traveling through the maximum number of nodes)
- the network should not be loaded above the optimal load (corresponding to worst case)
- the optimal packet size and the optimal dejittering delay corresponding to the worst case should be used

It can be concluded that at low codec bit rates the smaller overhead of ATM using AAL-1 makes that ATM has better performance than IP. At higher bit rates the larger packet sizes of IP are more efficient.

This paper identifies the performance under ideal conditions, but if the network conditions are not homogeneous (for example codecs of different bit rates, different packet sizes) a new model has to be studied. Also the case where a mixture of data traffic and voice traffic is allowed on the network with priority given to voice over data is subject of future works.

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