Quality of Service in Voice over Packet Infrastructures

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Introduction

This white paper compares the attainable QoS of **Voice over IP (VoIP)** and **Voice over Asynchronous Transfer Mode (VoATM)** packet technologies as investigated by various carriers and equipment manufacturers today.

Voice over Packet technology is designed to break down the barrier between traditional public switched telephone networks (PSTN) and data (ATM or IP) networks. For the carrier, the opportunity of Voice over Packet technologies lies in optimizing network infrastructure and in extending customer reach beyond PSTN subscribers, while at the same time maintaining end-to-end service capabilities and service offerings. The concept of transporting voice across packet backbones is an important building block of the so-called "Next Generation Network" infrastructure or NGN for short.
NGN concepts are analyzed today in discussions from various angles and a thorough understanding of Voice over Packet infrastructures is crucial to success in the future.

In this white paper, emphasis is put on the question how voice can efficiently be forwarded in a packet network so that Quality of Service (QoS) is maintained. This paper shall also serve as a technical input to the strategic decision how to integrate voice into NGN infrastructures. It shall provide a thorough analysis of the advantages and disadvantages of both, VoATM and VoIP, infrastructures.
The aspect of network planning and engineering is discussed by Stademann and Wimmreuter at Networks 2002, and is not part of this white paper.

Network Model

The transmission of telephone calls across a packet-based backbone has primarily two competing technical approaches:

- Voice over Internet Protocol (VoIP) or
- Voice over Asynchronous Transfer Mode (VoATM)

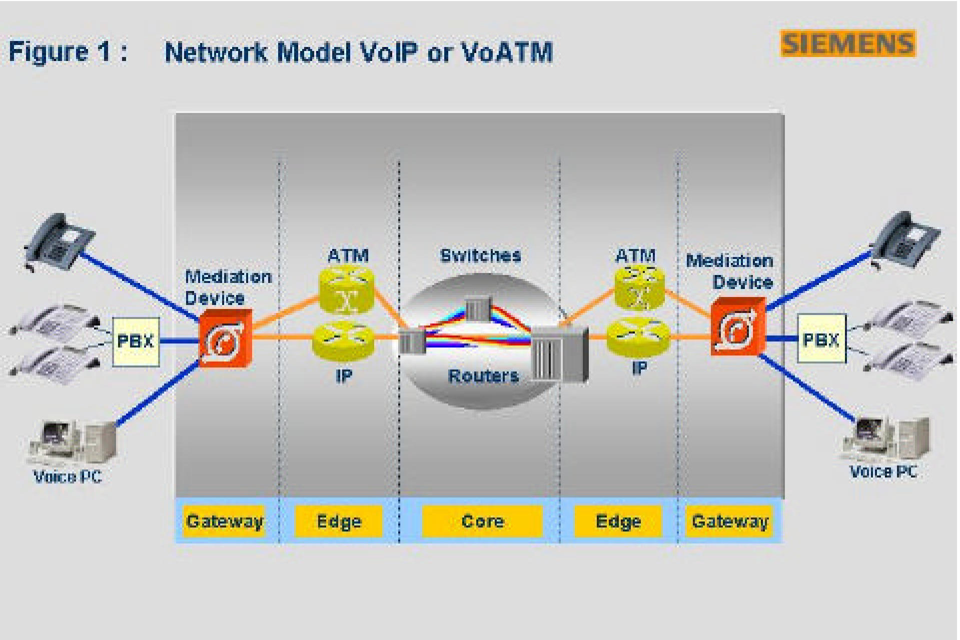


Figure 1 : Network Model VoIP or VoATM

The topology as described in Figure 1 is remarkably similar for both approaches with the most important difference being the type of edge switch: Either an ATM switch for the VoATM approach or an IP router for the VoIP approach. Modern Dense Wavelength division multiplexing DWDM transmission equipment is accepting both ATM and IP interfaces in a similar way.
Mediation Devices are used to transform the Voice over Packet stream into traditional TDM signals and vice versa.

Examples of Mediation Devices are:

- Central Office Gateways terminating classical TDM lines towards a packet backbone
- Integrated Access devices (IADs) or Customer Premise Gateways (CPGs)
- Mediation Devices for IP enabled PBXs (Private Branch Exchange)
- Dedicated IP Phones or IP PBX systems
- PCs with an integrated voice software package or an integrated voice processing module

There are market drivers, which are supporting either IP or ATM packet backbone choices:
Drivers in favor of ATM are e.g. possible line emulation services and bandwidth management granularity for low bandwidth access solutions. This is the reason why e.g. Digital subscriber loop DSL and Wireless Local Loop WLL access systems are using ATM based access solutions.

In favor of IP is the large number of installed end user applications based on IP. IP is also the protocol of choice for in-house communication local area networks (LANs). This makes IP the natural choice for all CPE based IP phones, PC/Server based software voice solutions or IP enabled PBXs. IP access solutions are also increasingly used in UMTS networks, CATV access infrastructures and in IPoDSL wire line solutions.

The following chapters assume that:

- A converged voice and data packet network for all services is the target network layout.
- Optimum bandwidth utilization is desired in order not to waste network resources.
- The packet infrastructure allows the handling of different services with different priority.
- Enough bandwidth is available to transport all required services in an end-to-end network path.

Telephony Service Quality

End user expectations for carrier grade telephony are: (based on current experience)

- System always ready to accept call attempts
- Calls are not denied due to lack of network resources
- Fast connection establishment
- An established connection does not break down
- Good voice quality
- The voice channel can be used for the transport of other services like fax and modem.

Whereas the first four points are not in discussion as a major blocking point for either VoATM or VoIP, the last two points “good voice quality” and “channel transparency” are subject for heavy debate between VoATM and VoIP advocates.

The former market perception of Voice over Packet technologies mainly stems from past end user experiences with voice calls via narrowband dial-in lines, heavily overbooked by their Internet Service Provider (ISP). However, VoIP has long left this niche of low price telephony and is competing with VoATM on the carrier grade Voice over Packet market. Thus carrier grade telephony sets the framework for the comparison between VoATM and VoIP.

In Voice over Packet networks, voice quality is influenced by

- Voice Encoding/Decoding (Codec)
- Delay and Delay Variation (Jitter)
- Packet Loss
- Echo

The Codec aspects will be evaluated in section “Encoding” below.

Delay is incurred by all network elements along the voice path. The delay contribution of the Mediation Devices depends mainly on the Codec and its implementation. Delay is of particular importance in interactive voice communication and it should be kept at a minimum in carrier grade networks.

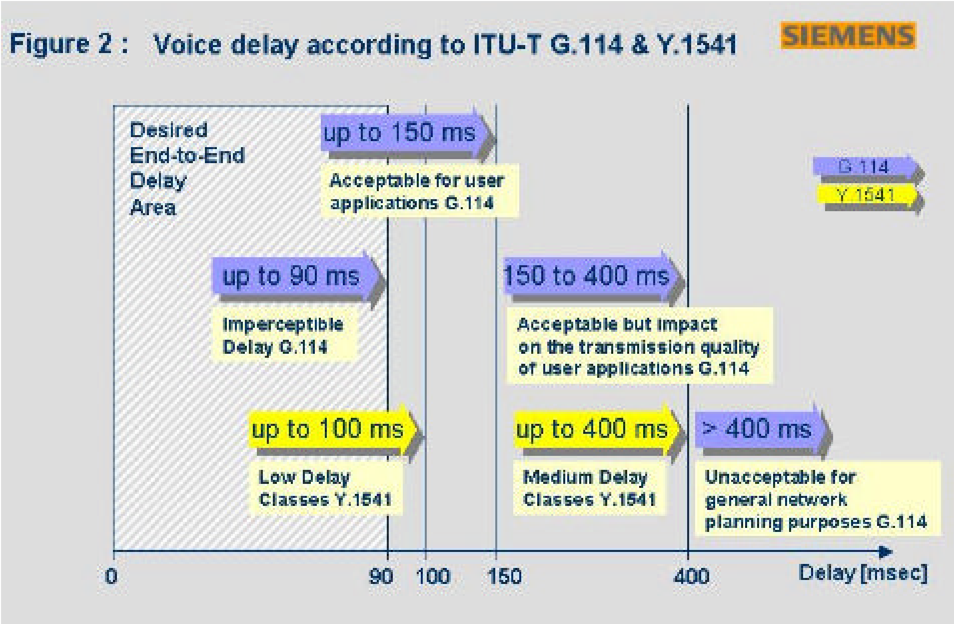


Figure 2 : Voice delay according to ITU-T G.114 & Y.1541

As outlined in figure 2 above, ITU-T recommendation G.114 suggests that delay impacts will be imperceptible below an end-to-end delay of 90ms. ITU is currently integrating this and the findings of ETSI TIPHON into recommendation Y.1541 that is especially tailored towards Voice over Packet issues.

In all practical implementations the target is to keep the delay lower than this threshold for a carrier grade network design. We will describe in detail how this is achieved in later sections.

In a similar way, also packet loss is kept under control by appropriate design of IP and ATM networks. Both technologies offer methods to keep packet loss low and this subject is not a major point of concern in practical voice implementations.

Another voice quality issue is Echoing:
As ITU-T recommendation G.131 shows, the human perception of echo depends on signal intensity and delay. And as a rule of thumb: the longer the delay, the more disturbing the echo. In voice telephony there are two main sources of echoes:

a) the electrical hybrid echo
b) the acoustic echo.

Since an echo delay of more than 20 ms reduces the conversation quality significantly, it is reasonable to generally introduce echo compensators capable to keep the electrical hybrid echo intensity low. This strategy is the same as in regular voice networks and is independent of the voice over packet technology. Acoustic echoes must be dealt with by the customer premises devices either through appropriate terminal design or through acoustic echo compensation. Again there is no difference from regular voice networks.

The assessment of the resulting voice quality is still a complex issue. Quality should not only be measured under ideal conditions, but also in the presence of background noise and interference talkers. The perceived quality will depend on the talker and the language. Subjective statistical tests for a comparative ranking of different Codecs and networks have been defined by ITU-T [P.800, P.810, P.830, P.831 and P.832]. The results are given in terms of Mean Opinion Scores (MOS). We will give some test results for the most commonly used Codecs in section "Encoding".

In the following sections we will describe the technological basis for carrier grade Voice over Packet networks which provide high voice quality and low delay for a public telephony service.

Voice over Packet Technologies

Before voice can be transported via a packet network, the continuous voice stream must be encoded and converted into packets,. i.e. the incoming signal is sampled over a predefined time period (t_{SAMPLE}) and a digital value is determined. Codecs may be distinguished according to digitizing scheme, frequency cut off, silence suppression techniques, compression, etc. In the TDM world, the 64 kbps capacity unit is well established, e.g. in case of PSTN/ISDN every 125 µsec a Byte is put on the line. The chosen Codec has a significant influence on the voice quality – e.g. due to the sampling time. In particular, when compression is applied the sampling time can be significantly increased. In an IP based network the digital values are collected over a (adjustable) Packetisation Period (t_{PP}) to aggregate an effective amount of data to put into a single packet (e.g. t_{PP} =10ms).

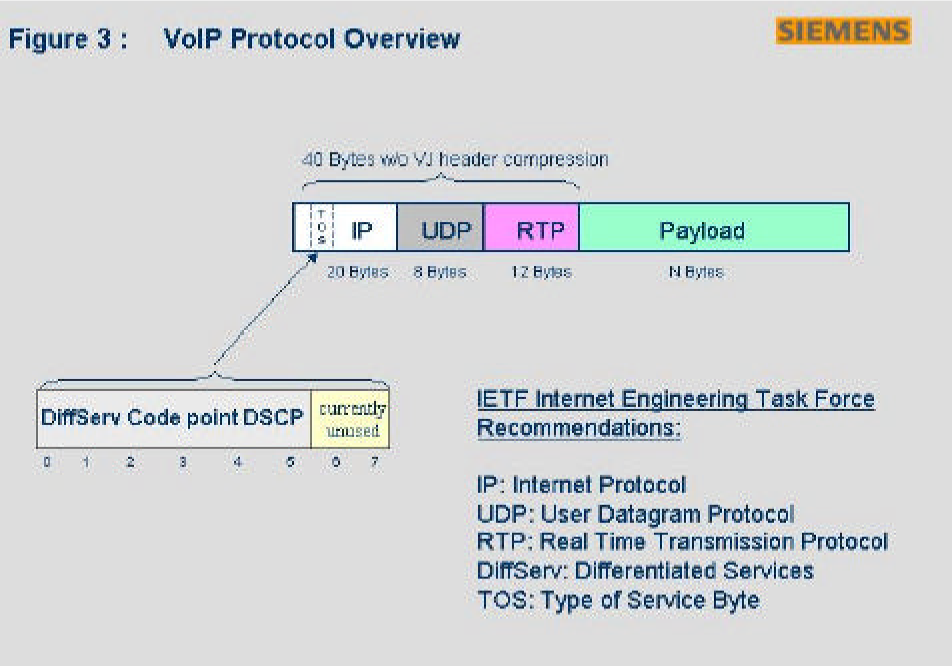
In an end-to-end path, in addition to the packetisation period PP there is the encoding and decoding process in the Mediation Devices on both sides. To accommodate the potential differences in the arrival time of voice packets there needs to be a jitter buffer at the destination to smoothen the outgoing voice stream.
Hence, the signal delay a voice stream experiences across an IP infrastructure until being heard at the destination can be estimated as:

$$\mathbf{t_{DELAY} = t_{Encode} + t_{PP} + t_{Transport} + t_{Jitter} + t_{Decode}}$$

Obviously, there is a trade off: The larger the Packetisation Period, the more efficient the bandwidth usage. Unfortunately, t_{PP} fully adds to the overall delay.
Therefore, the packetisation period is often reduced to around 5 to 10ms in order to minimize delay. This equals 40 to 80 bytes payload for non-compressing Codecs.

Voice over IP

In Voice over IP technology, the voice payload is attached to several header blocks as described in figure 3.



While the IP-header allows to route the packet to the right destination device, the User Data Protocol (UDP) header information provides the application (e.g. for voice, the particular connection the packet supports). The Real-Time-Protocol allows the application on the receiving side to keep track of the order the packets were sent, e.g. when a packet overtakes its predecessor the application is capable of reordering.
Voice quality is much more sensitive with respect to delay and jitter than with respect to packet losses. This means that a fast packet transfer is much more important than a reliable packet transfer – hence, the choice of UDP instead of TCP.

In order to prioritize IP traffic with voice payloads from other types of IP traffic e.g. E-Mails and File transfers, the so-called DiffServ (named according to the IETF working group “Differentiated Services”) codepoint marking can be used.

As shown in Figure 3, the IETF DiffServ group defined the “Type Of Service (TOS)” byte in the IP header with the intent „to provide scalable service discrimination in the Internet without the need for per-flow state and signaling at every hop“. RFC 2474 redefines the semantics of the IPv4 TOS byte and IPv6 traffic class byte, which is now called DS (differentiated services) byte.
The IP router implementation must offer at least two independent queues in order to support the differentiated services approach.
Expedite forwarding is then defined according to some bit values xxx (the higher the bit values, the lower the average queuing delay) in the DSCP field (typical pattern “xxx000”).

Voice over ATM

The ATM standard defines several adaptation layers, each designed to support adaptation (conversion to and from ATM cell format) for different types of services. ATM Adaptation Layer (AAL) protocols have been defined to specify how these conversions are performed. The AAL-1 and AAL-2 protocols were defined to transport real-time, delay-sensitive applications, such as voice. The AAL-3, AAL-4 and AAL-5 protocols were defined to transport delay-tolerant data or multi-service traffic.

The AAL-1 protocol made it possible to perform unstructured or structured circuit emulation over an ATM backbone for voice traffic. This type of connection would typically carry end-to-end transparent circuits between TDM voice switches or digital PBXs. It is possible to deploy voice-over-ATM through AAL-1 circuit emulation, but it is inefficient because no bandwidth optimization is available (AAL-1 connections allocate bandwidth even when it is not being used) and additional overhead in the ATM cell and AAL-1 layer of 12-15 percent is required.

Recognizing that AAL-1 had limitations, the ITU-T (International Telecommunications Union) developed an AAL protocol geared to packetised voice over ATM (ITU-T, I.363.2).
The ATM Forum adapted the ITU-T standards and approved the ATM Trunking Using AAL-2 for Narrowband Services. Specification (AF-VTOA-0113.000) in February 1999.
The ATM Adaptation Layer Type 2 (AAL-2) standard supports variable bit rate transmission and constant bit rate, making it applicable for a wider variety of services. AAL-2 provides bandwidth-efficient transmission of low-rate, short and variable packets for delay-sensitive applications like voice.

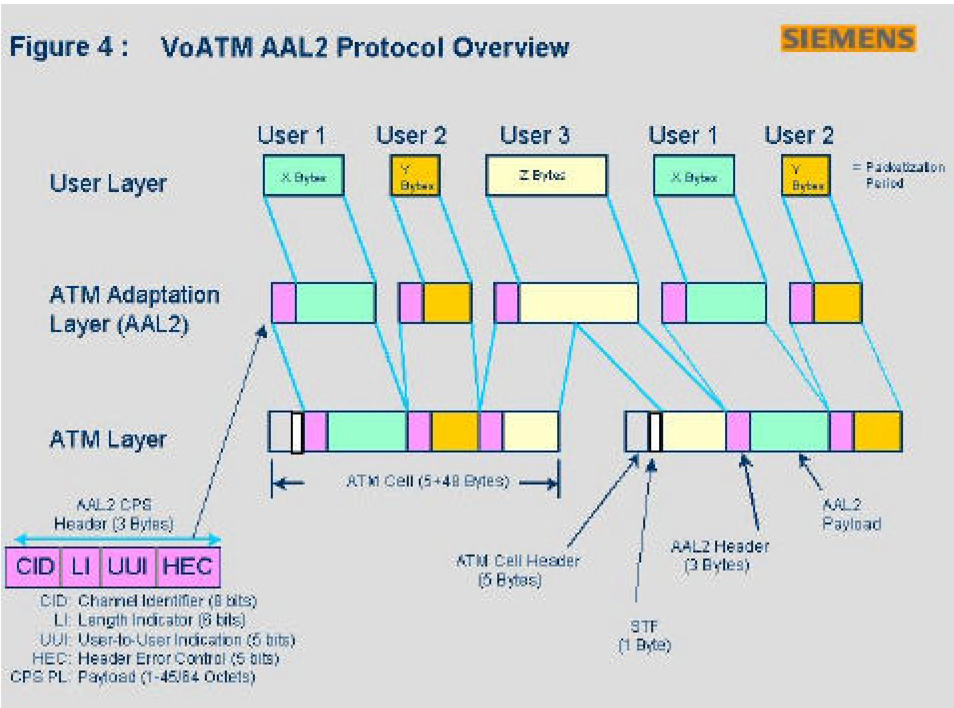


Figure 4 : VoATM AAL2 Protocol Overview

AAL-2 goes beyond AAL-1 by supporting the variable bit rate, real time (VBR-RT) traffic class. That means users can send traffic at variable rates depending on the availability of user information, while still satisfying applications that are sensitive to variations in cell delay.

As shown in Figure 4, AAL-2 statistical multiplexing is provided to make optimum use of network resources. Unlike the fixed bandwidth allocation of AAL-1 circuit emulation, with AAL-2 unused bandwidth can be instantly allocated to other traffic demands. Multiple voice channels with varying bandwidth can be carried on a single ATM connection.

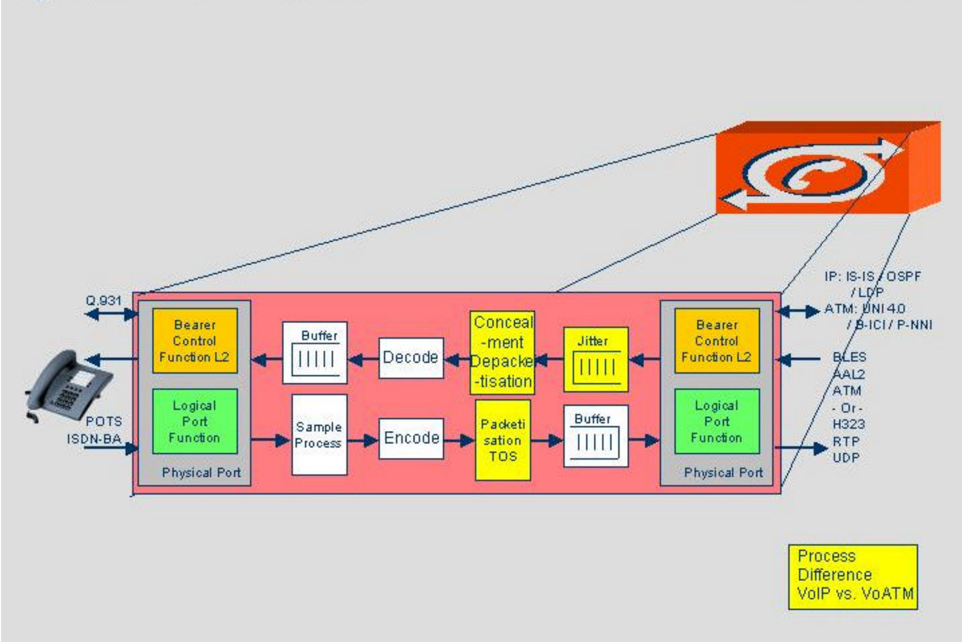
AAL-2 supports variable payload size in each cell and allows several short-length packets from different sources or from several packetisation periods to be packed into one or more ATM cells. This paper assumes AAL-2 encapsulation for all following VoATM discussions.

A typical packetisation period in the ATM case is less than 6msec or 48 bytes for uncompressed voice. This means each period fits into one cell. The associated AAL2 header overhead and cell tax (3 and 5 bytes per 48 bytes payload) is lower than the IP header (40 bytes per typically 80 bytes payload).

Mediation Impact on Voice Quality

The Mediation Device overview in Figure 5 is using functional block definitions as described by the Multiservice Switching Forum (MSF).

Figure 5 : Mediation Device



We will now trace the path from a regular telephone connection through the ingress Mediation Device and leaving the network via the egress Mediation Device to the destination telephone connection. The backbone switch/router is considered transparent in this section of the paper. We will analyze its impact in the backbone section.

There are three functional blocks, which differ between VoIP and VoATM gateway implementations:

- Packetisation ingress
- Jitter Buffer egress and
- De-Packetisation egress

However, they are only a small contribution compared to the steps that are identical between ATM and IP, as can be seen by the following analysis of the end-to-end path:

Sampling Process

The sampling process analyzes (“samples”) the incoming digital voice stream and collects data blocks (e.g. 8 bit values) at defined time intervals. The length of this sampling interval only depends on the Codec. Hence, there are no differences in the sampling process itself between VoIP and VoATM by definition.

Encoding

The following list in Figure 6 provides an overview of the most important Codec definitions from the International Telecommunication Union (ITU):

Figure 6 : Technical data for the most important Codec´s SIEMENS

CODEC	Bitrate	Algorithm	Silence Suppression	Sample Time	Look Ahead
G.711	64 / 56 kbps	PCM	proprietary like G.729 Annex B, with CNG according to G.711 Appendix II	0,125 msec	-
G.726	40, 32, 24, 16 kbps	ADPCM	VAD as G.711	0,125 msec	-
G.729A	8 kbps	CS-ACELP	G.729 Annex B	10 msec	5 msec
G.723.1	6,3 – 5,3 kbps	ACELP / MP-MLQ	- / G.723 Annex A	30 msec	7,5 msec

G.711 is the standard Codec in today's telephony networks. Thus it sets the benchmark for carrier grade voice quality. The bit rates supported are 56 kbps (US version) and 64 kbps (European version). G.726 is available from 40 kbps down to 16 kbps. Both G.711 and G.726 down to 32kbps are transparent to tones and allow for a reasonable playback of music during announcements. Fax and modems are transported in band by both Codecs, as will be described later. Highest modem speeds can only be achieved with G.711. This transparency makes G.711 a natural choice for a smooth evolution from the PSTN/ISDN.

Figure 7 : Codec ITU-T P.800 Measurement Results

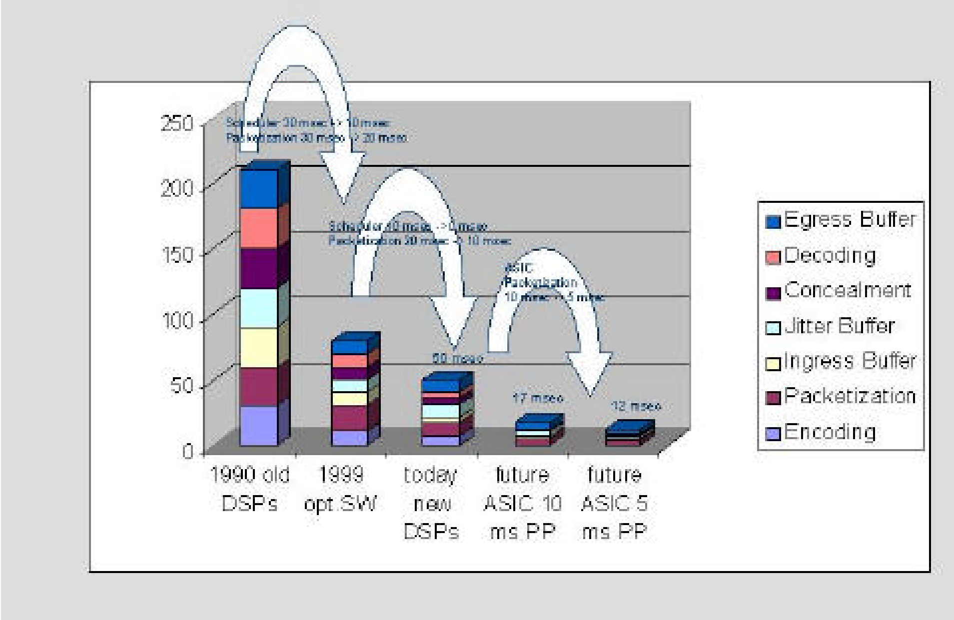
CODEC	Bitrate	MOS result
G.711	64 kbps	4,3
G.726	32 kbps	4,0
G.729	8 kbps	4,0
G.723.1	6,3 kbps	3,8

Mean Opinion Score (MOS) results for the most important Codec´s under good network conditions.
[Rudkin et al. BT Journal]

If one would like to compress to bit rates below 32kbps, G.729 A is a good alternative. As can be seen from figure 7, its voice quality is comparable to G.726 at 32kbps. However, this high compression rate needs an additional look-ahead for the sampling process and results in a larger processing delay, which limits its dialogue quality. Also, tones as well as fax and modem data have to be carried out-of-band. G.723 goes further down in bandwidth (5.3 or 6.3 kbps), but compromises on quality. Therefore it should only be used in specific access networks where bandwidth is scarce, or for the interworking to given PC clients.

The encoding process is usually done in digital signal processor (DSP) chips and depends on the chip generation in use for this process. G.711 Codecs are already available in faster ASIC chips today.
There are certain delay values absolutely necessary for the look ahead algorithms as defined by ITU. However, these significant delays are the same for both, VoIP and VoATM flows, by definition. So the real difference here makes the chip generation in use. Figure 8 gives an outline of the ongoing developments in this area:

Figure 8 : Progress in CODEC developments



The last years have seen short development cycles in a rush to improve the Codec delay situation. Current DSP technology complies with the requirements of most networks. The new generation of G.711 ASIC based Codecs for example generates not more than 12 msec delay including packetisation and jitter buffer in the end-to-end path. Again these developments are applicable for both, VoIP and VoATM flows, and they are dependent on vendor implementations.

The Codec implementations should include echo cancellation procedures to eliminate the electrical hybrid echo. Due to the good experience with these devices, no further issues are expected for Voice over Packet applications. The echo cancellation strategies of regular voice processing units will be also applicable here.

Voice Activity Detection

During voice encoding, the Mediation Device may detect and suppress the silent periods. In this way, the IP network load can be reduced. The Codecs G.723.1 and G.729A have their own silence suppression strategies (described in G.723.1 Annex A and G.729 Annex B). G.711 has no standardized silence suppression strategy. For this reason, Mediation Devices often use the same mechanism to suppress the silent period and insert comfort noise generation as defined in G.729 Annex B. G.726 uses the same silence suppression and voice activity detection algorithms as G.711.

With Comfort Noise Generation (CNG), the users involved in the conversation have the impression that valid information is sent during the silent periods. Comfort Noise consists of a message generated in the source gateway that informs the destination gateway to play comfort noise at the level specified. This message is sent at the beginning of a silent period and repeated periodically with the current noise level.

Fax, Modem and ISDN data

The Mediation Device should also support transport of modem and fax traffic.

When a modem call is detected, the Codec is switched to G.711 without echo suppression and the silence suppression is switched off. This ensures that the modem signals are transported transparently to the destination Mediation Device.

In case of Fax traffic, a similar procedure can be applied. The fax connection is switched to G.711 without silence suppression but with echo cancellation. The T.30 protocol is transported transparently across the packet network. An alternative is provided by T.38. Here, the T.30 protocol is terminated in the Mediation Device and the data is sent to the destination Mediation Device via T.38 (Fax over IP). The destination Mediation Device then establishes a T.30 connection to the destination fax machine.

An ISDN data call (unrestricted bearer call 64 kbps) is detected already in the Mediation Device controller. Then the Mediation Devices are informed in the first MGCP command to use G.711 transparent mode without echo suppression and without silence suppression.

Packetisation

Data units delivered by the encoding process are grouped into fixed size packets corresponding to the packetisation period. VoIP packets are then expanded with the required RTP/UDP/IP header and VoATM packets are expanded with AAL2/ATM headers and processed as cells.
The difference between the Voice over Packet technologies depends on the setting of the packetisation period. If the same setting is taken for VoATM and VoIP flows, the only difference is the longer header length of the VoIP framing process. This might take some µsec longer but is negligible compared with the rest of the delay values.
Hence, the characteristics of voice packetisation are the same for VoIP and VoATM .

Ingress Buffer

The buffer length of the ingress Mediation Device depends on the configuration of the Codec functional block and is also decreasing in new implementations.

Network Backbone

The packet flow carrying the original voice message is now traveling across the ATM or IP based backbone. We will discuss the backbone issues more specifically in the next chapter.

Jitter Buffer

At the destination Mediation Device, the first process in action is the Jitter buffer, which is responsible for aligning the incoming packets in the right order and also compensating network jitter generated by switches and routers. In order to play the analog waves in the correct timely order, a buffer needs to adjust the transmission time differences inherent in packet networks. Such a buffer is called Jitter buffer.
The jitter buffer in the Mediation Device behaves as described in Figure 9.

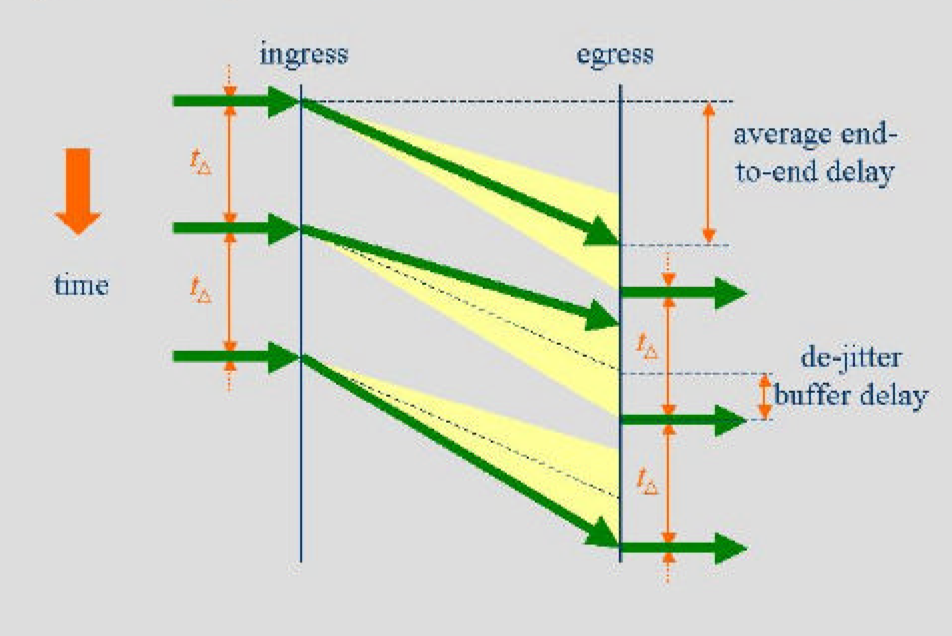


Figure 9 : Principle of a Jitter Buffer

Packet Loss Concealment and De-Packetisation

Packet loss concealment reduces the impact of lost packets via generating new samples based on information from past sample periods. This technology allows being more generous towards the IP router and ATM switch policies of dropping packets in case of congestion. However, since Packet Loss Concealment introduces an additional processing delay, it is only recommended under adverse network conditions.

Decoding

The decoding process is performing the inverse action from the encoding process. Together with the encoding process, the decoding process may produce significant delay but is again identical for both Voice over Packet technologies.

Egress Buffer

Finally the Egress buffer merges the resulting data streams into uplink packets. Special attention has to be paid if traffic with long packets is mixed with real time traffic on a low-bandwidth uplink. In an IAD with 2Mbit/s uplink, the largest IP packets of 1500 Bytes may result in a delay of 6msec. For ATM, this effect is avoided by fragmenting all traffic into cells of 48 Bytes payload. For IP, fragmentation of packets in wire speed should be included. Having taken care for this effect, the resulting traffic characteristics are the same for ATM and IP.

Mediation Impact Conclusion

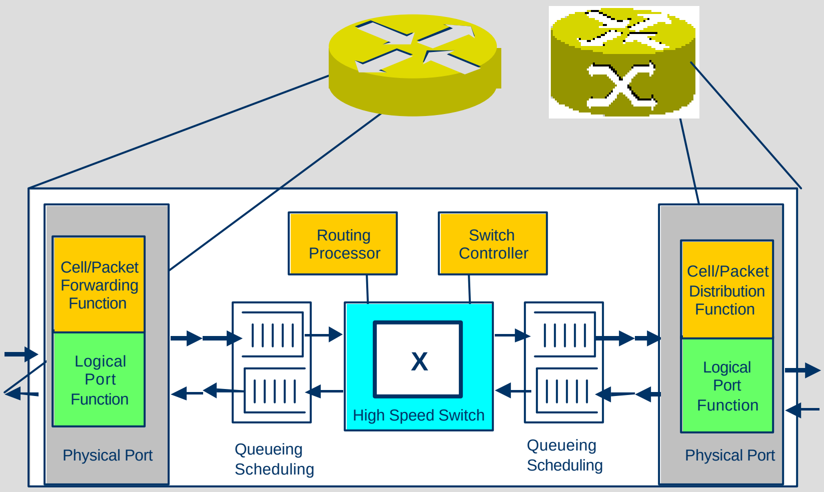
We have shown that no significant differences exist between VoIP and VoATM flows through Mediation Devices because existing differences are negligible compared with the main delay driver: the encoding/decoding process.

Now, what is the difference between ATM switches and IP routers in the backbone?

Backbone Impact on Voice Quality

Modern ATM switches and IP routers use a remarkably similar hardware architecture as shown in Figure 11:

Figure 11 : ATM Switch / IP Router Basic Architecture



Simplified Basic Architecture

Both are using a non blocking switching matrix surrounded by line cards for port termination, cell/packet forwarding and queuing and scheduling functions. This means there is no real architectural difference between switches and routers.

IP packet header processing is more complex (due to its length) and thus more time consuming than ATM cell header processing. However, this does not result in a real difference, since modern switches/routers can do this at wire speed.

Then the overall time required to switch an entire ATM cell or IP packet through a modern switch / router depends mainly on the transmission speed of the ingress and egress interfaces and the transfer characteristics of the high speed switch. Modern router implementations can forward even small VoIP packets at wire speed across high speed interfaces. Therefore the delay contribution of IP routers and ATM switches in the backbone is small compared with the overall end to end delay for Voice over Packet implementations.

In case of a rather low access link speed of 2 Mbps, a typical G.711 IP packet of 80 bytes payload (from 10ms sampling) and 40 bytes header would experience a delay at the ingress interface in the range of up to 0,5 msec.

An even smaller effect results from the waiting time for a voice packet in case of congestion in a backbone node. Let us consider the rare case that a router/switch just started forwarding a long file transfer packet (typical 1500 byte packet size – maximum Ethernet packet length) when the voice packet arrives. In this case the precedence in the TOS byte of the voice packet cannot immediately be respected and the voice packet has to wait until the file transfer packet is completely processed. Fortunately, the backbone connection speeds of routers and switches are rather fast: typically more than 155 Mbps.
The waiting time for the voice packet is then 1500 bytes payload + 40 bytes header at 155 Mbps which gives 0,08 ms.
Even for a large number of core nodes (>=10) to be passed the resulting delay contribution does not exceed the packetization delays of the mediation devices – and this even improves in case faster interconnection links are used..

Further impact to the switch / router delay is coming from filter tables handling exception rules to apply to certain destination addresses or packet flow types. The delay impact of such filter tables is hard to determine and depends heavily on the implementation of the individual switch / router devices. State of the art hardware implementations can perform all required packet processing functions at full line speed with only microseconds of delay.

Historically, Mulitprotocol Label Switch MPLS headers were defined by the IETF to achieve similar “Pseudo” layer 2 processing as in the ATM case.. These MPLS labels allow an IP router to bypass the filter tables by forwarding the packets immediately to the label defined output port in the same way as the virtual path identifier/virtual channel identifier VPI/VCI label in the ATM switch case. However, the role of MPLS in view of performance improvements is considered very small in live implementations today. MPLS has now more a role in the network engineering and Virtual Private Network VPN area.
However, the delay jitter in ATM networks is typically smaller than that of IP networks due to the relatively small fixed cell length. This effect may also result in slightly larger jitter buffer delays in an IP network.
Packet Loss is not really an issue in modern packet backbones as shown by an investigation from S.Casner at NANOG22.

Overall, it is safe to say that a switch as well as a router is able to forward packets in less than 1 msec. In modern implementations and with higher access and backbone speeds, this value can be significantly lower. This means that in practice the forwarding time difference between ATM switches with ATM cells and IP routers with RTP/UDP/IP packets is negligible compared to the delay times discussed in the encoding/decoding process above.

Conclusion

We have shown that there are no significant differences between Next Generation Networks (NGN) based on VoATM or VoIP. The differences in processing the different VoIP/VoATM protocol stacks are negligible compared to the more dominant Codec processing time. Current developments to reduce the Codec delays will further improve voice quality in Next Generation Networks.

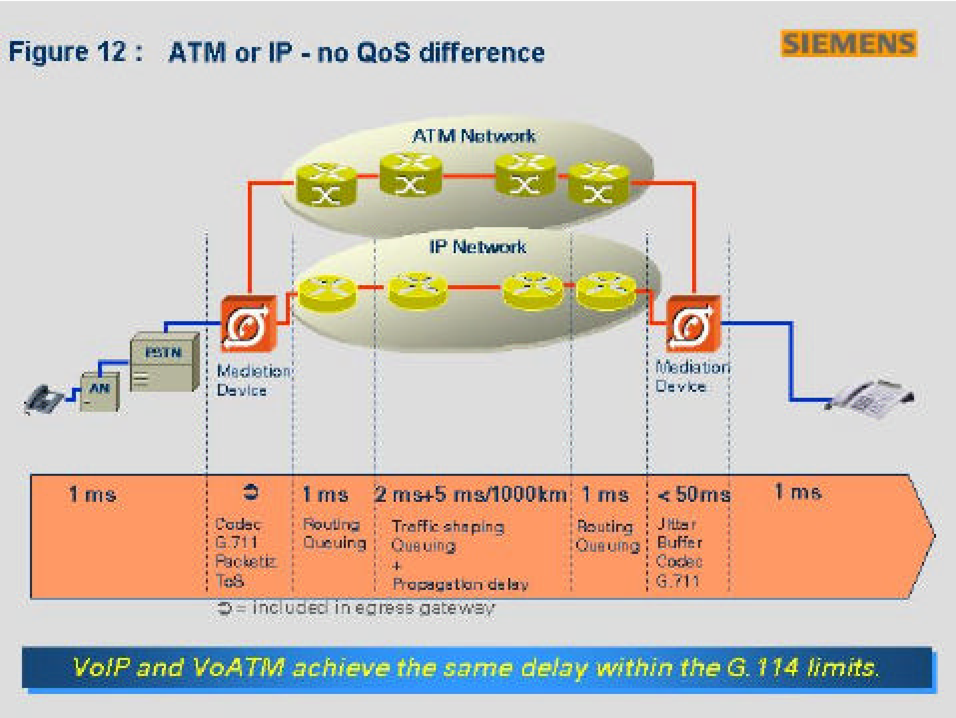


Figure 12 : ATM or IP - no QoS difference

However, non-QoS arguments like:

- VoIP offers the best platform for all-IP service integration at the end user.
- New revenues for the carrier can be generated via IP converged services.
- VoIP (not VoATM) will be required to serve enterprise VoIP solutions (e.g. IP-PBXs).
- IP is the most economical way to provide a backbone network for voice, multimedia and data services because the majority of traffic will anyhow be IP.

are more in favor of VoIP instead of VoATM.

This should give VoIP the leading edge in the race between the Voice over Packet technologies.

With modern voice encoding and network implementations, Next Generation Networks show the same Quality of Service as classic telephony networks.

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