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TITLE: Technical Issues concerning Real-Time Protocol in H.32Z Systems in ATM and Other Packet-Switched Computer Networks
PURPOSE: Discussion

# 1 Introduction

In [5] it was proposed that the framework of H.32Z would be bandwidth guaranteed networks. This would make H.32Z unsuitable to the large base of currently installed computer networks, which already have demonstrated real time applications. Indeed, H.320 based video phones already operate on such networks. In our opinion, the framework should be extended to the so called best-effort networks with the understanding that the users will have to cope with a lower quality when connecting from/to these networks. In order to utilize the packet-switched computer networks (PSCN) and the applications already developed for it, we propose that RTP (Real Time Protocol) be adopted in PSCN environment instead of H.221 multiplexing. An introduction to RTP is given in an accompanying document [1], this document addresses a number of technical issues associated with this choice.

The following points are considered:

- ISDN – PSCN gateway
- ISDN and PSCN address conversion
- Mapping of H.320 on RTP
- Mapping RTP on PSCN

# 2 Network Configuration

In this document following network types are taken into account: ISDN, Packet Switched Computer Network (PSCN), B-ISDN, and bandwidth guaranteed computer networks, like ATM and ISO-Ethernet.

Figure 1 displays the network configuration for H.32Z operation. It should be noted that the computer networks are formed from a cloud of connections, not from a straight connection. There may be several repeaters, bridges, routers and workstations along the way.

The primary interests for H.32Z systems design is the H.320 transportation in PSCNs and the relay between the ISDN and PSCN networks.

# 3 ISDN Network

On the ISDN H.320 system standard is used. This system consists of H.242/H.230 control stream, H.261 video stream, G.700 series audio stream, and data streams (MLP, LSD, and HSD) which are framed according to the H.221 standard.

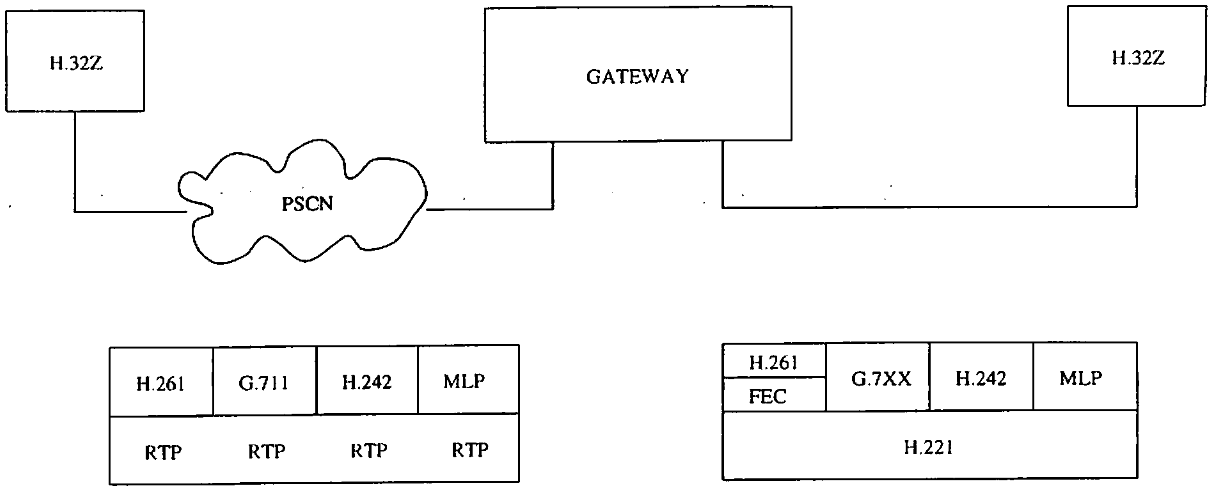


Figure 1: Network configuration

# 4 Packet-Switched Computer Networks

One of the most important design goals on the PSCN standard is that it can be used together with existing H.320 equipment and that the communications between the ISDN and PSCN is possible.

The transportation of the data packets to the receivers is left on the existing layer of protocols already present in the computer networks. The timing model and other real-time specific transportation issues are handled by RTP.

RTP specifies that each elementary stream is transferred using separate RTP connection, and that the timing is tailored for each of these separately. This means that for each H.32Z call at least 3 streams are opened: audio, video and control (see section 4.2). Note that this allows routing of data effectively: for example, if someone wishes to subscribe to audio channel only in a video conferencing application, the network will route only audio and control stream to that subscriber.

## 4.1 H.320 Call components on PSCN

### 4.1.1 H.221 Encoding

H.221 framing standard is designed specifically for ISDN environments and it is not feasible to use it in computer networks, instead a different framing should be used, we propose the use of RTP. H.221 framing will be generated when the data is traveling into ISDN network from encoded media streams.

In PSCN, the frame alignment signal (FAS) is not used and bit-rate allocation signal (BAS) and encryption control signal (ECS) (optional) are transferred in the RTP control stream using dedicated packet structure.

### 4.1.2 H.242 Encoding

We define the H.242/H.320 control stream as conference control stream (CCS) in PSCN. In call setup, the PSCN terminal uses CCS mechanisms to connect with the other PSCN terminal or the gateway.

The control messages can be defined in the RTP control channel (RTCP) (see section 4.2). The detailed definitions of the end-to-end BAS and ECS codes and of the call setup mechanism is for further study.

### 4.1.3 Audio Encoding

G.700 series audio streams can be easily encoded into RTP packets, there are several reference drafts and applications within the IETF's audio-video transport work group [3] which could be used as a basis for study.

### 4.1.4 H.261 Encoding

H.261 video stream can be easily packed into RTP data packets. The exact size of the packets is left for further study. There is a proposition of how to packetize H.261 streams into RTP [2], which could be used as a base for the official recommendation.

### 4.1.5 Data Encoding

The MLP standard defined in ISDN can be utilized in PSCN also. In PSCN a separate virtual connection is opened for each data application. The transfer capacity of this connection is divided among the MLP connections.

It should be noted, that there may be other alternatives in this case. For example, some data applications may not require RTP protocol at all. Also, in a PSCN–PSCN call the MLP multiplexing is somewhat artificial, it might be better that the data applications could talk directly using separate connections. However, MLP is required to maintain compatibility to the ISDN terminals that already utilize data transfer.

## 4.2 H.320 System in the PSCN

For best performance and minimal overhead in computer networks the H.221 framing should be removed from the H.320 system data. Splitting the data into several streams improves routing and reduces network overhead especially in multi cast transmission.

Each RTP connection is accompanied by a control connection (RTCP). To avoid 6 connections minimum for each call (video, audio, CCS and their RTP control connections) the CCS connection is transferred in the RTCP channel of the audio stream.

# 5 ISDN – PSCN Gateway

The needed functionality in gateways are:

- Address resolution and conversion
- Flow control
- Data conversion ISDN – PSCN
- Possible stream re-encoding
- Synchronization restoration

## 5.1 Address resolution and conversion

The gateway must do address conversion between the ISDN and PSCN. There are several possible address formats on the PSCN side, which makes the address conversion complex issue. Also the existing H.320 equipment do not support a way to transfer the PSCN address from the ISDN side. As a near-term solution the working scheme could be similar to ISDN multi point control unit (MCU) where both participants take separate connections to the gateway which then binds the connections to each other. On the long term H.320 BAS -codes could be extended to cover the necessary data structures. There is no problem on the PSCN to ISDN connections because the standard is not yet finished and can be extended easily accordingly.

## 5.2 Flow control

Flow control takes care of the underflow and overflow situations on the computer network side. Short-time congestion in PSCN are likely to occur suddenly and unpredictably and they are resolved equally fast, whereas permanent increase in PSCN traffic manifests frequent problems in network throughput. Short term and long term congestion are dealt with separately.

Short-term congestion is handled with suitable buffers in the gateway, whose average fullness is low. In overflow situation the excess data is stored in the buffer which is cleared when congestion has resolved.

Long-term congestion is detected when the average buffer occupation (e.g. delay) increases. End-to-end signaling between terminals is utilized to reduce the data rate: the basic principle of action in this congestion type is to first reduce the video data that goes into PSCN using a BAS code, then reduce traffic in the data channel (which is normally transfered using a protocol that supports correction of transmission errors, e.g. MLP) and finally reduce audio data (see section 5.2.1 below).

### 5.2.1 ISDN → PSCN

If there is congestion on the computer network side when the data is coming from ISDN connection the gateway may change encoding parameters accordingly or drop data until congestion resolves and then request full update (i.e. intra frames). Depending on the computer network data may be priorized in order to suggest that the most important data will be transported first.

More specifically, the gateway first requests the H.320 terminal to switch the line rate from 2B to 1B using a BAS code. If this is not enough, a possibility to temporarily freeze the video signal should be investigated. If these measures are not enough, then we end up dropping data (starting from the data stream). Data is lost only if the line rate is 1B and the gateway buffers are full. A fast update BAS command is issued after data has been lost.

The PSCN terminal may also reduce the data it is generating. It can independently of the gateway observe network load and reduce its own traffic load without being instructed to do so by the gateway. Proper CCS message is for further study.

### 5.2.2 PSCN → ISDN

When the data is traveling from the PSCN to the ISDN, the overflow should never occur if the parameter negotiation works correctly, except in cases when a burst of late packets are received after a congestion in PSCN. These packets should be dropped. Underflow can be fixed by sending enough padding as H.261 FEC -data to satisfy the ISDN data rate.

## 5.3 Data conversion ISDN – PSCN

In data conversion, the gateway transforms the continuous stream of data from the ISDN to packets in computer network and vice versa. This mapping depends on whether the system stream is split on PSCN or not. The H.221 framing must be generated for PSCN to ISDN data.

## 5.4 Possible stream re-encoding

Depending on computer network congestion, the gateway might renegotiate the transfer parameters and change encoding accordingly. Recoding requires a powerful gateway-hardware.

It should be noted that H.320 allows to change the capability set during a call and defines other similar commands. These services should be offered by the CCS in PSCN–PSCN and PSCN–Gateway connections.

## 5.5 Synchronization restoration

The synchronization restoration must be done in both ways. On the PSCN side the timing model is based on RTP timing model and network clock. When the data is going from ISDN to PSCN, video and audio are synchronized within a H.221 frame. The gateway has to add appropriate time stamps to the audio and video RTP packets.

When the data is traveling from PSCN to ISDN the sender appends the time stamps to the packets, which then are collected (using suitable buffering) by the gateway so that H.221 framing with correct synchronization can be created.

## 5.6 Types of gateways

Due to the complex nature of the gateway, two different kinds gateways are proposed: enhanced gateway and simple gateway. The two different gateways differ in the complexity of the implementation which gives them a different set of flow control capabilities.

### 5.6.1 Enhanced Gateway

Enhanced gateway is similar to the multi point control unit (MCU) in the ISDN network. This gateway may do media encoding conversions according to the network congestion and may renegotiate the encoding parameters accordingly. It does not utilize data dropping as a flow control method.

### 5.6.2 Simple Gateway

This gateway type does not have capabilities to recode the data. Therefore, the simple gateway does just minimal protocol conversion required for converting the the data from ISDN to PSCN. The parameters are statically negotiated at the beginning of the conversation. Flow control is established changing the coding parameters during a call, and data dropping is used as a last resort.

# 6 RTP mapping in the packet-switched computer networks

RTP is independent from the underlying transport and networks layers and supporting different networks is one the main design goals of RTP. RTP protocol packets can be efficiently put on the the top of variety of network protocols. Currently the UDP/IP protocol stack is the most studied environment as a underlying protocol for RTP. Currently there are implementations of RTP over IP and ST networks.

## 6.1 RTP Mapping in the ATM networks

RTP can be easily mapped on the top of ATM AAL type 5, which is the most popular adaptation layer in the computer environment. No difficulties should arise in mapping RTP on AAL1 either.

### 6.1.1 RTP on AAL 5

If AAL5 is used in the ATM LAN environment the service provided for upper layers is very similar to those of IP. Here ATM is considered as one of the network technologies used in the LAN environment.

Here are the cases based on current recommendations from ATM-Forum and IETF on how ATM connections are managed.

**Case 1:** Permanent Virtual Circuits

- RTP header together with multiplexing transport level multiplexing (port number) can be put as a payload of a AAL5 PDU.
- In the case of multi cast address, original address cannot be decoded from the ATM cell header and some kind of of-band signaling is required.
- Overhead of this method is 2 octet port number and a minimum of 12 octet for RTP header per 1 AAL5 PDU.

**Case 2:** Permanent Virtual Circuits with LLC-Encapsulation

- RTP header and UDP/IP headers can be encapsulated using the LLC-encapsulation. [4]
- This method provides a means of resolving the address of the receiver were multi cast address is used, because the original IP address is transported.
- Overhead of this method is 8 octet LLC-header, 28 octet UDP/IP headers and a minimum of 12 octet for RTP header per 1 AAL5 PDU.

**Case 3:** Switched Virtual Circuits

- Here are several possible ways to transport the stream: LLC encapsulation using VC instead of header, ATM-Forum LAN emulation or VC encoding of UDP port number.
- Overhead for this method is depending on the exact method used, in the best case only a minimum of 12 octet for RTP header per 1 AAL5 PDU has to be transported.

### 6.1.2 RTP over AAL1

For further study.

## References

[1] Tsokkinen M. "Real-Time Transport Protocol in H.32Z Systems in ATM and Other Packet-Switched Computer Networks", AVC-649, June 1994

[2] Turletti T., Huitema C. "Packetization of H.261 Video Streams", Internet draft, December 1993

[3] Schulzrinne H. "Media Encodings", Internet draft, September 1993

[4] Heinnen J. "Multiprotocol Encapsulation over ATM Adaptation Layer 5", RFC 1483, Telecom Finland, July 1993

[5] AVC-627, General Approach to H.32Z (UK)