



INTERNATIONAL TELECOMMUNICATION UNION

CCITT

G.764

THE INTERNATIONAL
TELEGRAPH AND TELEPHONE
CONSULTATIVE COMMITTEE

**GENERAL ASPECTS OF DIGITAL
TRANSMISSION SYSTEMS;
TERMINAL EQUIPMENTS**

**VOICE PACKETIZATION –
PACKETIZED VOICE PROTOCOLS**

Recommendation G.764



Geneva, 1990



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FOREWORD

The CCITT (the International Telegraph and Telephone Consultative Committee) is a permanent organ of the International Telecommunication Union (ITU). CCITT is responsible for studying technical, operating and tariff questions and issuing Recommendations on them with a view to standardizing telecommunications on a worldwide basis.

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Recommendation G.764 was prepared by Study Group XV and was approved under the Resolution No. 2 procedure on the 14 of December 1990.

CCITT NOTE

In this Recommendation, the expression "Administration" is used for conciseness to indicate both a telecommunication Administration and a recognized private operating agency.

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Recommendation G.764

VOICE PACKETIZATION – PACKETIZED VOICE PROTOCOLS

1 Introduction

This Recommendation defines a packet voice protocol for speech packetization in permanent virtual circuit applications. The principal application of the packetized voice protocol (PVP) is at the primary rate and in fractional primary rate applications.

The protocol defines formats and procedures for the transport of voice information and channel associated signalling over a wideband packet network.

The Recommendation accommodates additional future types including optional internetworking capabilities with the digital cellular radio networking applications currently under development. The extension of this Recommendation for baseband facsimile traffic is currently under study.

This Recommendation does not address performance issues.

This Recommendation does not address methods for coding speech samples, although particular recommended coding algorithms are specified in the protocol (e.g. the algorithms in Recommendation G.726). In particular, the Recommendation allows dynamic bandwidth allocation and graceful congestion control when the speech samples are coded with embedded algorithms such as those specified in Recommendation G.727.

This Recommendation does not address the following issues:

- 1) services based on the interface;
- 2) implementation techniques;
- 3) performance guidelines relative to the use of packetized speech;
- 4) equipment aspects;
- 5) trunk signalling, link establishment and call establishment procedures for switched virtual circuits;
- 6) data only issues, combined data/voice issues and frame relay issues;
- 7) speech packetization in Asynchronous Transfer Mode (ATM) (B-ISDN) systems.

2 Overview

This Recommendation contains the specification of a protocol for packetized speech (PVP). PVP defines formats and procedures for the transport of voice information and channel associated signalling over a packet network.

Before packetization, the input speech samples may be coded at the originating endpoint of the transmitting side by one of the coding methods indicated in this document. The stream of coded speech is transformed into packets with the format specified in this document. The samples are collected over a period of 16 ms and divided into blocks of 128 bits each. Silent intervals may be removed. The blocks are arranged to facilitate block dropping.

Periods of activity and inactivity are respectively called “bursts” and “gaps”. It is not necessary to transmit packets during gaps.

The terminating end at the receiving side reconstructs a continuous stream of speech from the incoming packets using the information in the packet header. The build-out delay procedure described in this document compensates for the variable delay that packets may experience within the network. Packets that arrive before their scheduled play-out time are placed in the proper sequence in a packet queue. Packets that arrive after their scheduled play-out time are discarded. The voice packet header contains information about the level of noise that was measured by the originating endpoint. The terminating endpoint uses this information to play out a matching noise level.

An additional feature of PVP is the ability to drop blocks from a packet as a congestion control mechanism. The n -th block consists of the n -th bit from each sample collected during the sampling interval. The packet header indicates the number of droppable blocks contained in the packet. Congested nodes may use this information to drop the least significant block from packets to abate the congested state.

The signalling associated with each voice connection shall be transported in signalling packets. Signalling packets shall be sent separately on a different logical channel. Transport of the signalling information requires a set of procedures, similar to those for voice transport, which are described in this document.

Note – In a national network, the time stamp (TS) and the build-out procedures of §§ 5.1.1, 5.2 and 6.3 may be replaced by a fixed-delay for the first packet. At the originating end, the TS will be set to zero. In an intermediate node, the TS field will not be updated. However, the build-out procedure will be used always on network-network and user-network interfaces.

3 Formats

3.1 Physical layer

For operations at 1536 kbit/s or 1920 kbit/s, the electrical characteristics and formats of the interface are those defined in Recommendations G.703, G.704 and I.431 for the primary rates of 1544 kbit/s and 2048 kbit/s, respectively. The packetized signal consists of one digital stream sent over conventional primary rate facilities. Hybrid situations containing one or more $N \times 64$ kbit/s packet streams and M conventional 64 kbit/s channels are also considered.

3.1.1 Bit inversion

For primary rate applications that require code restrictions that maintain one's density, bit inversion is necessary so that the combined result of bit stuffing and bit inversion is to prevent the all 0 octet and to satisfy the one's density requirements of restricted DS 1 facilities.

3.1.2 Order of transmission

Bit 1 is the least significant bit (LSB) and is transmitted first. Bit 8 is the most significant bit (MSB) and is transmitted last.

3.2 Link layer

The link layer of PVP uses a similar approach to Recommendation Q.921/I.441 with the additions indicated in this Recommendation. Frames that transport voice and frames that transport channel associated signalling are assigned different layer 2 addresses, i.e. they are carried on two separate logical links. This, together with the use of a different unnumbered frame type for each type of traffic, provides an additional measure of security to protect from the misrouting of signalling information.

3.2.1 Address field

The address field is two octets in length with the first bit of each defined as an extension bit and bit 2 of octet 1 defined as the command/response (C/R) bit. The 13 bits that remain are concatenated to form a single data link connection identifier (DLCI). Address assignment starts with 128 and ends with 8063. Layer 2 addresses are already assigned and the implementation starts from the DLCI_ASSIGNED state.

3.2.2 *Command/response bit*

The C/R bit (bit 2 of octet 1) is set to 0.

3.2.3 *Frame types*

The following two frame types are allowed in PVP.

3.2.3.1 *Unnumbered information frames*

When a layer 3 or management entity requests unacknowledged information transfer, the unnumbered information (UI) command is used to send information to its peer without affecting data link layer variables. UI command frames do not carry a sequence number and therefore, the UI frame may be lost without notification.

The control field for the UI command frame is a single octet in length. The format and encoding are the same as specified in Recommendation Q.921/I.441. The UI frame is used to transport channel associated signalling.

3.2.3.2 *Unnumbered information with header check frame*

The unnumbered information with header check (UIH) frame has the same applications as the UI frame. The difference between the two is that the cyclic redundancy check (CRC) sequence is derived over the frame and packet headers (the first 8 octets excluding flags) rather than over the entire frame. The check sequence fills the last two octets of the UIH frame.

The control field of the UIH is a single octet in length and is shown in Figure 1/G.764.

8	7	6	5	4	3	2	1	Bit number
1	1	1	P	1	1	1	1	

FIGURE 1/G.764

Control field of the UIH frame

The UIH frame is used to transport voice (see Note).

Note – The CRC check of the UIH protects 8 octets which contain the address field (to ensure correct delivery), the control field (to guarantee the validity of the frame type) and the layer 3 header. It does not protect the voice information because voice traffic is more sensitive to delay due to retransmission than to bit errors and because this allows reduction of the voice information by block dropping under congestion without recalculating the CRC check. As a consequence, the test for invalid frames in § 3.2.7 uses the minimum frame length of 10 octets.

3.2.4 *Poll bit*

The poll bit (P) is bit 5 of the UI/UIH frame control field. The P bit shall be set to 0.

3.2.5 Check sequence

The check sequence (CS) algorithm is the same as that described in ISO Recommendation ISO-3309. The CS field shall be a 16-bit sequence. It shall be the ones complement of the sum (modulo 2) of:

- 1) The remainder of $(x \text{ raised to } k \text{ power}) (x^{15} + x^{14} + x^{13} + x^{12} + x^{11} + x^{10} + x^9 + x^8 + x^7 + x^6 + x^5 + x^4 + x^3 + x^2 + x^1 + 1)$ divided (modulo 2) by the generator polynomial $x^{16} + x^{12} + x^5 + 1$, where k is the number of bits in the frame existing between, but not including, the final bit of the opening flag and the first bit of the first octet of the non-droppable octets for the header check sequence (HCS) or the first bit of the check sequence for the frame check sequence (FCS), excluding bits inserted for transparency, and
- 2) the remainder of the division (modulo 2) by the generator polynomial $x^{16} + x^{12} + x^5 + 1$, of the product of x^{16} by the content of the frame existing between, but not including, the final bit of the opening flag and the first bit of the first octet of the non-droppable octets for the HCS or the first bit of the CS for the FCS, excluding bits inserted for transparency.

As a typical implementation at the transmitter, the initial content of the register of the device computing the remainder of the division is preset to all ones and is then modified by division by the generator polynomial (as described above) of the address, control and appropriate portion of the information fields; the ones complement of the resulting remainder is transmitted as the 16-bit CS.

As a typical implementation at the receiver, the initial content of the register of the device computing the remainder is preset to all ones. The final remainder after multiplication by x^{16} and then division (modulo 2) by the generator polynomial $x^{16} + x^{12} + x^5 + 1$ of the serial incoming protected bits and the CS, will be 0001110100001111 (x^{15} through x^0 , respectively) in the absence of transmission errors.

3.2.6 Frame abort

Receipt of seven or more contiguous 1 bits shall be interpreted as a frame abort, and the link layer entity shall ignore the frame currently being received. A frame following an abort must begin with an opening flag.

3.2.7 Invalid UI/UIH frames for PVP

For the purposes of PVP, an invalid UI/UIH frame is a frame which:

- 1) is not properly bounded by two flags; or
- 2) has fewer than 10 octets between flags; or
- 3) has greater than 490 octets between flags; or
- 4) does not consist of an integral number of octets prior to zero bit insertion or following zero bit extractions; or
- 5) contains a FCS error.

Invalid frames shall be discarded without notification to the sender. No action is taken as the result of that frame.

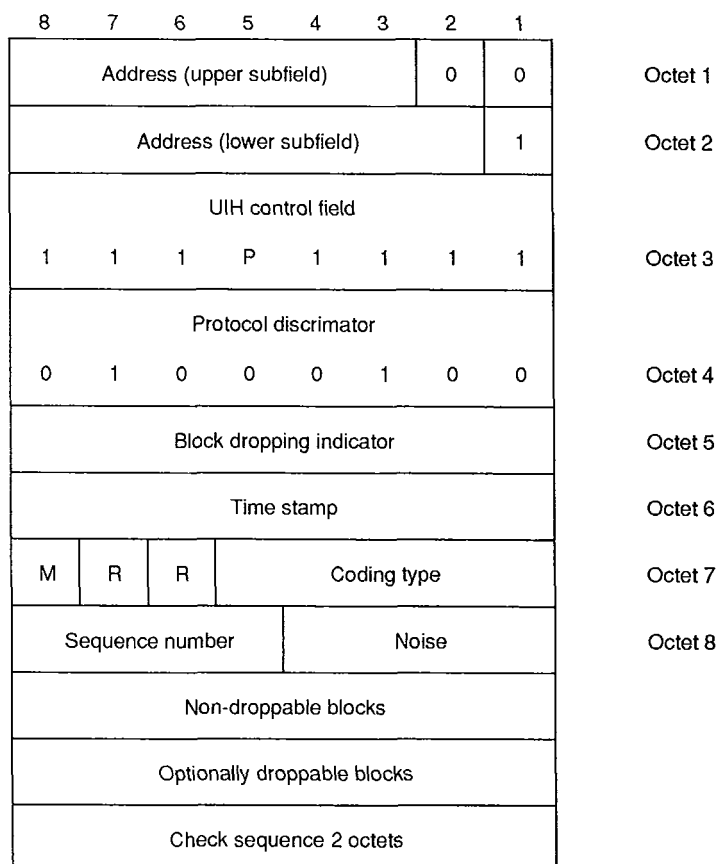
3.3 Packet layer

The packet layer procedures apply to the information transfer phase only. Call control procedures are outside the scope of this Recommendation.

3.3.1 Voice packet format

The format of voice packets is shown in Figure 2/G.764 within the UIH voice frame.

Note – The reserved bits are set to 0 at the originating endpoint. They shall be ignored at the terminating endpoint. They shall not be used for testing or maintenance purposes in anticipation of possible future uses.



M More bit
P Poll bit = 0 (see § 3.2.4)
R Reserved for future use

FIGURE 2/G.764
UIH voice frame format

3.3.1.1 Protocol discriminator

The protocol discriminator (PD) field is the first octet of the packet header (octet 4, the frame in Figure 2/G.764). Its value for the PVP is given in Figure 3/G.764.

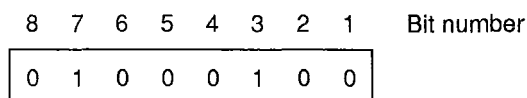


FIGURE 3/G.764
Protocol discriminator for the PVP