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TITLE : **Comments on H.225.0** (Media Stream Packetization and Synchronization for Visual Telephone Systems on Non-Guaranteed Quality of Service LANs)

DATE: December 13, 1995

Telecommunication Standardization Sector
English
(TSS)

Determined H.225.0
Original:

SG15/WP1

November 13-24, 1995

Question 3/15

STUDY GROUP 15 CONTRIBUTION

Source: SG15/WP1

Title: H.225.0 Media Stream Packetization and Synchronization for Visual Telephone Systems on Non-Guaranteed Quality of Service LANs

Date: November 24, 1995

Summary: This recommendation describes a method for combining audio, video, data, and control information on a non-guaranteed quality of service LAN to provide conversational services in H.323 equipment. Included are such topics as audio coding, video coding, control and signaling PDUs, and methods for providing improved quality of service in this environment.

Notes on reading: ***{Editors notes are generally in braces}*** while underlined text is new. ~~The deleted-text-has-strikethrough-font.~~ Please ignore all references to other sections both inside and outside this document; these will be updated in the final version.

This version (Plenary H.225.0) reflects changes from the version input to the November 13-24 SG15/WP1 meeting, and reflects changes agreed to at that meeting, including at the SG15 Plenary.

The following major technical directions constitute an integral part of the determination being made at this meeting:

a)Overall call model

b)Use of RTP by INCLUSION AS REQUIRED BY THE SG15 PLENARY.

c)Use of Q.931 as described in H.225.0

d)Elements such as gatekeeper, terminal, MCU, gateway, MC, MP

e)separation of terminal/gatekeeper admission signaling, call signaling, and H.245 signaling onto different channels

f)Use of H.245

g)gatekeeper control over call model in use

h)optional nature of gatekeeper, gateway, MCU, MP, MC

i)support for multicast audio/video with centralized control/data

j)an optional, H.245 capability based mode in which audio and video are mixed in the same packet will be added. Note that only terminals with the optional capabilities will be able to make use of this mode.

j)an optional H.245 capability based mode that allows for auido and video re-transmission will be added. Note that only terminals with the optional capabilities will be able to make use of this mode.

j)gatekeepers are made aware in the ARQ message that a terminal is or is not a gateway, and is or is not MC capable.

Global editorialchanges agreed to include:

a)BYTES To OCTETS
b)units of 100 bits in ASN.1
c)H.245 object identifier instead of YearOfSpec field
d)Proprietary extension as in H.245 on all messages(which is exactly like the extension method of H.221)
e)editor to consider 4 letter command abbreviations.
f)
f)The use of the terms "PDU" and "message" will be rationalized.
g)The use of the terms "dynamic" and "transient" port will be rationalized.

Minor Issues:
1)Add maximum packet rate to cap exchange? (Did we resolve this??)

INTERNATIONAL TELECOMMUNICATION UNION

ITU-T

TELECOMMUNICATION
STANDARDIZATION SECTOR
OF ITU

DRAFT H.225.0

(November 24, 1995)
Determined November 1995

LINE TRANSMISSION OF NON-TELEPHONE SIGNALS

Media Stream Packetization and Synchronization on Non-Guaranteed Quality of Service LANs

DRAFT ITU-T Recommendation H.225.0

FOREWORD

The ITU Telecommunication Standardization Sector (ITU-T) is a permanent organ of the International Telecommunication Union. The ITU-T is responsible for studying technical, operating and tariff questions and issuing Recommendations on them with a view to standardizing telecommunications on a worldwide basis.

The World Telecommunication Standardization Conference (WTSC), which meets every four years, established the topics for study by the ITU-T Study Groups which, in their turn, produce Recommendations on these topics.

The approval of Recommendations by the Members of the ITU-T is covered by the procedure laid down in WTSC Resolution No. 1 (Helsinki, March 1-12, 1993)

ITU-T Recommendation H.225.0 was prepared by the ITU-T Study Group 15 (1993-1996) and was approved under the WTSC Resolution No. 1 procedure on the xxth of xxxx 199x.

NOTE

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

SUMMARY

This Recommendation covers the technical requirements for narrow-band visual telephone services defined in H.200/AV.120-Series Recommendations, in those situations where the transmission path includes one or more Local Area Networks (LAN), each of which is configured and managed to provide a non-guaranteed Quality of Service (QoS) which is not equivalent to that of N-ISDN such that additional protection or recovery mechanisms beyond those mandated by Rec. H.320 need be provided in the terminals. It is noted that Recommendation H.322 addresses the use of some other LANs which are able to provide the underlying performance not assumed by the H.323/H.225.0 Recommendations.

This recommendation describes how audio, video, data, and control information on a non-guaranteed quality of service LAN can be managed to provide conversational services in H.323 equipment

Media Stream Packetization and Synchronization on Non-Guaranteed Quality of Service LANs

(Place, 199x)

The ITU,

considering
the widespread adoption of and the increasing use of the H.320 Recommendation for videophony and videoconferencing services over networks conforming to the N-ISDN characteristics specified in the I series Recommendations,

appreciating
the desirability and benefits of enabling the above services to be carried, wholly or in part, over Local Area Networks while also maintaining the capability of interworking with H.320 terminals

and noting
the characteristics and performances of the many types of Local Area Network which are of potential interest

recommends
that systems and equipment meeting the requirements of the H.322 or H.323 Recommendations are utilized to provide these facilities.

1. Scope

This recommendation describes the means by which audio, video, data, and control are associated, coded, and packetized for transport between H.323 terminals on a non-guaranteed quality of service LAN, or between H.323 terminals and an H.323 gateway, which in turn may be connected to H.320, H.324, or H.310/H.321 terminals on N-ISDN, GSTN, or B-ISDN respectively. This gateway, terminal descriptions, and procedures are described in H.323 while H.225.0 covers protocols and message formats. Communication via an H.323 gateway to an H.322 gateway for guaranteed quality of service (QOS) LANs and thus to H.322 endpoints is also possible.

H.225.0 is intended to operate over a variety of different LANs, including IEEE 802.3, Token Ring, etc.. Thus, H.225.0 is defined as being above the transport layer such as TCP/IP/UDP, SPX/IPX, etc.. Specific profiles for particular transport protocol suites are included in Annex A of this recommendation. ***Thus, the scope of H.225.0 communication is between H.323 terminals and H.323 gateways on the same LAN, using the same transport protocol.*** This LAN may be a single segment or ring, or it logically could be an enterprise data network comprising multiple LANs bridged or routed to create one interconnected network. It should be emphasized that operation of H.323 terminals over the entire Internet, or even several connected LANs may result in poor performance. The possible means by which quality of service might be assured on this LAN network, or on the Internet in general is beyond the scope of this recommendation. However, H.225.0 provides a means for the user of H.323 equipment to determine that quality problems are the result of LAN congestion, as well as procedures for corrective actions. It is also noted that the use of multiple H.323 gateways connected over the public ISDN network is a straightforward method for increasing quality of service.

H.323/H.225.0 are intended to extend H.320/H.221 conferences/connections onto the non-guaranteed QOS LAN environment. As such the primary conference model[1] is one with size in the range of a few participants to a few thousand, as opposed to large-scale broadcast operations, with strong admission control, and tight conference control. This is in contrast to various IETF (Internet Engineering Task Force) protocols that focus on very large conferences with weak admission and conference control.

H.225.0 makes use RTP/RTCP(Real Time Protocol/Real Time Control Protocol) for media steam packetization and synchronization for all underlying LANs as provided for in Annex ZZZ.. Please note that the usage of RTP/RTCP as specified in H.225.0 is not tied in any way to the usage of TCP/IP/UDP. H.225.0 assumes a call model where initial signaling on a non-RTP LAN port is used for call establishment and capability negotiation (see H.323 and H.245), followed by the establishment of one or more RTP/RTCP like connections. H.225.0 contains details on the usage of RTP/RTCP.

All references to RTP, RTCP and RTP-P in this recommendation refer to the relevant Annexes (X,Y,Z).

Since H.225.0 supports only single connection operation in the H.221 sense (i.e. any "channel" structure provided by the LAN using transport network port numbers is analogous to the "channels" of H.221, such as audio, video, LSD, etc. Thus, on the LAN side, for example, there are only single "connection" calls with a maximum rate limited to 128 kbps, not 2 * 64 kbps fixed rate calls). Another example has single "connection" LAN calls with a maximum rate limited to

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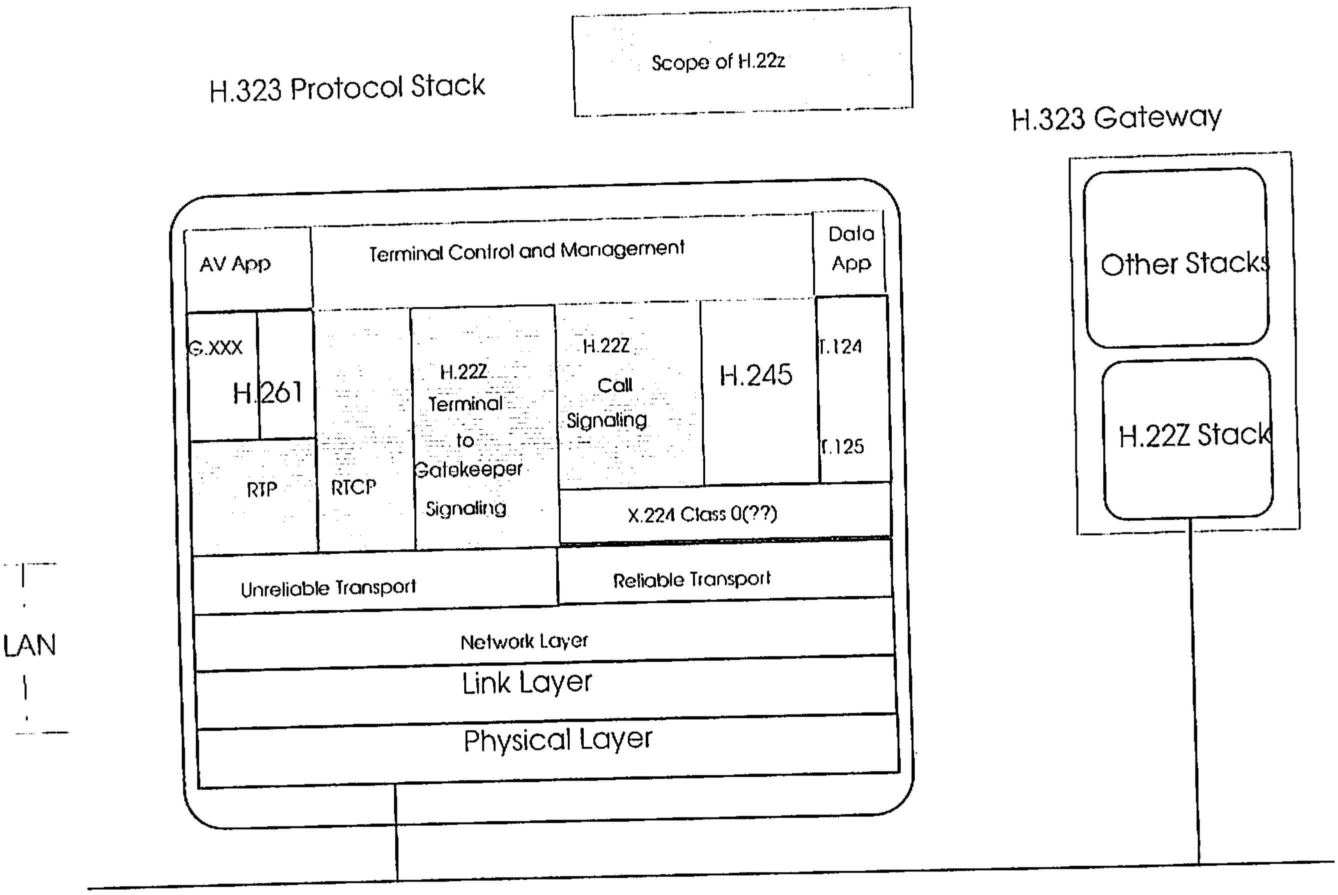
[1] An optional broadcast only conference model is also supported; of necessily the broadcast model does not provide tight admissions or conference control.

384 kbps interworking with 6 * 64 kbps on the WAN side[2]. The primary rationale is to put complexity in the gateway rather than the terminal and to avoid extending onto the LAN features of H.320 that are tightly tied to ISDN unless this is absolutely necessary.

In general, H.323 terminals are not aware directly of the H.320 transfer rate while interworking through an H.323 gateway; instead, the gateway using H.245 FlowControl messages to limit the data rate on each logical channel in use to that allowed by the H.221 multiplex. The gateway may allow the LAN side video rates to substantially underrun the WAN side rates (or the reverse) though the usage of a rate reducing function and H.261 fill frames; the details of such operation are beyond the scope of H.323/H.225.0. Note that the H.323 terminal is indirectly aware of the H.320 transfer rates via the video maximum bit rate fields in H.245,and the data maximum bit rate fields.

H.225.0 is designed so that, with an H.323 gateway, interoperability with H.320(1990), H.320(1993), and H.320(1996) terminals is possible. However, some features of H.225.0 may be directed toward allowing enhanced operations with future versions of H.320. It is also possible that the quality of service on the H.320 side may vary based on the features and capabilities of the H.323 gateway.



[2]Note that video and data rates on the LAN side must match the video and data rates in the WAN side H.320 multiplex; the audio and control rates are not required to match. Also note that the LAN rate may under-run the WAN rate for either/both video or/and data, but it cannot exceed the maximum amount that fits into the WAN side multiplex.

Figure 1/H.225.0
Scope of H.225.0

The general approach of H.225.0 is to provide a means of synchronizing packets that makes use of the underlying LAN/transport facilities. H.225.0 does not require all media and control to be mixed into a single stream, which is then packetized. The framing mechanisms of H.221 are not utilized for the following reasons:

- Not using H.221 allows each media to receive different error treatment as appropriate.
- H.221 is relatively sensitive to the loss of random groups of bits; packetization allows greater robustness in the LAN environment.
- H.245 and ~~Q.931~~ H.225.0 can be sent over reliable links provided by the LAN.
- Flexiblity and power of H.245 as compared to H.242

2. References

The following ITU-T Recommendations and other references contain provisions which, through reference in this text, constitute provisions of this Recommendation. At the time of publication, the editions indicated were valid. All Recommendations and other references are subject to revision; all users of this Recommendation are therefore encouraged to investigate the possibility of applying the most recent edition of the Recommendations and other references listed below. A list of the currently valid ITU-T Recommendations is regularly published.

{Editors Notes:

As per the decision of the SG15 Plenary, RTP will be included as three annexs. (I suppose the three Annexes are the RTP, The Profile and the H.261 Payload. What about the other payloads. We have plenty of optional audio and video options. I assume we need a payload as Annex for each of them, otherwise there is no way to use them.) There will be no reference to an IETF document in H.225.0 in a normative fashion; use of such references in non-normative appendicies are allowd}

1. CCITT Recommendation G.711, *Pulse Code Modulation of 3kHz Audio Channels*, November 1988
2. CCITT Recommendation G.722, *7kHz Audio Coding within a 64 kbit/s Channel*, November 1988}
3. CCITT Recommendation G.728, *Coding of Speech at 16 kbit/s Using Low-delay Code Excited Linear Prediction (LD-CELP)*, May 1992
4. ITU-T Recommendation H.221(1993)[3], *Frame Structure for a 64 to 1920 kbit/s channel in audiovisual teleservices.*

[3]Previously CCITT Recommendation

5. ITU-T Recommendation H.230(1993)[4], *Frame Synchronous Control and Indication Signals for Audiovisual Systems*, December 1990

6. ITU-T Recommendation H.233(1993) *Confidentiality System for Audiovisual Services*, March 1993

7. ITU-T Recommendation H.242(1993)[5], *System for Establishing Communication Between Audiovisual Terminals Using Digital Channels up to 2 Mbit/s.*

8. ITU-T Recommendation H.243(1993), *Procedures for Establishing Communication Between Three or More Audiovisual Terminals using Channels up to 2 Mbit/sec.*

9. ITY-T Recommendation H.320(1993)[6], *Narrowband Visual Telephone Systems and Terminal Equipment.*

10. ITU-T Recommendation T.122(1993), *Multipoint Communication Service for Audiographics and Audiovisual Conferencing Service Definition*

11. ITU-T Recommendation T.123(1993), *Protocol Stacks for Audiovisual and Audiographic Teleconference Applications.*

12. ITU-T Recommendation T.125(1994), *Multipoint Communication Service Protocol Specification.*

13. H.324.

14. H.321

15. H.322

16. H.310

20. Q.931 add reference

21. Q.932 add reference

3. Definitions

Internet: Any system for the connection of LANs at the network layer. The operation of such systems is beyond the scope of H.225.0.

LAN: In the context of this document, a LAN(Local Area Network) is a mechanism for the switching of packet data over a limited area, including the physical andlink but not the transport layers. Thus, the "LAN" is beyond the scope of H.225.0 for the most part, and H.225.0 is intended for use with a variety of "LANs."

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[4] Previously CCITT Recommendation
[5] Previously CCITT Recommendation
[6] Previously CCITT Recommendation

Transport Address~~Channel~~: The combination of a Transport Network Address and a Port number.~~—The channel may be multicast or unicast.—In H.225.0 each media uses a different LAN channel.—Not to be confused with an ISDN channel.~~

~~Transport~~ Network Address: The addressing entity by which a terminal is referred to externally. An example is the IP address.

Transport Port Number: a value that when associated with a Transport Network Address makes up a Transport Address~~channel~~. A Port Number may be specified in various ways for differnt transport protocolss, but it is that entity which allows sub-addressing within a Transport Network Address. The Transport Port number may provide guaranteeed delivery or non-guaranteeded delivery of data. Not to be confused with an MCU port.

Well Known Port - Need for a definition

Multipoint Control Unit (MCU): a multi-port device, by means of which two or more audiovisual terminals may intercommunicate in a conference call; See recommendations H.231 and H.243 for details.~~—The MCU is used to connect terminals over a wide area, but the H.323 gateway may optionally act as an MCU for either LAN or WAN terminals.~~

See H.323 for definitions of ~~domain~~ zone and super-zone ~~sub-domain~~, and MC.

4. Conventions

In this document, "shall" refers to a mandatory requirement, while "should" refers to a recommended but optional feature or procedure.

5. Abbreviations

BAS	Bit rate Allocation Signal
CIF	Common Intermediate Format
ECS	Encryption Control Signal
FAS	Frame Alignment Signal
FAW	Frame Alignment Word
GOB	Group of Blocks
H-MLP	High speed Multi-Layer Protocol
HSD	High Speed Data
IETF	Internet Engineering Task Force
IP	Add
LAN	Local Area Network
LD-CELP	Low Delay - Code Excited Linear Prediction
LSB	Least Significant Bit

LSD	Low Speed Data
MBE	Multi-Byte Extension
MCC	Multipoint Command Conference
MCN	Multipoint Command Negating
MCS	Multipoint Command Data Symmetrical Data Transmission
MF	MultiFrame
MLP	Multi-Layer Protocol
MPI	Minimum Picture Interval
MSB	Most Significant Bit
NS	Non-Standard
PCM	Pulse Code Modulation
QCIF	Quarter Common Intermediate Format
QOS	Quality of Service
RTP	Real Time Protocol
RTCP	Real Time Control Protocol
SBE	Single Byte Extension
SC	Service Channel
SCM	Selected Communications Mode
SMF	Sub-MultiFrame
SP	Still Picture
TCP	Transport Control ProtocolAdd
VCF	Video Command "Freeze Picture Request"
VCU	Video Command "Fast Update Request"
UDP	Unicast Data ProtocolAdd

6. FramingPacketization and Synchronization Mechanism

6.1. General Approach

The general approach is to send an optional admission request, followed by an initial Setup Message on a guaranteed channel address returned in the admission confirmation message, or which was known to the calling terminal to a terminal (which may also be an H.323 gateway) or to a gatekeeper. As a result of this initial message, a call setup sequence commences based on Q.931 operations with enchancements described below. The sequence is complete when the terminal receives in the Connect message(either from the gatekeeper or from the destination terminal) a guaranteed port on which to send H.245 control messages.

Once this guaranteed delivery control Transport Port has been established, additional Transport Ports for audio, video, and data may be established based on the outcome of the capability exchange using H.245 logical channel procedures. Also, the nature of the LAN side multi-media conference (centralized vs distributed/multicast) is negotiated on a per connection basis.[7] This negotiation is performed per media, in the sense that, for example, audio/video may be distributed, while data and control are centralized.

Audio and video are sent using RTP via a non-guaranteed delivery means to minimize delay. Error concealment or other recovery action must be applied to overcome lost packets; in general audio/video packets are not re-transmitted since this maywould result in excessive delay in the LAN environment.[8] However, an optional mode where retransmission is support can be signaled via H.245. The assumption is that bit errors are detected in the lower layers, and errored packets are not sent up to H.225.0. Note that audio/video and call setup/control are never sent on the same transport channel, and do not share a common PDU structure. Audio and video may be are sent to separate Transport Ports using separate instances of RTP to allow for media-specific frame sequence numbers and seperate quality of service treatment for each media.. However, an optional mode can be signaled via H.245 where audio and video are mixed in a single packet.

T.120 capabilities are negotiated using H.245, and upon receipt of appropriate messages, T.125 sessions are established using the transport/LAN stacks of T.123 as appropriate. There is no synchronization of T.120 data with the audio/video in RTP, or with the H.245 or Call Control~~Q.931~~ signaling. T.120 is conveyed over the LAN between endpoints, or between the endpoint and the gateway on another Transport Port (. Thus, the typical point-to-point, or point-to-gateway link can be expected to have at least six and up to 7 if T.120 is used. Note that T.120 may require up to four connections using the T.120 port for all four MCS priority levels.per T120 conference

#of ports	Usage	Reliable or Unreliable	Well Known or Dynamic	# of Connections
1 or more	Audio/RTP	unreliable	dynamic	1
1 or more	Audio/RTCP	unreliable	dynamic	1
1 or more	Video/RTP	unreliable	dynamic	1
1 or more	Video/RTCP	unreliable	dynamic	1
1	Call Signaling/H.225.0	reliable	well known	1
1	H.245	reliable	dynamic	1
1 or more	Data (T.120)]	reliable	dynamic	up to 4
1	Terminal to GK Admissions	unreliable	well known	1 (more for GK)

Add single UDP/RTP port for muxed channel as per AVC 863

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[7]The LAN side conference may be part centralized and part distributed, as decided by the MC controlling the conference. However, the terminal is not aware of this fact. Generally, of course, all terminals will see the same Selected Communications Mode(SCM)[see H.243 for a definition].

[8]This is not to imply that retransmission is never used; restransmission of video in MacroBlocks is an important error recovery strategy. However, a simple reliance on traditional retransmission protocols is not sufficient due to the requirement for low delay operation. **{Editor's Note: This statement must be confirmed by review of new IETF H.261 coding document; if re-transmission is not provided for perhaps we should add it as an optional feature.}**

Note that a well-known guaranteed delivery port is used for initial call setup for the terminal to terminal case, and also for the gatekeeper mediated case.. The reliable port shall be kept active throughout the call in either case.

Note that more than one connection may be open at a given time, e.g. a terminal may be in more than one conference. Note also that within a specific conference, a terminal may have more than one channel (of the same type) open (e.g. 2 Audio channels or even 2 H.245 channels). The only limitation is that between two terminals there shall be only one H.245 conference control channel in each direction.

H.245 logical channel signaling is used to start and stop video, audio, and data protocol usage. This process calls for closing the open channel, and then re-opening with a new mode of operation. The delays inherent in this process may lead to a situation where either (1)the RTP header has changed in advance of the H.245 signaling, or (2)data is lost waiting for the H.245 open channel to be acknowledged. For some cases, e.g. switching from H.261 to H.263, this additional delay is desirable, and corresponds to the use of Video-off in H.320, which is the propposed procedure for video coder changes. However, for audio mode changes the situation requires a 20 msec change, which only the RTP header can be assured of providing. (There is a case were even this is not sufficient. The assumption that the gateway is capable of sending a control (BAS code) 20 msec before a real change in a data occures on the H.320 network based on the RTP header change is not feasible in case that the Gateway was in a middle of sending an MBE. There is no alternative to have more tight control. The terminal that wish to change media type that requires synchronization with control should ask the other side (gateway) for a permition to make the change and make the change only after permition is received) Thus, the audio channel shall use the multi-mode capable audio channel in H.245. On this channel, the RTP header shall be taken as the indicator of current channel contents. *{Editor's Note: **Will this require an instantaneous mode change between coders? Will this cause problems?**}* In general, two types of conference modes of operation on the LAN side are possible: distributed and centralized. It is also possible that different choices may be made for different media, e.g. distributed audio/video and centralized data. Procedures for determining what sort of conference to establish are in H.323; the PDUs of H.225.0 are intended to support all allowed combinations, noting that distributed control and data are for further study although supported by the signaling.

6.2. Use of RTP/RTCP

A separate Transport Port mayshall-be established for audio and video. On each Transport Port a separate instance of RTPmay-shall be used, one for audio and one for video. Additional Transport Ports are needed for RTCP, one for each RTP Transport Port. Using H.245 signaling, additional audio and video channels (2 Transport Ports per channel, one for RTP and one for RTCP) may be established if the terminal supports this capability.

In this section two documents (Annex X and Annex Y), RTP and RTP profile (RTP-P for brevity) will be referred to(see references section for more information).

Unless an exception is specifically mentioned here, implementations shall follow those of the AnnexesRTP. Implementations shall follow RTP-P only as specifically mentioned in this document.

It should be kept in mind that the gateway may be acting as an RTP mixer (We don't see how according to the definition of a Gateway it may act as a mixerr), or it may be representing a single endpoint, or it may be representing many H.320 endpoints as one RTP source. The Gateway may also act as an RTP monitor. *{Editor's Note: The relationship of the MCU to the RTP mixer/translator must be described here}*

The H.323 LAN terminal, when engaged in a any conference, whether point-to-point or multi-point, shall restrict the logical channel bit rate averaged over a period defined by the manufacturer to that signaled in the H.245 FlowControlCommands, H.245 logical channel commands, and the T.120 flow control mechanism.

When the H.323 LAN terminal is connected to an H.323 gateway, the gateway shall use the means of H.245 and T.120 to force the H.323 terminal to be less than or equal to the WAN side media rates, with the following exceptions:

- Control bandwidth on the LAN need not match that in H.221.
- Audio bandwidth on the LAN may match that in H.221 on the WAN, but with gateway transcoding a match is not required.
- In the case where the gateway is using a rate reducer; the LAN side H.323 terminal shall match the H.245 signaled rate, which will probably be less than the rate being sent over the WAN.

6.2.1. Audio

Version (V): Version 2 of RTP shall be used
CSRC Count(CC): This shall be set to zero for the H.323 endpoint and for the gateway when the H.323 gateway is not performing audio mixing (this is not clear. Is the audio mixing part of Multipoint conference and it happens that the MCU is in the Gateway or we have a different mixing.). When the H.323 gateway supports centralized audio mixing, the CSRC information may*{should this be SHALL?}* be used to provide information to the terminal concerning what terminals are present in the audio sum. *{Editor's note: This seems less problematic now than it did a couple of months ago. Is provision of CSRC mandatory in RTP?? Should we make it mandatory here?}*

Payload Type(PT): Only ITU-T payload types such as (0)[PCMU], (8)[PCMA}, (9)[G722], and (15)[G728] G.723 shall be used. *{Editor's Note: It is our hope that codepoints will be allocated for G.723, G.729, and MPEG1 audio and G.dsvd so that we can make use of them. } There is a need to codepoint and a full definition of a payload like the one we have for H.261. Otherwise there is no use for these options.*

It is recommended that If an interruption in sequence numbers is observed, the receiver should repeat the most recent octet such that the value of the repeated octet decays to silence in 10 msec.

When sending 48/56 kbit/sec PCM, the H.323 gateway shall pad the extra 1 or 2 bits in each octet, and use the RTP values for PCMA or PCMU(8 or 0). For Mu-law the padding consists of "1" in both the 7th and 8th bit. For A-law the 7th bit shall be 0 and the 8th bit 1.In the reverse direction the H.323 gateway shall truncate 64 kbit/sec G.711 on the LAN side to fit the G.711 rate being used in H.320. Thus, on the LAN side only 64 kbit/sec G.711 shall be used.

{Editor's note: At Yokosuka we decided to pad out G.722 on the LAN as well. We need a format and a set of rules for doing the padding. I also note that payload formats for G.728 need to be proposed and sent to the IETF as well.}

When using RTCP, both RR and SR packets should be sent periodically as described in RTP. Only the CNAME SDES PDU[9] should be used, and instead of the canonical name mentioned in RTP, the H.243 terminal identifier shall be sent. Other SDES PDUs are optional. CNAME is used to associate audio and video for purposes of lip synchronization. When more than one audio channel is open, no H.245 level signal can be used to associate a particular audio channel with a particular video channel, so the CNAME is used for this purpose. Audio/Video pair must have a unique CNAME, and the procedure of RTP section 6.4.1 is not adequate to this task. The H.323 terminal shall append an H.243 terminal identifier to the <M><T> pair to make it unique. The H.323 terminal identifier is the same as the H.230 terminal identifier (coded as per H.230, limited to 32 characters). It is possible but not required that this terminal identifier could be a transport level domain name. The terminal identifier might also be a user name or a user location. Note that the <M><T> pair shall be coded as H.230 characters, not numbers in this case, with a "-" seperator. Thus, a CNAME might be "1-27-abc defgeh" or "1-27-xyz@bigmachine.company.country. In the event that the CNAME used on the LAN side exceeds 32 characters, the H.323 gateway shall truncate it when passing it to the WAN side.*{Editor's Note: a suggestion has been made that the above procedure requires further study; it will be considered again at Ipswich}*

If possible, the H.323 terminal should make use of the silence suppression feature of RTP, especially when the conference is multicast. The H.323 terminal shall be able to receive silence compressed RTP streams.

6.2.2. Video PDUs

Version (V): Use RTP version 2.

CSRC Count(CC): This shall be set to zero for the H.323 endpoint and the gateway. What about MCU that does video mixing

Payload Type(PT): Only ITU-T payload types such as that for H.261 shall be used. *{Editor's Note: It is our hope that the IETF will include H.263 as well.}*

When using RTCP, both RR and SR packets should be sent periodically as described in RTP. The CNAME SDES PDU shall be used(other SDES PDUs are optional), and when the gateway is involved in the call instead of the canonical name mentioned in RTP, the H.243 terminal identifier shall be sent as the CNAME as described in section 6.2.1.above. On a point-to-point call CNAME shall follow the conventions of RTP.

{Editor's Note: The usage of Freeze picture in RTP and in H.245, and also picture release, must be rationalized}

H.261 is packetized on the LAN side as per RFCH261(reference to be added). As long as sufficiently large RTP packets are available, fragmentation on MB boundries by the transmitter is not required. However, if the H.323 terminal fragments H.261 packets on the RTP level, this fragmentation shall occur on MB boundries. All H.323 terminals shall be able to receive GOBs and MB fragmented packets. Note that failure to support MB fragmentation in the transmitter may result in the loss of an entire GOB, and may also lower the packet rate. MBs shall not be split across packets; all packets shall end on a GOB or MB boundry. The H.323 transmitter may chose to fill out a packet containing a small GOB with additional MBs, but this is not required.

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[9]MCUs may change CNAMEs during conferences.

{Editor's Note: It was requested that the experts consider requiring packets to align with picture headers as in H.324; this must be considered in the context of H.324 compatibility}

It is recommended (if not specified in RFCH261) to have an RTP packet a full or part of one video frame and not to have two frames in one RTP packet.

6.2.3.Data PDUs

There are no special data PDUs; T.120 is used on the LAN as per T.123, Centralized vs distributed data conferencing on the LAN is described in H.323, and is negotiated via H.245. T.120 flow control on the LAN is managed using LAN protocols, and not H.245 FlowControlCommands.

{Editor's Note: Further consideration should be given to how T.120 is associated with the H.320 conference}

6.2.4. Overall Call Management using RTP/RTCP

{Editor's note: this section is probably not needed, but I am trying to clarify this in my own mind. The fate of this section will be decided at Ipswich.}

In the simplist case of a point-to-point connection on the LAN, the SSRC is used to sychronization the received audio/video from a terminal-A, and the two streams are associated by a CNAME as specified in RTP.

When a gateway or MCU becomes involved in the call (either for a LAN/WAN point-to-point call, or a multi-point call involving one or more LAN endpoints), the H.323 gateway or MCU -uses H.245 to assign H.243 terminal identifiers of the form <M><T> to each terminal on the LAN side. The <M><T> pair is unique within the conference, and shouldcan thus be used instead of the RTP CNAME.

In the event that a terminal sources more than one audio/video pair, a new CNAME is needed. This requires that suffixsomething be added to the <M><T> pair to make it unique. The H.243 terminal identifier is appended to the <M><T> pair to produce a globally unique value.
In the event that a terminal participates in more than one conference at a time, it may potentially be receiving two pairs (or more) of audio/video that have the same <M><T> pair being used as the CNAME, since <M><T> pairs are only unique within a conference. The addition of a terminal id to the <M><T> pair should have the effect of making the CNAME unique within the conference, but the terminal should also be able to use its knowledge of H.225.0/H.245 signaling to distinguish the two pairs of audio/video since they will have separate, unique signaling channels.

These procedures shall apply to H.323 calls even if a gateway is not involved; RTP does not deal with having more than one CNAME per terminal.

{Editor's Note: This was an attempt to deal with the affect of the MCU on RTP. correct text on this topic shall be added as agreed}

7. Initialization and Call Setup PDU Definitions

This section concerns the definition of PDUs for call setup, call control, and communications between terminals, gateways, gatekeepers, and MCUs.

General Note: Extensions to Q.931 messages follow the format used in Q.931. H.225.0 only messages are specified using ASN.1.

7.1. Use of Q.931

Implementations shall follow the Q.931 as specified in H.225.0. Terminals may also support optional Q.931 and messages. The use of such messages is for further study. The Q.931 messages shall contain all of the mandatory information elements and may contain any of the optional information elements as defined in Q.931 as described in H.225.0. Note that the H.323 terminal, may, according to Q.931, ignore all optional messages it does not support without harming interoperability.

H.225.0 makes use of User to User informatioon in Q.931 to convey transport addresses and other information.

In this version of H.225.0, all references are to the 1993 version of Q.931/Q.932. The procedures of Q.931/Section 3.1 for circuit mode connection setup are followed. However, the implementor is reminded that although "bearer" is being signaled for, no actual "B-channels" of the ISDN type exist on the LAN side. Successful comletion of the "call" results in an end-to-end reliable channel supporting H.245 messaging. Actually "bearer" setup is done using H.245. However, the use of Q.931 on the LAN side enables interworking with Q.931 on the WAN side as well as providing a well-tested framework for general connection oriented calling features.

{Editor's Note: A suggest has been made that in gatewayed operation, the gateway be required to copy IEs from the network to the terminal. This seems reasonable - do you agree?} *(What is an IE ?)*

The following table taken from Table 3-1/Q.931 show what messages are mandatory and optional for H.323/H.225.0 call setup using Q.931 on the LAN:

Call Establishment Messages	**Status (M or O)**
Alerting	M
Call Proceeding	M
Connect	M
Connect Acknowledge	M
Setup	M
Setup Acknowledge.	O(consider at Ipswich)
Call Clearing Messages	
Disconnect	M(consider at Ipswich)
Release	M?(consider at Ipswich)
Release Complete.	M(consider at Ipswich)
Call Information Phase Messages	
Resume	O
Resume Acknowledge	O
Resume Reject	O
Suspend	O
Suspend Acknowledge	O

Suspend Reject	O
User Information	O?
Miscellaneous Messages	
Congestion Control	O
Facility	O
Information	O
Notify	O
Status	O
Status Inquiry	O

For all Q.931 messages, there two common fields that are mandatory in addition to the message type:

1. The Protocol Discriminator Information Element in the Q.931 message shall contain the Q.931 user-network call control message identifier value of 08H.

2. The Call Reference Information Element in the Q.931 message shall be two octets long and shall contain a call reference code as specified by H.225.0.

{Editor's Note: It was agreed in Geneva that all call messges would contain a provision for a proprietary extension based on H.221 and H.245. This shall be added}

7.2. Alerting

Follow Table 3-2/Q.931 as modified below, keeping in mind that Note 1 can be ignored, i.e. Channel identification is not needed as it has no meaning on the LAN. Shaded parts are not applicable to H.323 terminals.

This message is sent by the called user to the network(either the gatekeeper or the calling terminal) and by the network to the calling user, to indicate that called user alerting has been initiated. In everyday terms, the "phone is ringing." See Table 3-2.

TABLE n3-2/H.225~~Q.931~~

ALERTING message content

Message type:			ALERTING
Significance:			global
Direction: both			

Information element	Direction	Type	Length
Protocol discriminator	Both	M	1
Call reference	Both	M	2-*
Message type	Both	M	1
Bearer capability	Both	O (Note 1)	4-12
Progress indicator	Both	O (Note 4)	2-4
Display	n -> u	O (Note 5)	(Note 6)

NOTES

1 The Bearer capability information element is included when the procedures of 5.11 for bearer capability selection apply. When present, progress description No. 5, *interworking has occurred and has resulted in a telecommunication service change* shall also be present.

~~2 Included in the network to user direction for support of the procedures in Annex D.~~

~~3 Mandatory if this message is the first message in response to a SETUP, unless the user accepts the B channel indicated in the SETUP message.~~

4 Included in the event of interworking. Included in the network-to-user direction in connection with the provision of in-band information/patterns. Included in the user-to-network direction in connection with the provision of in-band information/patterns if Annex K (of what. If it annex of Q.931 it should be an Annex to H.225)is implemented or in accordance with the procedures of 5.11.3 and 5.12.3. (of what ? see comment above)

5 Included if the network provides information that can be presented to the user.

6 The minimum length is 2 octets; the maximum length is network dependent and is either 34 or 82 octets. When 34 and when 82.

~~7 Included if the network optionally provides information describing tones or alerting signals.~~

~~8 The High layer compatibility information element is included when the procedures of 5.12 for high layer compatibility selection apply. When present, progress description No. 5, *interworking has occurred and has resulted in a telecommunication service change* shall also be present.~~

{Editor's Note: the proprietary extension method shall be added at the end of the message}

Note 2 is not required either.

Note 6 When does this 34 octets and when 82 octets.

{Editor's Note: Request seq num is not needed as this a reliable channel. The connection id is now the Q.931 CRV, the Q.931 progress indicator contains connection status. Exactly what role the channel number played is unclear and requires further consideration}

7.3. Call Proceeding

Follow Table 3-3/Q.931 as modified below, noting that Note 1 can be ignored, i.e. Channel identification is not needed as it has no meaning on the LAN.

This message is sent by the called user to the network(gatekeeper or calling terminal) or by the network to the calling user to indicate that requested call establishment has been initiated and no more call establishment information will be accepted. See Table 3-3. ***{Editor's Note: We may wish to add some more detail on when this message is used; my sense is that it may not be needed for direct terminal to terminal connections}***

TABLE n-33-3/H.225~~Q.931~~

Message type: CALL PROCEEDING Significance: local Direction: both			
Information element	**Direction**	**Type**	**Length**
Protocol discriminator	Both	M	1
Call reference	Both	M	2-~~4~~
Message type	Both	M	1
Bearer capability	Both	O (Note 5)	4-12
Progress indicator	Both	O (Note 2)	2-4
Display	n → u	O (Note 3)	(Note 4)

NOTES

~~1 Mandatory in the network to user direction if this message is the first message in response to a SETUP message. It is mandatory in the user to network direction if this message is the first message in response to a SETUP message, unless the user accepts the B-channel indicated in the SETUP message.~~

2 Included in the event of interworking. Included in the network to user direction in connection with the provision of in-band information/patterns. Included in the user to network direction in connection with the provision of in-band information/patterns if Annex K (of what) is implemented or in accordance with the procedures of 5.11.3 and 5.12.3. (of what)

3 Included if the network provides information that can be presented to the user.

4 The minimum length is 2 octets; the maximum length is network dependent and is either 34 or 82 octets.

5 The Bearer capability information element is included when the procedures of 5.11 (of what?) for bearer capability selection apply. When present, progress indication No. 5, *interworking has occurred and has resulted in a telecommunication service change* shall also be present.

~~6 The High layer compatibility information element is included when the procedures of 5.12 for high layer compatibility selection apply. When present, progress description No. 5, *interworking has occurred and has resulted in a telecommunication service change* shall also be present.~~

CALL PROCEEDING message content

{Editor's Note: see Alerting for how connection in progress fields are supported using Q.931. Also, a proprietary extension field shall be added}

7.4. Connect

Follow Table 3-4/Q.931, as modified below and noting that Note 1 can be ignored, i.e. Channel identification is not needed as it has no meaning on the LAN.

This message is sent by the called user to the network(gatekeeper or called terminal) and by the network to the calling user, to indicate call acceptance by the called user. See Table 3-4 below.

TABLE 3-4/H.225~~Q.931~~

Message type: C
Significance:
Direction: both

Information element	Direction	Type	Length
Protocol discriminator	Both	M	1
Call reference	Both	M	2-
Message type	Both	M	1
Bearer capability	Both	O (Note 1)	4-12
Progress indicator	Both	O (Note 4)	2-4
Display	n -> u	O (Note 5)	(Note 6)

NOTES

1 The Bearer capability information element is included when the procedures of 5.11 for bearer selection apply.

~~2 Included in the network to user direction for support of the procedures in Annex D.~~

3 Mandatory if this is the first message in response to a SETUP, unless the user accepts the E indicated in the SETUP message.

4 Included in the event of interworking or in connection with the provision of in-band information/pattern

5 Included if the network provides information that can be presented to the user.

6 The minimum length is 2 octets; the maximum length is network dependent and is either 34 or 82 oct

~~7 As a network option, may be included to provide date and time information to the calling user for a for calls involving specific telecommunication services.~~

~~8 Included if the network optionally provides additional information describing tones.~~

~~9 Included in the user to network when the answering user wants to return low layer compatibility info the calling user. Included in the network to user direction if the user awarded the call included a L compatibility information element in the CONNECT message. Optionally included for low layer co negotiation, but some networks may not transport this information element to the calling user (see Annex J).~~

~~10 The High layer compatibility information element is included when the procedures of 5.12 for h compatibility selection apply.~~

CONNECT message content

Also, in the User-user field the following information shall be provided:

> **destinationAddress** - this is a specific transport address on which the called terminal or gatekeeper handling the call would like to establish H.245 signaling.
> **conferenceID** - Will contain a unique number , as specified by the gatekeeper cloud or the called terminal. This will allow the conference to be uniquely identified from all others.
> *There is a need for the extention field here too.*
> ***{Editor's Note: the relationship of the conference id to the H.243 conference id needs to be explored and clarified at Ipswich}***

7.5. Connect Acknowledge

Follow Table 3-6/Q.931. ***{Editor's Note: A table as above will be added}***

7.6. Progress

Follow table 3-10/Q.931. ***{Editor's Note: A table as above will be added}***

{Editor's Note: See Alerting for a discussion of how the fields in Connection in Progress were dealt with}

7.7. Setup

Follow Table 3-16/Q.931 as modified below

This section is not clear and there is no way to comment on that.

The Bearer Capability Information Element in the Q.931 message shall be four octets long and shall contain the following information:

- Coding Standard: 00 (binary) CCITT standardized in Q.931
- Information Transfer: 01000 (binary) Unrestricted digital information
- Transfer Mode: 00 (binary) Circuit-mode ***{Editor's Note: Input on the possible use of packet mode are requested}***
- ***{This can be used for Gatekeeper Bandwidth control. Should we use it? Probably not - consider at Ipswich}***
- The following fields are not used: Structure, Configuration, Establishment, Symmetry, Information Transfer Rate (dest to orig), Layer Identification, Protocol Identification.

The following items will be added to user-to-user information as part of the setup message.

{Editor's Note: Calling number is in Q.931}.

{Editor's Note: This is specified by the called party, I am told}

Editor's Note: the terminal is sending to this address so there is not need to include it here, correct?}

{Editor's Note With the demise of MRQ, these are not needed, correct? Discuss at Ipswich. I have received two opposing inputs.}

destinationWanInfo - this specifies further contact information that a gateway might use. *{Editor's Note: the actual E.164 number is in Q.931, but we may wish to send additional numbers for additional channels. This requires more discussion}*

{Moved to ARQ}

{Moved to ARQ}

conferenceID - This value is always zero in the setup message for a new call, or the current CID if the call is to be conferenced.

{Editor's Notes:the above text must be converted to Q.931 format, and its meaning re-defined to address possible additional channels}

7.8. Setup Acknowledge

Follow Table 3-16/Q.931.

The Cause Information Element in the Q.931 message shall contain the following information:

- Coding Standard: 00 (binary) CCITT standardized in Q.931
- Location: as coded in Q.931
- Recommendation: 00H Q.930
- Cause Value: as coded in Q.931

{Editor's Note: the use of the Setup Acknowledge message requires further consideration at Ipswich; A table as above will be added.. It is not clear to me that any network addresses are carried in this message. This is a strong candidate for an optional message}

7.9. Disconnect

Follow table 3-7/Q.931. ***{Editor's Note: We may not be using this; details will be added at Ipswich}***

7.10. Release

Follow Table 3-11/Q.931. ***{Editor's Note: We may not be using this; details will be added at Ipswich}***

7.11. Release Complete

Follow Table 3-12/Q.931. ***{Editor's Note: We may not be using this; details will be added at Ipswich; one proposal is that only Release Complete is needed}***

{Editor's Note: this must be converted to Q.931 from the ASN.1 below and updated. Possibly some of the deleted information in the box above should be added as well.}

```
RejectReason            ENUMERATED
{
        No Bandwidth            (1),
        Gatekeeper Resources    (2),
        Unreachable Destination         (3),
        Destination Rejection   (4),
        Invalid Revision        (5),
        No Permission           (6),
        UnreachableGatekeeper           (7),
        Destination Busy                (8),
        Not Bound               (9),        -- From local Gatekeeper
        Gateway Resources       (10),
        Bad Format Address      (11),
```

```
                Caller Not Bound              (12),        -- Destination Gatekeeper
                Caller Not Bound              (13),        -- Destination Gatekeeper
                {Does not seem appropriate}
                Undefined Reason        (65535)
        }
```

{Editor's Note: Is there some proposal for some other protocol? If so we should consider that}

{Editor's Note: This appears too restrictive for source routing; this must be considered}

7.12. H.225.0 PDU Common Parts

requestSeqNum in PDUs are used to keep track of multiple outstanding requests. It is expected that any associated response PDUs (success or failure) will have the corresponding **requestSeqNum** returned with it.

{Editor's Note: YearOfSpec will be globally replaced with the appropriate H.245 mechanism. Also, a proprietary extension field will be added to all messages. Also, all units will be set at 100 bits}

The **NetworkAddress** structure is meant to capture the various transport formats and includes any transport specific scheme in addition to the possibly local reference to a 'port' number.

```
NetworkAddress              ::=CHOICE
{
        IPAddress           SEQUENCE
        {
                transport   OCTET STRING (SIZE(4)),
                port        INTEGER(0..4294967295)
        },
        IPXAddress          SEQUENCE,
        {
                node        OCTET STRING (SIZE(6)),
                netnum      OCTET STRING (SIZE(4)),
                port        OCTET STRING (SIZE(2))
        },
        IP6Address          SEQUENCE,
        {
                transport           OCTET STRING (SIZE(16)),
                port        INTEGER(0..4294967295)
        },
        NetBios             OCTET STRING (SIZE(16)),
        PropExpansion       ToBeAdded,
}


                NodeType                ENUMERATED
                {
                        Gatekeeper              (1),
                        Gateway                 (2),
                        MCU                     (4),
                Terminal                (8),
                        Undefined Node          (268435456)
                        propExtesnion           ToBeAdded
```

}

7.13. Terminal and Gateway Registration PDUs

{Editor's Comment: How many GRQs are sent? One per address? One per device? This must be clarified}

```
GuardianRequest              ::=SEQUENCE --(GRQ)
{
        requestSeqNum          INTEGER (1..65535),
        terminalIdentifier     OCTET STRING (SIZE(128)),
        terminalAddress        NetworkAddress,
        terminalType           NodeType,
        gatekeeperIdentifier   OCTET STRING (SIZE(64)),
        propExtesnion          ToBeAdded
}
```

requestSeqNum - this is a monotonically increasing number unique to the caller. It should be returned by the called in any PDUs associated with this specific PDU.
terminalIdentifier - this is a terminal/user specific string used to identify the caller. It is presumed that application software has made appropriate authentication and this can be 'trusted'. It should be passed unmolested from application end to end.
ControlAddress - this is the network control address for this terminal. If multiple transports are supported, they must be requested separately. This address includes local port information.
terminalType - this specifies the type(s) of the terminal that is registering (note that a H.323 terminal may also be an H.323 MCU).
gatekeeperIdentifier - this is a string value that is used to logically identify a called gatekeeper. It should be initialize to a zero (0) value by the caller.

```
GuardianConfirmation ::=SEQUENCE --(GCF)
{
        yearOfSpec             OCTET STRING (SIZE(4))
        requestSeqNum          INTEGER (1..65535),
        gatekeeperIdentifier   OCTET STRING (SIZE(64)),
        gatekeeperAddress      NetworkAddress,
        extensionCount         INTEGER (0..65535)
}
```

requestSeqNum - This should be the same value that was passed in the GRQ by the caller.
gatekeeperIdentifier - this is a string value that is used to logically identify a called gatekeeper. It may be used by the caller for future RRQs.
gatekeeperAddress - this is an array of transport addresses; one for each transport that the gatekeeper will respond to. This address includes local port information.

```
Guardian Reject          ::=SEQUENCE --(GRJ)
{
        requestSeqNum          INTEGER (1..65535),
        gatekeeperIdentifier   OCTET STRING (SIZE(64)),
        rejectReason           GuardianRejectReason,
        propExtension          ToBeAdded

}

        GuardianRejectReason          ENUMERATED
        {
                Resource Unavailable          (1),
                Terminal Excluded             (2),
                Invalid Revision              (5),
                Undefined Reason              (65535)
        }
```

{Editor's Note: A suggestion has been made that registration should optionally be a "login" type procedure. Ths must be considered}

```
RegistrationRequest          ::=SEQUENCE --(RRQ)
{
                requestSeqNum          INTEGER (1..65535),
        bindRequest            BOOLEAN,
        terminalIdentifier     OCTET STRING (SIZE(128)),
        ControlAddress         NetworkAddress,
        terminalType           NodeType,
        terminalExtNumE.164Address

        propExtesnion          ToBeAdded
}
```

requestSeqNum - this is a monotonically increasing number unique to the caller. It should be returned by the called in any PDUs associated with this specific PDU.
bindRequest - set to TRUE if requesting a new binding with called gatekeeper; set to FALSE if registering only.
terminalIdentifier - this is a terminal/user specific string used to identify the caller. It is presumed that application software has made appropriate authentication and this can be 'trusted'. It should be passed unmolested from application end to end.
ControlAddress - this is the network control address for this terminal. If multiple transports are supported, they must be registered separately. This address includes local port information.
terminalType - this specifies the type(s) of the terminal that is registering (note that a H.323 terminal may also be an H.323 MC).
terminalExtNum - This optional value is a phone number by which external (to the LAN) terminals may identify this terminal.

```
RegistrationConfirmation        ::=SEQUENCE --(RCF)
{
            requestSeqNum          INTEGER (1..65535),
    gatekeeperIdentifier      OCTET STRING (SIZE(64)),

    propExtesnion             ToBeAdded
}
```

requestSeqNum - This should be the same value that was passed in the RRQ by the caller.
gatekeeperIdentifier - this is a string value that is used to logically identify a called gatekeeper. It may be used by the caller for future RRQs.
gatekeeperAddress - this is an array of transport addresses; one for each transport that the gatekeeper will respond to. This address includes local port information.

```
Registration Reject             ::=SEQUENCE --(RRJ)
{

    requestSeqNum             INTEGER (1..65535),
    rejectReason              RejectReason,
    propExtesnion             ToBeAdded
}
```

requestSeqNum - This should be the same value that was passed in the RRQ by the caller.

```
    RejectReason              ENUMERATED
    {
            Not Bound Registration              (1),
            Duplicate Registration Request      (2),
            Invalid Ext Num            (3),
            Duplicate Bind Request              (4),
            Invalid Revision                    (5),
            Invalid Network Address(6),
            Undefined Reason                    (65535)
    }
```

7.14. Gatekeeper to Terminal Guardian Query Messages

```
Guardian Query Request              ::=SEQUENCE --(GQQ)
{
            requestSeqNum          INTEGER (1..65535),
    replyAddress              NetworkAddress,

    propExtesnion             ToBeAdded
}
```

Guardian Query Response ::=SEQUENCE --(GQRS)

```
{
        requestSeqNum          INTEGER (1..65535),
        gatekeeperIdentifier   OCTET STRING (SIZE(64)),
        gatekeeperAddress      NetworkAddress,

        propExtesnion          ToBeAdded
}
```

Guardian Query Reject ::=SEQUENCE --(GQRJ)

```
{
        requestSeqNum          INTEGER (1..65535),
        rejectReason           GuardianQueryRejectReason,
        propExtesnion          ToBeAdded
}
        GuardianQueryRejectReason          ENUMERATED
        {
                No Gatekeeper          (1),
                Invalid Revision       (5),
                Undefined Reason       (65535)
        }
```

7.15. Terminal to Gatekeeper Admission Messages

AdmissionRequest ::=SEQUENCE --(ARQ)

```
{
        requestSeqNum          INTEGER (1..65535),
        terminalIdentifier     OCTET STRING (SIZE(128)),  -- may be E.164 number
        callType               CallType,
        CallMedia              CallMedia
        replyAddress           NetworkAddress,
        bandWidth              INTEGER (1..4294967295),  -- measured in 1k bit increments
        reserveRequest         BOOLEAN( 0 - terminal is asking the Gatekeeper to make a
                               bandwidth reservation request for it.
                                        1  terminal is asking the gatekeeperto NOT make
                               any bandwidth reservation for it; such a reservation may have
                               already been made.
        propExtension          To be provided
}
```

CallType ENUMERATED *{Editor's Note: May need revision}*

```
        {
                PointToPoint       (1)        -- Point to point
                OneToN             (2),       -- no interaction (a podium)
                NToOne             (4),       -- no interaction (a listener)
                NToN               (8),       -- interactive
                BroadCast          (16),      -- Multicast included
        }
```

requestSeqNum - this is a monotonically increasing number unique to the caller. It should be returned by the called in any PDUs associated with this specific PDU.

terminalIdentifier - this is a terminal/user specific string used to identify the caller/called. It is presumed that application software has made appropriate authentication and this can be 'trusted'. *{Editor's Note: This requires further discussion}*
callType - Using this value, gatekeeper can make determine 'real' bandwidth usage.
replyAddress - this is the transport address to which the **ACF**, or **ARJ** is to be sent.
bandWidth - the number of 1k BITS/sec requested for the connection.

```
Admission Confirmation        ::=SEQUENCE --(ACF)
{
        requestSeqNum         INTEGER (1..65535),
        bandWidth             INTEGER (1..4294967295)  -- measured in 1k bit increments
        callModel             BOOLEAN (0  call signaling to gatekeeper,
                                       1  call signaling directly to terminal)
        replyAddress          NetworkAddress,
        propExtension         To be provided
}
```

requestSeqNum - This shall be the same value that was passed in the **ARQ** by the caller.
bandWidth - the maximum that *might* be offered with a BRQ.
callModel - tells terminal whether call signaling sent on replyAddress goes to a gatekeeper or to a terminal
replyAddress - the address to send Q.931 call signaling; uses the reliable well known port, but may be the terminal or the gatekeeper address.

```
Admission Reject              ::=SEQUENCE --(ARJ)
{
        requestSeqNum         INTEGER (1..65535),
        rejectReason          AdmissionRejectReason,
        propExtension         To be provided
}

        AdmissionRejectReason              ENUMERATED {Editor's Note: requires
        revision}
        {
                Not Bound                  (1),
                Invalid Permission         (3),
                Request Denied             (4),
                Undefined Reason           (65535)
        }
```

Gatekeeper to gatekeeper messages are for further study.7.16. Terminal to Gatekeeper Requests for Changes Bandwidth

{Editor's Note: Issues related to ensuring that all terminals in a conference coordinate their requests for more bandwidth need to be addressed at Ipswich}

```
BandwidthRequest          ::=SEQUENCE --(BRQ)
{
        requestSeqNum       INTEGER (1..65535),
        terminalIdentifier  OCTET STRING (SIZE(128)), {Requires discussion}
        ConferenceID        INTEGER (1.. 4294967295), {Editor's Note:
                perhaps this should be the CRV}
        callType            CallType,

        replyAddress        NetworkAddress,
        ,bandIncDec         BOOLEAN (0 decrease, 1 increase),
        E164add             OPTION,

        bandWidth           INTEGER (1..4294967295)  -- measured in 1k bit increments
        propExtension       ToBeAdded

}
```

requestSeqNum - this is a monotonically increasing number unique to the caller. It should be returned by the called in any PDUs associated with this specific PDU. **terminalIdentifier** - this is a terminal/user specific string used to identify the caller/called. It is presumed that application software has made appropriate authentication and this can be 'trusted'. **Conference idID** - ID of connnection that is to have the bandwidth changed. **callType** - Using this value, gatekeeper can make determine 'real' bandwidth usage. **replyAddress** - this is the transport address to which the BCF, or BRJ is to be sent. **bandWidth** - the NEW number of 1k BITS/sec requested for the connection.

```
Bandwidth Confirmation    ::=SEQUENCE --(BCF)
{
        requestSeqNum       INTEGER (1..65535),
        bandWidth           INTEGER (1..4294967295)  -- measured in 1k bit increments

        propExtension       ToBeAdded
}
```

requestSeqNum - This should be the same value that was passed in the BRQ by the caller. **bandWidth** - the maximum that *might* be offered with a new BRQ.

```
Bandwidth Reject          ::=SEQUENCE --(BRJ)
{
        requestSeqNum       INTEGER (1..65535),
        rejectReason        BandRejectReason,
        propExtension       ToBeAdded
}
```

```
BandRejectReason            ENUMERATED {Editor's Note: Requires expansion?}
{
        Not Bound                   (1),
        Invalid ConferenceID        (2),
        Invalid Permission          (3),
        Request Denied              (4),
        Invalid Revision            (5),
        Undefined Reason            (65535)
}
```

7.17. Status Request Messages

Status Request ::=SEQUENCE --(SRQ)

```
{
        requestSeqNum       INTEGER (1..65535),

        propExtension       ToBeAdded
}
```

Status Report Response ::=SEQUENCE --(SRR) {Editor's Note: requires revision}

```
{
        requestSeqNum           INTEGER (1..65535),
        nodeType                NodeType,
        conferenceID            INTEGER (1..4294967295),
         (needs revision)
        originatingID           OCTET STRING (SIZE(128)),
        originatingAddress      NetworkAddress,
        destinationAddress      NetworkAddress,
        originatingGatekeeper   NetworkAddress,
        destinationGatekeeper   NetworkAddress,
        destinationWanInfo      WanInfo,
        callType                CallType,
        callMedia               CallMedia,
        bandWidth               INTEGER (1..4294967295),
        bytesSent               INTEGER (1..4294967295),
        bytesRcvd               INTEGER (1..4294967295),
        {Editor's Note:  A method of passing a list of all ports in use shall be added
        here as agreed}

        propExtension           ToBeAdded
}
```

7.18. QOS Related PDUs

7.18.1. Adaptive Busy Indication (ABI)

Will be signaled via Q.931 busy indication.

8. Mechanisms for maintaining QOS

It was discussed in Geneve to revise the whole section.
{Editor's Note: It was agreed to do the following:

a)Add a definition of QOS
b)add consideration of echo control
c)add an option for a terminal to either ask the gatekeeper to make a QOS reservation on its behalf, or for this not be be done. This has been added to the ARQ message.}

8.1.General Approach and Assumptions

Any QOS related signaling (e.g. a reservation request to a router) is done by the terminal before the call request to the gatekeeper, or by the gatekeepr on its behalf. The terminal indicates which should be done via the ARQ message. The terminal may wish to make any reservations since the gatekeeper may not be logically near the terminal, or be in a position to make QOS related requests on behalf of the terminal. The means by which either the terminal or the gatekeeper make QOS or bandwidth reservations are beyond the scope of this recommendation.
The Sender and Receiver Reports of RTCP shall be the means by which QOS will be assessed. Logically, there are two types of congestion related delay that might be measured:

- Short term increases in delay that will result in a perceptible but not annoying slowing of the frame rate. These bursts are less than 1 ***{Editor's note: what is the right figure; this is too short}*** second in duration.
- A general rise in delay due to LAN congestion over time such that a feedback based mechanism is useful. This rise in congestion is measured over 1 minute intervals***{Editor's note: is this the right interval?}***Essentially, short term bursts are approached by error concealment, and a longer term congestion is approached by reducing the multi-media load. The assumption is made that all LAN multimedia terminals are H.323 terminals, and all will attempt to reduce LAN usage as congestion rises rather than "steal" bandwidth from each other.

8.2. Use of RTCP in Measuring QOS

8.2.1. Sender Reports

The sender report serves three main purposes:

1. Allow synchronization of multiple RTP streams, such as audio and video.
2. Allow the receiver to know the expected data rate and packet rate.
3. Allow the receiver to measure the distance in time to the sender.

Of these three purposes, (1) is the most relevant to H.225.0. Manufacturers may make use of the sender reports in other ways at their discretion.

The relevant field for stream synchronization is the RTP timestamp and the NTP timesamp in the sender report of RTCP. The NTP timestamp (if available) gives "wall clock" time and corresponds to the RTP timestamp which has the same units and random offset as the RTP capture timestamp in the media packets. Although not required in RTCP, the H.323 terminal should provide the NTP timestamp. The CNAME in RTCP binds the different SSRC identifiers from the same sender and can be used for synchronization.

8.2.2. Receiver Reports

{Editor's Note: All numbers in this section are not considered to be determined}

Three parts of the Receiver Reports are used in H.225.0 to measure QOS:

1. Fraction Lost
2. The cumulative packets lost
3. The extended highest sequence number received
4. Interarrival jitter

Items 2 and 3 are used to compute the number of packets lost since the previous receiver report. This can be taken as a long term measure of LAN congestion. See RTP section 6.3.4 for a sample computation. If this loss rate exceeds 1% ***{Editor's note: What is a good value?}*** the H.225.0 terminal should reduce the media rates on the LAN side according to the procedures in section XXX below. If item 1 exceeds 1%, it may also be desirable to take corrective action. If the interval between receiver reports exceeds 5 minutes, H.323 terminals shall use item 1 as an indicator of serious congestion requiring media rate reduction on the LAN side.

Item 4 should be used as an indication of impending congestion. If interarrival jitter increases for three consecutive receiver reports, the H.323 sending terminal should take corrective action.

{Editor's note: There is a strong element of guesswork in these numbers and rules. I have attempted to provide a kind of framework that will support interoperablity in behavior. Comments are especially welcome from those with experience in this area.}

8.3. Audio/Video Skew Procedures

{Editor's Note: this section is not considered to be determined; the possible need for decoder requirements will be resolved at Ipswich}

An issue exists concerning audio/video skew and possible need for a requirement on encoders to allow decoders to reliably operation. If we agree on this, a section will be added here describing the specific limits and explaining the issue.

8.4. Procedures for Maintaining QOS

.A number of methods exist for the H.323 gateway/terminal to respond to an increase in packet loss or interarrival jitter in the far end receiver. These methods can be grouped into those that are appropriate for a rapid response to a short term problem,'such as a lost or delayed packet, and those that are appropriate for a response to a longer term problem such as growing congestion on the LAN.

Short term responses:

- Reducing the frame rate for a short period of time. This may result in the H.323 gateway sending additional H.261 fill frames in the LAN->WAN direction to compensate for the packet underflow.

- Need to add additional short term responses for H.261

- Reduce packet rate by switching to the optional mode where audio/video are mixed in one packet. Packet rate can also be reduced via the use of MB fragmentation of the video stream. ***{Editor's Note: Should these methods be moved to longer term respnse below?}***

Longer term responses:

- Reducing media bit rate(e.g. switching from 384 kbit/sec to 256 kbit/sec). This may involve a simple instruction to the encoder in a terminal, or it may involve the use of a rate reducer function in the H.323 gateway. These changes are signaled via H.245 FlowControl commands, or by logical channel signaling as appropriate.

- Turning off media of lessor importance (e.g. turning off video to allow a large amount of T.120 traffic). These changes are signaled via closing H.245 logical channels.

- Returning a busy signal (adaptive busy) to the receiver as an indication of LAN congestion. This may be combined with turning off a media, or even all media other than the control Transport Port. Adaptive busy is signaled via a Q.931 cause value as described in XXX.

9. Annex A: RTP/RTCP(Normative)

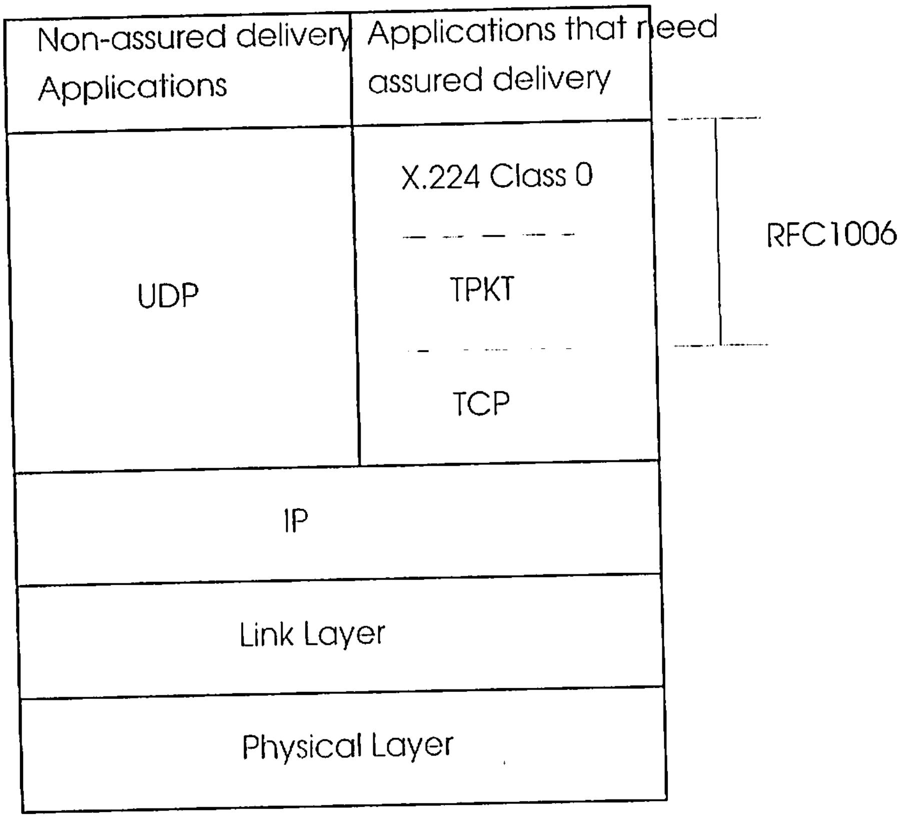
10. Annex B: H.261 Packetization(Normative)

11. Annex C: RTP Profile(Normative)

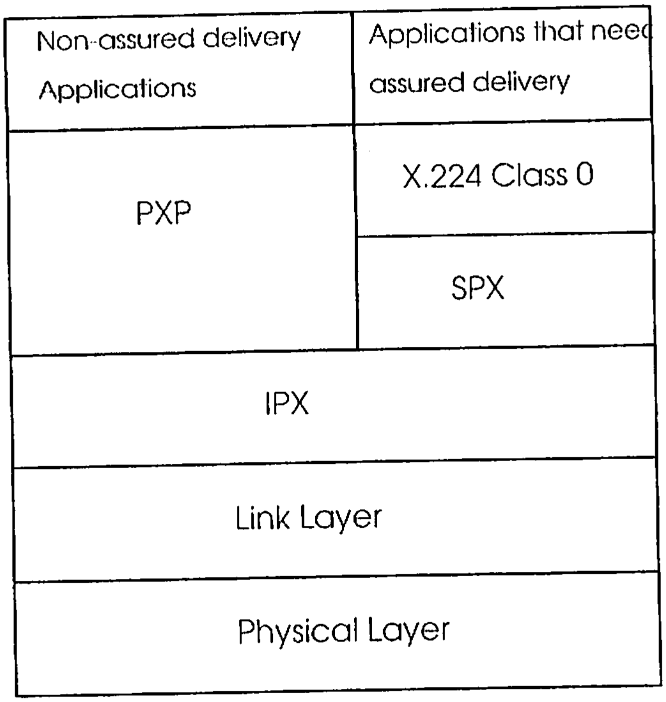
12. Appendix DA (Informative)

This annex provides additional details concerning the operation of H.225.0 on various actual LAN protocol stacks. This annex is non-normative.

12.1. TCP/IP/UDP



12.2. SPX/IPX



12.3. AppleTalk (Could someone provide this???)