Telecommunication Standardization Sector
English
(TSS)

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Question 2/15

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: January 15, 1996

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 messages, 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 AVC869v3 reflects changes from AVC869v2.

The following major technical directions constitute an integral part of the determination being made at the SG15 meeting in November 95:

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.

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

l) It has been agreed that this(letting the endpoint know whether it is calling a gateway or not) will be done by adding the node type (gateway/MCU) to the Setup/Connect sequence.}

The editor has been asked to consider 4 letter command abbreviations. If we used four letter abbreviations we would have:

RRQ ->RGRQ GRQ ->GBRQ GQQ ->GQRQ
RCF ->RGCF GCF ->GBCF ->GQRS
RRJ ->RGRJ GRJ ->GBRJ ->GQRJ

ARQ ->ADRQ, ACF ->ADCF, & ARJ ->ADRJ

SRQ->STRQ & SRR->STRR

BRQ ->BWRQ, BCF ->BWCF, & BRJ ->BWRJ

What do you think? If we are going to change, let's do it ASAP. I don't think it has any great importance, but I note that we already need four letters in one case, so it probably makes sense.

New Notes for this version:

a) The following additional terminology changes have been made:

c)I have attempted to updated the QOS section; comments welcome. (Still Applies, no comments so far)

d)With regard to Q.931 - there has never been an intent to bring all parts of Q.931 used into H.225.0 - this is probably impractical and unnecessary. However, we need to make every effort to be completely clear what (messages, fields, times, etc) are required, optional, and forbiddden. This work has been extended in this version.

e)The two areas that need the most work are the Q.931 and the ASN.1. Would anyone be willing to volunteer to compile the ASN.1??? Gary and I will continue to try to work the Q.931 issues. Still looking for a volunteer!

Minor Issues:

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

INTERNATIONAL TELECOMMUNICATION UNION

ITU-T
H.225.0

TELECOMMUNICATION
STANDARDIZATION SECTOR
1995
OF ITU

DRAFT

(January 15~~2~~, 1996)
Determined November

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.

H.225.0 makes use RTP/RTCP(Real Time Protocol/Real Time Control Protocol) for media steam packetization and synchronization for all underlying LANs (See Annex A, B, and C). 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 transport address~~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 connections. H.225.0 contains details on the usage of RTP/RTCP.

~~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.~~ In H.221, audio, video, data, and control are multiplexed into one or more synchronized physical SCN calls. On the LAN side of an H.232 call, none of these concepts apply. There is no need to carry fromthe SCN side the H.221 concept of a P*64 Kbbs call, e.g. 2 by 64 Kbps, 3 by 64 Kbps, 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 384 kbps interworking with 6 * 64 kbps on the WAN side[2]. The primary rationale of this approach is to put complexity in the gateway rather than

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

[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. Stated

the terminal and to avoid extending onto the LAN features of H.320 that are tightly tied to ISDN unless this is 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 uses~~ing~~ 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, and shall not exceed these rates.

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.

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another way we would normally expect that, using H.245 flow control, the LAN/CSN gateway will force the video and data rates to fit into the H.221 CSN multiplex. However, since audio may be transcoded in the gateway often, we will frequently find that the LAN audio rate and the CSN rate do not match. Also there should be no expectation that the H.221 bit rate for control (800 bps) will generally match the H.245 bit rate on the LAN side. 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.

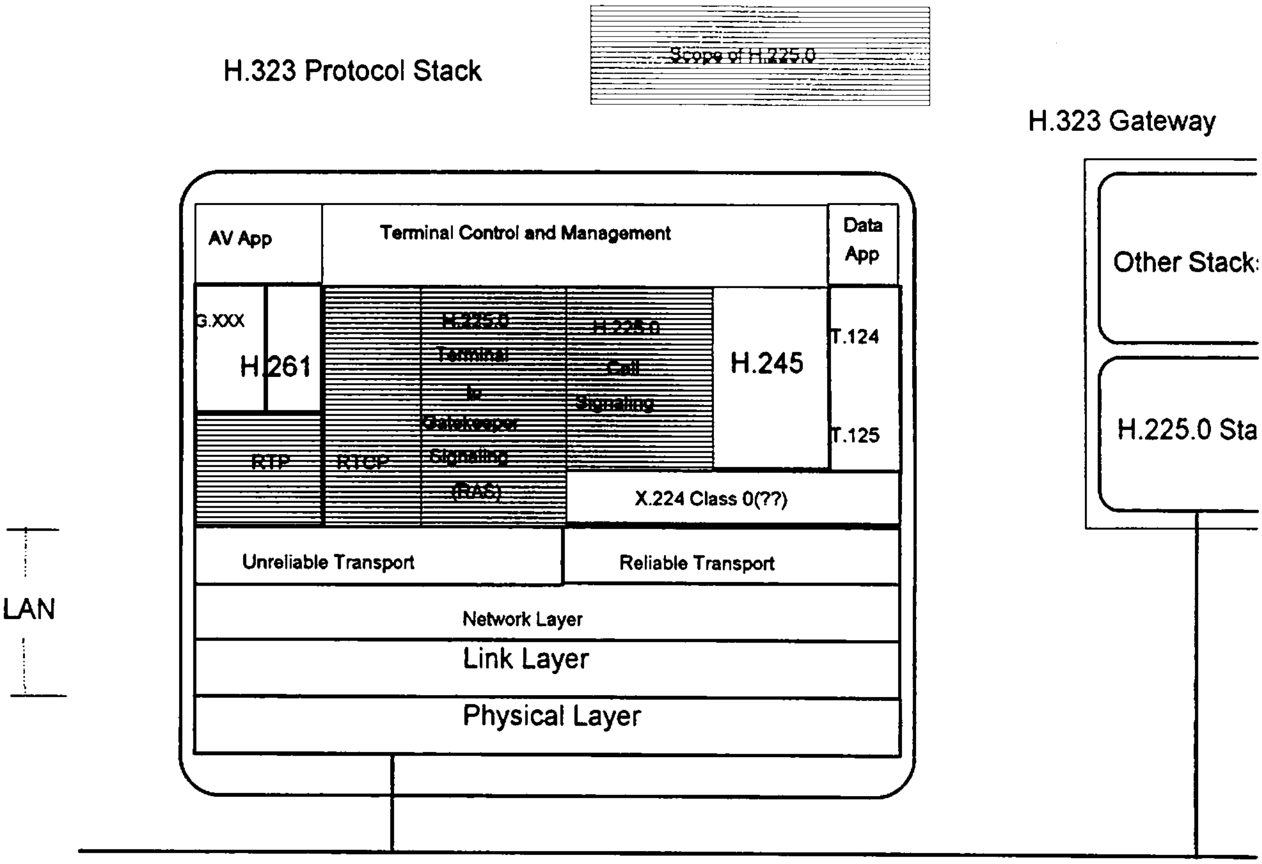

Figure 1/H.225.0
Scope of H.225.0

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.931can be sent over reliable links provided by the LAN.
- The f~~F~~lexiblity 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.

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.*

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. ITU-T Recommendation H.321 (1995): "Adaptation of H.320 Visual Telephone Terminals to B-ISDN Environments".

14. ITU-T Recommendation H.322 (1995): "Visual Telephone Systems and Terminal Equipment for Local Area Networks which Provide a Guaranteed Quality of Service".

15. ITU-T Recommendation H.324 (1995): Terminal for Low Bitrate Multimedia Communications".

16. ITU-T Recommendation H.310 (1996): "Narrow-band ISDN visual telephone systems and terminal equipment".

~~13. H.324.~~

~~14. H.321~~

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

~~15. H.322~~

~~16. H.310~~

17. Q.931 add reference

18~~6~~. Q.932 add reference

3. Definitions

See new definitions in H.323.

4. Conventions

In this document, "shall" refers to a mandatory requirement, while "should" refers to a recommended but optional feature or procedure. The term "may" refers to either something which logically follows as a result of a circumstance, or to a set of alternatives that exist where neither one is required or suggested.

When a term such as "MCU" is used, an H.323 MCU is referred to. If an H.231 MCU is intended, this will be explicitly noted.

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	Internet Protocol~~Add~~
LAN	Local Area Network
LD-CELP	Low Delay - Code Excited Linear Prediction
LSB	Least Significant Bit
LSD	Low Speed Data
MB	Macro Block (see H.261)
MBE	Multi-Byte Extension
MCC	Multipoint Command Conference
MCN	Multipoint Command Negating
MCS	Multipoint Command Data Symmetrical Data Transmission

MCU	Multipoint Control Unit
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 Protocol
VCF	Video Command "Freeze Picture Request"
VCU	Video Command "Fast Update Request"
UDP	Unicast Data Protocol

6. Packetization and Synchronization Mechanism

6.1.General Approach

Before any calls are made, an endpoint may bind/register with an associated gatekeeper. If this is the case, it is desirable for the endpoint to know the vintage of the gatekeeper it is regis~~si~~tering with. It is also desirable for the gatekeeper to know the vintage of endpoints that register with it. For these reasons, both the binding and registration sequences contain an H.245 style OBJECT IDENTIFIER that allows the vinatge to be determined in terms of the version of H.323 implemeted. This sequence also may contain optional non-standard message parts ~~in the style of H.245~~ to allow endpoints to establish non-standard relationships. At the end of this sequence, both gatekeepers and endpoints are aware of version numbers and the non-standard status of each other.

The version number is mandatory and non-standard information is optional in the Setup/Connect sequence described below to allow two endpoints to inform each other of their vintage and non-standard status.

The general approach starting a call is to send a mandatory admission request[7], followed by an initial ***Setup*** Message on a reliable channel transport address (this address may have been returned in the admission confirmation message, or ~~which~~ may have been ~~was~~ known to the calling terminal) . ~~to a terminal (which may also be an H.323 gateway) or to a~~

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[7] A terminal that is not regisitered with a~~bound to a~~ gatekeeper is not required to send an admissions request.

~~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 reliable transport address~~port~~ on which to send H.245 control messages.

Once the~~this~~ reliable H.245 control channel ~~control Transport Port~~ has been established, additional channels~~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.[8] This negotiation is performed per media, in the sense that, for example, audio/video may be distributed, while data and control are centralized.

H.225.0 teminals shall be capable of sending audio and video using RTP via unreliable channels ~~links~~ 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 would result in excessive delay in the LAN environment.[9] However, an optional mode where retransmission is supported can be signaled via H.245[OPEN ISSSUE #1]. It is assumed 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 signaling~~setup~~/H.245 control are never sent on the same ~~transport~~ channel, and do not share a common message structure. H.225.0 terminals shall be capable of sending and receiving audio and video on separate transport addresses~~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 packets are mixed in a single frame which is sent to a single transport address ***{Editor's Note: I await the detailed description of this mode of operation; perhaps more will be added here***}[OPEN ISSUE #2]

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 Q.931 signaling. T.120 shall be conveyed over the LAN between endpoints~~, or between the endpoint and the gateway~~ on another t~~T~~ransport address~~Port~~ . Thus, the typical point-to-point, or point-to-gateway link can be expected to have at least seven and up to 9 TSAP identifiers~~transport ports~~ if T.120 is used. Note that T.120 may require up to four connections using the T.120 TSAP identifier~~port~~ for all four MCS priority levels per T.120 conference the terminal is participating in. Table 1/H.225.0 shows the number of TSAP identifiers~~transport ports~~ used for each media on a point-to-point call~~conference~~. It is also true that a given H.323 terminal may be able to participate in more than one conference at a time, resulting in the use of additional TSAP identifiers~~ports~~. All H.245 logical channels used are uni-directional.***[OPEN ISSUE #3]***~~***{Editor's Note: Have we properly taken account of the bi-directional vs uni-direcitonal operations???}***~~

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[8] 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].

[9] 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.}**

#of TSAP Ids ~~ports~~	Usage	Reliable or Unreliable	Well Known or Dynamic	# of Connections
1.	Audio/RTP	unreliable	dynamic	1
1	Audio/RTCP	unreliable	dynamic	1
1	Video/RTP	unreliable	dynamic	1
1	Video/RTCP	unreliable	dynamic	1
1	Call Signaling~~/H.225.0~~	reliable	well known	1
1	H.245	reliable	dynamic	1
1	Data (T.120)]	reliable	dynamic	up to 4
1	RAS~~Terminal to GK Admissions~~	unreliable	well known	0?~~1 (more for GK)~~

Table 1-A/H.225.0
TSAP IDs~~Transport Ports~~ used by H.225.0 per point-to-point call~~conference~~

Although the transport address for, say, audio and video, may share the same LAN address and differ only by TSAP identifier, some manufacturers may chose to use different LAN addresses for audio and video. The only requirement is that the convention of Annex A/B be followed in the numbering of TSAP identifiers within each RTP session ***{Editor's note: Do we agree on this?? How might this apply to Figures 1-B and 1-C??}***

Table 1-A/H.225.0 describes the basic case of point-to-point operations between two terminals. In the above case, the use of a reliable well known port for call signaling prevents the reception of other calls using that well known port at the same time; an unacceptable situation for gateways, MCUs, or gatekeepers. The following tables illustrate port usage for the gateway/MCU case, and for the gatekeeper case.

#of TSAP Ids ~~ports~~	Usage	Reliable or Unreliable	Well Known or Dynamic	# of Connections
1.	Audio/RTP	unreliable	dynamic	1
1	Audio/RTCP	unreliable	dynamic	1
1	Video/RTP	unreliable	dynamic	1
1	Video/RTCP	unreliable	dynamic	1
1	Call Signaling~~/H.225.0~~	reliable	dynamic	1
1	H.245	reliable	dynamic	1
1	Data (T.120)]	reliable	dynamic	up to 4
1	RAS~~Terminal to GK Admissions~~	unreliable	well known	0??

Table 1-B/H.225.0
TSAP IDs used on one MCU/Gateway Port

#of TSAP Ids ~~ports~~	Usage	Reliable or Unreliable	Well Known or Dynamic	# of Connections
1	Call Signaling/~~H.225.0~~	reliable	dynamic	1
1	H.245	reliable	dynamic	1
1	RAS~~Terminal to GK Admissions~~	unreliable	well known	0 or many?

Table 1-C/H.225.0

TSAP IDs use by H.225.0 ~~gateway~~ GK per endpoint supported using the gatekeeper ~~the~~ mediated call model of Figure 11/H.323

Note that a well-known reliable transport address~~port~~ is used for ~~initial~~ call setup for the terminal to terminal case, and also for the gatekeeper mediated case. The reliable call signaling connection~~port~~ shall be kept active according to the following rules:.

1. For terminal to terminal call signaling (Figure 9~~X~~/H.323), either terminal may chose to close the reliable call signaling channel, or to leave it open.

2. For the gatekeeper mediated call signaling case (Figure 8~~Y~~/H.323), the terminals shall keep the reliable port active throughout the call. However, the gatekeeper may chose to close this signaling channel for calls that do not involve gateways. The gatekeeper shall keep the reliable signaling channel open throughout the call if a gateway is involved to allow the end-to-end transmission of Q.931 information elements such as display information. ***{Editor's Note: one possible issue is that an H.320 terminal may be added later, requiring the reliable channel to be re-opened. It is important to keep the channel open to allow Q.931 information elements such as display lines to be transported end-to-end. It is possible that the best course is simply to have the gatekeeper keep the channel open at all times for all calls}*** ***[OPEN ISSUE #4]***

3. If for some reason the reliable link become inactive via a transport level failure or other problem, the link shall be re-opened, and the call shall not be dropped. {Editor's Note: Add material from Toga document here}

Note that more than one H.245 channel~~connection~~ may be open at a given time, e.g. an endpoint ~~terminal~~ may be in more than one call/conference~~conference~~at the same time. Note also that within a specific call~~conference~~, a terminal may have more than one channel of the same type open, e.g. two audio channels for stereo audio. The only limitation is that there shall be one and only one H.245 control channel in each direction per point-to-point call~~conference~~.

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. As part of the process of opening the channel, before sending the open logical channel acknowledgement the endpoint uses the ARQ/ACF sequence to assure that sufficient bandwidth is available for the new channel (unless sufficient bandwidth is available from a previous ARQ/ACF sequence).

OPEN ISSUE #5: IS THE ABOVE SUFFICIENT FOR AUDIO??

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 proppposed 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. 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? It has been noted that this procedure may not be adequate and that the MC may need to know the difference between terminals and gateways. When we resolve this issue text will be added here.}***

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 messages 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 H.245 capability signaling.

6.2. Use of RTP/RTCP

The H.225.0 terminal shall use as a default a separate pair of TASP IDs~~Transport Port~~ for audio/video and the associated RTCP channel~~video~~ as described in Annex A and B. ~~On each Transport Port a separate instance of RTP shall be used, one for audio and one for video. Additional Transport Ports are needed for RTCP, one for each RTP Transport Port.~~ Optionally, endpoints may chose to use different LAN addresses for audio and video, but for each LAN address the convention of Annex A/B shall be followed in the use of TSAP Ids. Using H.245 signaling, additional audio and video channels (2 TSAP IDs~~Transport Ports~~ per channel, one for RTP and one for RTCP) may be established if the terminal supports this capability.

An optional capability to use a single transport address~~port~~ for both audio and video may be signaled via H.245

~~In this section two documents, 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 RTP in Annex B. Implementations shall follow the RTP profile (Annex B)~~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~~~~***{Editor's Note: I have received a comment that this is not possible or desirable. We need to look at this further}***~~~~, 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}***~~

RTP translators and mixers are not elements of the H.323 system, and any information about them in Annex A/B shall be regarded as informative. Note that both gateways and MCUs have some aspects of both mixers and translators, and the information in Annex A/B should be helpful in the implementation of gateways and MCUs.

Version (V): Version 2 of RTP shall be used

SSRC:[OPEN ISSUE #6] ***{Editor's Note: some of the complexity of RTP derives from the the use of a randomly selected SSRC. Since all H.323 calls have a point-to-point reliable H.245 control link, this link could be used to negotiate SSRCs that are unique, and allow the removal from Annex A of the SSRC collision detection material. This***

was pointed out to me by Steve Casner, RTP co-editor. I suggest we give this serious consideration, as it would simplify implementations. One solution is to use a 4 octet CRV as the 4 octet SSRC. This is a bigger CRV than we probably need, and we would need to increment the CRV for each RTP session, but it is not clear that the CRV needs to increase in a strictly monotonic fashion. Since the CRV is known to the far end, and is exchanged prior to the start of all RTP sessions, collisions would never occur. Note that MCUs/Gateways generate their own CRVs, but gatekeepers would be constrained to leave them unaffected.}

CSRC Count(CC): Use of the CSRC count in H.225.0 is optional. When not in use, the value of CC shall be zero(0). The CSRC may be used by MCUs to provide information on contributors to the audio sum when distributed audio processing is occuring. ***[OPEN ISSUE #7 - Is there any need for H.245 level caps?]***

CNAME: [OPEN ISSUE #8]In the simplist case of a point-to-point connection on the LAN, the SSRC is used to sychronize the received audio/video from a terminal, and the two streams are associated by a CNAME as specified in Annex A. In classical RTP (see Appendix A), the CNAME is a domain name. However, classical RTP has no method to support an endpoint that makes use of more than one CNAME at the same time, e.g. if two pairs of audio/video are supplied from a single endpoint at the same time, perhaps making use of two cameras. However, support of this case is desirable.

Thus, H.323 requires a CNAME that has a part which is unique per endpint within a conference, a part that is unique between conferences to handle the case of the endpoint that is participating in more than one conference at the same time, and also has a third part which is incremented on a per source per conference basis. These three parts are:

1)MC assigned terminal number of the form <M><T>, which is assured to be unique within a conference,

2)The H.245 terminal identifier[10], H323_ID, or domain name believed by the user to be unique ***{Editor's Note: The E.164 address is another possible choice}***, and,

3)A source number that is incremented per source pair starting at one.

{Editor's Note: this requires confirmation}

When using RTCP, both RR and SR packets should be sent periodically as described in RTP. Only the CNAME SDES message[11] shall be used. Other SDES messages (see Annex A) are forbidden as being duplicative of H.245 and T.120 level conference control functions.

{Editor's Note: this requires confirmation}

The H.323 LAN terminal, when engaged in ~~a~~any conference, whether point-to-point or multipoint, 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.

---

[10] This terminal identifier is the same as the H.230 terminal identifier, and follows the same rules.

[11] MCUs may change CNAMEs during conferences.

- 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

Before considering the how audio is packetized using RTP, we must consider how it is signaled via H.245, and the relationship of this signaling to RTP. In gemeral, when the audio channel is opened, an H.245 logical channel is opened. H.245 signaling in the **AudioCapability** strucuture is given in terms of the maximum number of frames per packet. The frame size for H.225.0 varies with the audio coding in use. ~~is defined below.~~

All H.323 terminals offering audio communication shall support G.711. For all frame-oriented audio codecs and receivers shall signal the maximum number of audio frames they are capable of accepting in a single audio packet. Transmitters may send any whole number of audio frames in each packet, up to the maximum stated by the receiver. Transmitters shall not split audio frames across packets, and shall send whole numbers of octets in each audio packet.

Note: Sample based codecs, such as G.711 and G.722~~,~~ shall be considered to be frame-oriented, with a frame size of sample one.

For audio algorithms such as G.723 which use more than one size of audio frame, audio frame boundaries within each packet shall be signaled in-band to the audio channel. For audio algorithms which use a fixed frame size, audio frame boundaries shall be implied by the ratio of packet size to audio frame size. ***{Editor's Note: This paragraph seems questionable. G.728 has a fixed frame size of, I believe, 40 octets. G.722 has a single octet frame size. We need to discuss this further}[OPEN ISSUE #9: Frame sizes for G.728 and G.723 and G.729]***

The following discusses the use of RTP once H.245 signaling is complete.

~~**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. 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] shall be used. Dynamic payload types as described in Annex B shall be used for any ITU-T payload types not listed in Annex B. ~~***{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. }***~~

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.

When sending 48/56 kbit/sec G.722, the H.323 gateway shall pad the extra 1 or 2 bits in each octet, and use dynamic RTP payload types to differentiate between 64 Kbps (which uses PT= xx) and the reduced rate cases. In the reverse direction the H.323 gateway shall truncate 64 kbit/sec G.722 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.722 shall be used.

OPEN ISSUE #10*{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 message~~[12] ~~should be used, and instead of the canonical name mentioned in RTP, the H.243 terminal identifier shall be sent. Other SDES messages 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 messages

~~**Version (V):** Use RTP version 2.~~

~~**CSRC Count(CC):** This shall be set to zero for the H.323 endpoint and the gateway.~~
~~{Editor's Note: Consideration should be given to the MCU doing 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 message shall be used(other SDES messages 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.~~6.2.1.~~above. On a point-to-point call CNAME shall follow the conventions of RTP.

---

[12] MCUs may change CNAMEs during conferences.

{Editor's Note: The usage of Freeze picture in RTP and in H.245, and also picture release, must be rationalized} OPEN ISSUE #11

H.261 is packetized on the LAN side as per Annex C~~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 MB fragmented packets and well as GOB fragmented packets, or packets with a mix of MBs~~bs~~ and GOBs. 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. RTP packets used should not exceed the size of the MTU(Maximum Transportable Unit) on a given LAN to maximize robustness of operation. 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.

{Editor's Note: I have received a suggestion that 2 or more video frames should not be put into one packet? Comments?} OPEN ISSUE #12

{Editor's Note: It seems clear that a maximum decoder packet buffer size is needed to allow interoperability. If, for example I build a decoder that supports at most a 2K video GOB/MB, and you send me only 3K packets since you want to minimize packet rate, we will not interoperate at all. Perhaps this is signaled via H.245, and we need to note the procedure here, but it seems to require more attention. Comments?} OPEN ISSUE #13

{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 OPEN ISSUE #14}

6.2.3.Data messages

There are no special data messages; 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} OPEN ISSUE #14

~~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, and the two streams are associated by a CNAME as specified in RTP.~~

~~When a gateway acting on behalf of an MCU or an LMCU 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), itgateway 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 can 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 a suffix 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 Set Message~~up MESSAGE~~ Definitions

This section concerns the definition of ~~MESSAGE~~messages 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

OPEN ISSUE 16: Need to complete Q.931 messages, especially fields and timers.

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. A gateway or gatekeeper shall pass on any optional Q.931 information elements it receives to maximize terminal interoperability, as well as any optional Q.931 messages it does not support..

H.225.0 makes use of User to User informatioon in Q.931 to convey transport addresses and other information. This User to User information is encoded using ASN.1 as follows ... ***{Editor's Note: We need to consider the relationship of UUI to UNI, and also to require that the gateway terminate all ASN.1 UUI elements}***

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 completion 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.

~~***gatewaygateway***~~
The following table ~~taken from Table 3-1/Q.931~~ shows 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, F, or O)**[13]
Alerting	M
Call Proceeding	M
Connect	M
Connect Acknowledge	O~~M~~
Setup	M
Setup Acknowledge.	O(consider at Ipswich)
Call Clearing Messages	
Disconnect	F
Release	F
Release Complete.	M[14]
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***(Editor's Note: we have a proposal for four)***~~-~~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 1988 version as modified below in Table A/H.225.0, 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.

---

[13] M=Mandatory, F=Forbidden, O=Optional.

[14] Release Complete is required for any situation in which the H.225.0 reliable call signaling channel is open. If this channel is not open, H.245 session end may be used to terminate the conference.

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."

Information element	**H.225.0 status(M/F/O)**	**Length in H.225.0**
Protocol discriminator	M	1
Call reference	M	2??
Message type	M	1
Channel identification	F	NA
Facility	F(Note 1)	NA
Progress indicator	F	NA
Display	M(Note 2)	2-82
Signal	F	NA
Feature activation	F(Note 1)	NA
Feature indication	F(Note 1)	NA
User-to-User	M (Note 3)	2-131

Table A/H.225.0
Alerting

Note 1: The use of these fields may be added as part of our discussion of Q.951, etc.
Note 2: Reception of the display IE is mandatory; actually displaying it is optional.
Note 3: The User-to-User IE contains the ~~following~~ ASN.1 shown below ~~(to be added)~~.

~~{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}~~ There has been no proposal to add these things back in, but the one that seems most important to me is the channel number which might indicate which one of several calls being made as part of the setup this is in response to. We need to think further on this multiple call with one setup approach method. OPEN ISSUE 17}

Also, in the User-user field the following information shall be provided in the ASN.1 User-to-User IE:

```
Alerting_UUIE                ::=SEQUENCE
{
        protocolIdentifier   OBJECT IDENTIFIER,                      -
MANDATORY
                             -- shall be set to the value
                             -- {itu recommendation h 323 version (0) 1}
        nNonStandardData     NonStandardParameter,                   -
OPTIONAL
        destinationInfo      EndpointType,                           -
MANDATORY
        user-data            SEQUENCE,                               -
OPTIONAL
                                    {
                                    protocol-discriminator    INTEGER (0..255),
                                    user-information          OCTET STRING
                                    }
}
}
```

protocolIdentifier - set by the called endpoint to the version of H.323 supported. Shall not be set by the gatekeeper.
NonStandardData - A message from the called endpoint to the calling endpoint to inform it of any non-standard feature it may support. Shall not be set by the gatekeeper.
destinationInfo - Contains a EndpointType to allow the caller to determine whether the call involves a gateway or not.
user-Data - User to User information not specified by H.323

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}***

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 5)	4-12
Channel identification	Both	O (Note 1)	2-*
Progress indicator	Both	O (Note 2)	2-4
Display	n → u	O (Note 3)	(Note 4)
High layer compatibility	Both	O (Note 6)	2-5

TES

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

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

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

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

The Bearer capability information element is included when the procedures of 5.11 for bearer capabili ction apply. When present, progress indication No. 5, *interworking has occurred and has resulted in* *communication service change* shall also be present.

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

TABLE 3-3/Q.931
CALL PROCEEDING message content

Also, in the User-user field the following information shall be provided in the ASN.1 User-to-User IE:

```
CallProgress_UUIE                    ::=SEQUENCE
{
        protocolIdentifier      OBJECT IDENTIFIER,                     -
MANDATORY
                                -- shall be set to the value
                                -- {itu recommendation h 323 version (0) 1}
        NonStandardData         NonStandardParameter                   -
OPTIONAL
        destinationInfo         EndpointType,                          -
MANDATORY
        user-data               SEQUENCE,                              -
OPTIONAL
                                        {
                                        protocol-discriminator      INTEGER (0..255),
                                        user-information            OCTET STRING
```

}

}

protocolIdentifier - set by the called endpoint to the version of H.323 supported. Shall not be set by the gatekeeper.
NonStandardData - A message from the called endpoint to the calling endpoint to inform it of any non-standard feature it may support. Shall not be set by the gatekeeper.
destinationInfo - Contains a EndpointType to allow the caller to determine whether the call involves a gateway or not.
user-Data - User to User information not specified by H.323

7.3.1. Adaptive Busy Indication (ABI)

Will be signaled via Q.931 busy indication. This is, I believe, part of call proceeding, so more will be added here.

~~*{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.

Information element	H.225.0 status(M/F/O)	Length in H.225.0
Protocol discriminator	M	1
Call reference	M	2
Message type	M	1
Channel identification	F	NA
Facility	F(Note 1)	NA
Progress indicator	F	NA
Display	M(Note 2)	2-82
Signal	F	NA
Feature activation	F(Note 1)	NA
Feature indication	F(Note 1)	NA
User-to-User	M (Note 3)	2-131

Table C~~?~~/H.225.0
Connect

Note 1: The use of these fields may be added as part of our discussion of Q.951, etc.
Note 2: Reception of the display IE is mandatory; actually displaying it is optional.
Note 3: The User-to-User IE contains the ~~following~~ ASN.1 below~~(to be added)~~.

Also, in the User-user field the following information shall be provided in the ASN.1 User-to-User IE:

```
Connect_UUIE                ::=SEQUENCE
{
        protocolIdentifier      OBJECT IDENTIFIER,                      -
MANDATORY
                                -- shall be set to the value
                                -- {itu recommendation h 323 version (0) 1}
        H245Address             TransportAddress,                       -
MANDATORY
        nNonStandardData        NonStandardParameter,                   -
OPTIONAL
        destinationInfo         EndpointType,                           -
MANDATORY
        user-data               SEQUENCE,                               -
OPTIONAL
                                {
                                protocol-discriminator          INTEGER (0..255),
                                user-information                OCTET STRING
                                }

}
```

protocolIdentifier - set by the called endpoint to the version of H.323 supported
H245Address - 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. ***{Editor's Note: the relationship of the conference id to the H.243 conference id needs to be explored and clarified at Ipswich}{need to clarify role in ad hoc conference setup; relationship to CRV as well}***
nonStandardData - A message from the called endpoint to the calling endpoint to inform it of any non-standard feature it may support.
destinationInfo - Contains a EndpointType to allow the caller to determine whether the call involves a gateway or not.
user-Data - User to User information not specified by H.323

7.5. Connect Acknowledge

Follow Table 3-6/Q.931. ***{Editor's Note: A table as above will be added} {Is this needed in the terminal - terminal case? In any case?? What is its added value? OPEN ISSUE #18.***

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

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.

Also, in the User-user field the following information shall be provided in the ASN.1 User-to-User IE:

```
Setup_UUIE                    ::=SEQUENCE
{
        protocolIdentifier    OBJECT IDENTIFIER,                        -
MANDATORY
                              -- shall be set to the value
                              -- {itu recommendation h 323 version (0) 1}
        H245Address           TransportAddress,                         -
MANDATORY
        destinationAddress    SEQUENCE OF External Address,             -
MANDATORY
        nNonStandardData      NonStandardParameter,                     -
OPTIONAL
        destinationInfo       EndpointType,                             -
MANDATORY
        destCallSignalingAddress      TransportAddress,                 -
MANDATORY
        destExtraCallInfo     SEQUENCE OF ExternalAddress,              -
OPTIONAL
        user-data             SEQUENCE,                                 -
OPTIONAL
                                      {
                                      protocol-discriminator      INTEGER (0..255),
                                      user-information            OCTET STRING
                                      }
        ...
}
```

protocolIdentifier - set by the calling endpoint to the version of H.323 supported

H245Address - this is a specific transport address on which the calling terminal or gatekeeper handling the call would like to establish H.245 signaling. ~~***{Editor's Note: Should this be SEQUENCE OF for multiple transports???}***~~
destinationAddress - this is the address we wish to be connected to. It is suggested that either transport address be used or the E.164/H.323 ID style of address. For the direct terminal to terminal case this address is already known and the gatekeeper is not mediating the call, so this field is redundant. ***{Editor's Note: We need a proper formulation to avoid ambiguities. Should this be optional so that the GK doesn't have to sent it to the called terminal? These considerations may also apply to other fields. OPEN ISSUE #22}***
destCallSignalingAddress - needed to inform the gatekeeper of the destination terminals call signaling transport address; redundant in the direct terminal-to-terminal case but always filled in.
destExtraCallInfo - needed to make possible additional channel calls, i.e. for a 2*64 Kbps call on the WAN side. Shall only contain E.164 addresses. and shall not contain the number of the innitial channel.
nonStandardData - A message from the calling endpoint to the calling endpoint to inform it of any non-standard feature it may support.
destinationInfo - Contains a EndpointType to allow the caller to determine whether the call involves a gateway or not.
user-Data - User to User information not specified by H.323

~~***{Editor's Note: Calling number is in Q.931}***~~. ***We should have a discussion about how much info about the caller to pass in the Setup, and why we are doing it. OPEN ISSUE #21***

{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}***~~

{Editor's Note: some information for two stage dialing might be placed here, but there are dangers in making this a single-shot process; it might be more desirable to ask for another Setup. Needs further discussion. OPEN ISSUE #19}

~~***{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 Note: Requires further discussion - OPEN ISSUE #20}***

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 Transport Addresses are carried in this message. This is a strong candidate for an optional message} OPEN ISSUE #23

7.9. Release Complete

Follow Table 3-12/Q.931. The disconnect/release/release complete sequence is not used since the only added value is that a network-to-user information element can be appended to the release message. As this does not apply to the LAN environment, the single step method of sending only Release Complete is used.

```
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

        Undefined Reason        (65535),
        ...
}
```

{Editor's Note: This appears too restrictive for source routing; this must be considered. We need a complete and careful scrub of these codes, along with a comparision to the Q.931 cause values to avoid duplication. OPEN ISSUE #24}

7.10. H.225.0 RAS Message Common Parts

This section describes ASN.1 structures that are used in more than one RAS (Registration, Admission, and Status) messages. Some may also be used in the User-to-User part of the Q.931 messages.

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

The **protocolIdentifier** is exactly as in H.245. It is included as part of binding, registration and Setup/Connect to allow the parties involved to detemine the vintage implementations involved.

NonStandardData is exactly as in H.245 (see H.245/section 6; the definition is not reproduced here). This parameter is optional in the binding, registration, and Setup/Connect sequences to allow the parties involved to determine the non-standard status of the endpoints involved. A gatekeeper or gateway is not obligated to pass~~t~~ on nonStandardData it does not support or understand as this might interfere with its operations.

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

{Editor's Note: we have a comment to add subnet address}

{Editor's Note: Should we use TSAP id terminology here, or revert to the natural terminology?}

```
TransportAddress            ::=CHOICE
{
        IPAddress           SEQUENCE
        {
                transport   OCTET STRING (SIZE(4)),
                port        INTEGER(0..65535)
        },
        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)),
        NSAP                OCTET STRING (SIZE(1..20~~16~~)),
        ...

}

EndpointType~~NodeType~~            ENUMERATED
{
        Gatekeeper          (1),
        Gateway             (2),
        MCU                 (4),
        Terminal            (8),
        MC                  (16),    - shall not be set by itself
        Undefined Node      (268435456),
        ...
}
```

The **ExternalAddress** structure is meant to capture the various external address formats that reference a particular transport location on the LAN.

```
ExternalAddress               ::=CHOICE
{
        E164            OCTET STRING (SIZE(16)),
        H323_ID         OCTET STRING (SIZE(64))
        ...
}       -- as new types are defined, they can be added to the above table
```

7.11. Required Support of RAS Messages

The following table shows what RAS messages are supported by different endpoint types:

RAS Message	*Endpoint (Tx)*	*Endpoint (Rx)*	*Gatekeeper (Tx)*	*Gatekeeper (Rx)*
GRQ	O			M?
GCF		O	M?	
CRJ		O	M?	
RRQ	M			M
RCF		M	M	
RRJ		M	M	
URQ	M			M
UCF		M	M	
URJ		M	M	
GQRQ		M	M	
GQRS	M			M
GQRJ	M			M
ARQ	CM			M
ACF		CM	M	
ARJ		CM	M	
ARQ	CM			M
ACF		CM	M	
ARJ		CM	M	
SRQ		M	M	
SRR	M			M
Disengage				

7.12. Terminal and Gateway Binding Messages

~~{Editor's Comment: How many GRQs are sent? One per address? One per device? This must be clarified} One per logical endpoint. No need for seq of addresses. MCU or GW would send many~~.

Note that one GRQ is sent per logical endpoint; thus an MCU or a Gateway might send many.

```
GuardianRequest          ::=SEQUENCE --(GRQ)
{

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

protocolIdentifier OBJECT IDENTIFIER, -
MANDATORY

-- shall be set to the value
-- {itu recommendation h 323 version (0) 1}
NonStandardData NonStandardParameter, -
OPTIONAL

RA&SAddress TransportAddress, - MANDATORY
endpointType EndpointType~~NodeType~~, - OPTIONAL?

}

requestSeqNum - this is a monotonically increasing number unique to the caller. It should be returned by the called in any messages associated with this specific message.
RAS~~endpoint~~Address - this is the transport address that this endpoint uses for registration and status messages...
endpointType - this specifies the type(s) of the terminal that is registering (the MC bit shall not be set by itself).

~~**gatekeeperIdentifier** - this is a string value that is used to logically identify a called gatekeeper. It shall be initialized to a zero (0) value by the caller~~

GuardianConfirmation::=SEQUENCE --**(GCF)**
{

requestSeqNum INTEGER (1..65535),
protocolIdentifier OBJECT IDENTIFIER, -
MANDATORY

-- shall be set to the value
-- {itu recommendation h 323 version (0) 1}
NonStandardData NonStandardParameter, -
OPTIONAL

gatekeeperIdentifier OCTET STRING (SIZE(64)),
RAS~~&Sr~~Address SEQUENCE OF TransportAddress, -
MANDATORY
...

}

requestSeqNum - This should be the same value that was passed in the GRQ by the caller.
gatekeeperIdentifier - need procedures
RAS~~gatekeeper~~Address - this is an arrayof transport addresses; one for each transport that the gatekeeper will respond to. ~~This address includes local port information.~~ These are the addresses to which terminals shall send resgistration messages.

Guardian Reject ::=SEQUENCE --**(GRJ)**
{

requestSeqNum INTEGER (1..65535),
protocolIdentifier OBJECT IDENTIFIER, -
MANDATORY
-- shall be set to the value
-- {itu recommendation h 323 version (0) 1}
NonStandardData NonStandardParameter, -
OPTIONAL
gatekeeperIdentifier OCTET STRING (SIZE(64)),
rejectReason GuardianRejectReason,
...
}

GuardianRejectReason ENUMERATED
{
Resource Unavailable (1),
Terminal Excluded (2), - permission failure, not a resource failure
Invalid Revision (5),
~~...~~
Undefined Reason (65535),
...
}

7.13. Terminal and Gateway Registration Messages

Add part where GK assigs E.164 address. OPEN ISSUE #25

~~The ExternalAddress structure is meant to capture the various external address formats that reference a particular transport location on the LAN.~~

~~**ExternalAddress**~~ ~~::=CHOICE~~
{
~~E164~~ ~~OCTET STRING (SIZE(16)),~~
~~H323_ID~~ ~~OCTET STRING (SIZE(64))~~
~~...~~
} ~~-- as new types are defined, they can be added to the above table~~

RegistrationRequest ::=SEQUENCE ~~-~~**(RGRQ)**
{
requestSeqNum INTEGER (1..65535), -
MANDATORY
protocolIdentifier OBJECT IDENTIFIER, -
MANDATORY
-- shall be set to the value
-- {itu recommendation h 323 version (0) 1}
NonStandardData NonStandardParameter, -
OPTIONAL - OPTIONAL
bindRequest BOOLEAN, -
MANDATORY

CallSignalingAddress TransportAddress, -
MANDATORY
RAS~~&S~~Address TransportAddress, -
MANDATORY
terminalType EndpointType~~NodeType~~, -
MANDATORY

```
        terminalExt             SEQUENCE OF ExternalAddress,____________-
MANDATORY
________...


}
```

requestSeqNum -. this is a monotonically increasing number unique to the caller. It should be returned by the called in any response associated with this specific message.
protocolIdentifier - identifies the vintage of the calling terminal.
NonStandardData - if present, lists the non-standard capabilities of the calling terminal.
bindRequest - set to TRUE if requesting a new binding permission with called gatekeeper; set to FALSE if registering only. ***{Editor's Note the usage of this field requires further clarification***. Note that binding permission may age, and you will get a failure on re-reg and ~~they~~ you must re-bind.but you don't want to do the whole binding process.
CallSignaling~~Control~~Address - this is the transport call control address for this sendpoint.~~s terminal~~. If multiple transports are supported, they must be registered separately. This address includes local port information.***?***
RAS~~&S~~Address - this is the registration and status transport address for this endpoint.~~s terminal.~~
terminalType - this specifies the type(s) of the terminal that is(are) registering; note that the MC bit shall not be set by itself; either the terminal, MCU, gateway, or gatekeeper bit shall~~must~~ also be set.
terminalExt -This optional value is a list of external addresses, by which external (to the LAN) terminals may identify this terminal such as E.164 numbers or H323_IDs. ***If the terminalExt is null, or an E.1164 address is not present, an E.164 address may be assigned by the gatekeeper, and included in the RCF. An E.164 address from a gatekeeper shall override any previous E.164 address in use(proposed).***.- Note that multiple E.164 addresses or H323_IDs may refer to the same transport addresses.

```
RegistrationConfirmation        ::=SEQUENCE --(RGCF)
{
        requestSeqNum           INTEGER (1..65535),                     -
MANDATORY
        protocolIdentifier      OBJECT IDENTIFIER,                      -
MANDATORY
                                -- shall be set to the value
                                --  {itu recommendation h 323 version (0) 1}
        NonStandardData         NonStandardParameter,                   -
OPTIONAL
        callSignalingAddress    SEQUENCE OF TransportAddress,           -
MANDATORY

        RASAddress              SEQUENCE OF TransportAddress,           -
MANDATORY  terminalExt                  SEQUENCE OF ExternalAddress,        -
OPTIONAL
________gatekeeperIdentifier    OCTET STRING (SIZE(64)),________________- ???
        ...


}
```

requestSeqNum - This should be the same value that was passed in the RRQ by the caller.
protocolIdentifier - identifies the vintage of the accepting gatekeeper.
NonStandardData - if present, lists the non-standard capabilities of the gatekeeper.

callSignaling~~Control~~Address - this is an array of transport addresses for H.225.0 call control messages; one for each transport that the gatekeeper will respond to. This address includes the TSAP identifier.
RAS~~&S~~Address - this is an array of transport addresses for H.225.0 registration and status messages; one for each transport that the gatekeeper will respond to. This address includes the TSAP identifier.
terminalExt -***This optional value is a list of external addresses, by which external (to the LAN) terminals may identify this terminal such as E.164 numbers or H323_IDs. An E.164 address from a gatekeeper shall override any previous E.164 address being used by the endpoint. This SEQUENCE shall contain at most one E.164 address, but may contain more than one H323_ID.(proposed)***
gatekeeper identifier - needs text

```
Registration Reject              ::=SEQUENCE -(RGRJ)
{
        requestSeqNum            INTEGER (1..65535),                           -
MANDATORY
        protocolIdentifier       OBJECT IDENTIFIER,                            -
MANDATORY
                                 -- shall be set to the value
                                 -- {itu recommendation h 323 version (0) 1}
        NonStandardData          NonStandardParameter,                         -
OPTIONAL
        rejectReason             RejectReason,                                 -
MANDATORY
        ...

}
```

requestSeqNum - This should be the same value that was passed in the RRQ by the caller.
protocolIdentifier - identifies the vintage of the accepting gatekeeper.
NonStandardData - if present, lists the non-standard capabilities of the gatekeeper.
rejectReason - the reason for the rejection of the registration

{Editor's Note - we need a comment explaining some of these reasons that aren't self-explanatory. Why do we have a "duplicate bind request" when this is a registration message??? Aren't duplicate registrations valid??} **OPEN ISSUE #26**

```
        RejectReason            ENUMERATED
        {
                Not Bound Registration          (1),    - binding permission has aged
                Duplicate Registration Request  (2),    - needs comment
                Invalid Ext Num                         (3),      - ???
                Duplicate Bind Request          (4),    - need comment
                Invalid Revision                (5),
                Invalid Transport Address       (6),
                Duplicate Extension             (?),

                Undefined Reason                (65535)
        }
```

7.14. Terminal to Gatekeeper Unregistration Messages

Add from AVC-879.

7.15. Gatekeeper to Terminal Guardian Query Messages

```
Guardian Query Request          ::=SEQUENCE --(GQRQ)
{
        requestSeqNum           INTEGER (1..65535),                        -
MANDATORY
        protocolIdentifier      OBJECT IDENTIFIER,                         -
MANDATORY
                                -- shall be set to the value
                                -- {itu recommendation h 323 version (0) 1}
        NonStandardData         NonStandardParameter,                      -
OPTIONAL
        replyAddress            TransportAddress,                          -
MANDATORY
        ...


}
```

requestSeqNum -. this is a monotonically increasing number unique to the caller. It should be returned by the called in any response associated with this specific message.
protocolIdentifier - identifies the vintage of the gatekeeper sending the request
NonStandardData - if present, lists the non-standard capabilities of the gatekeeper making the request.
replyAddress - this is a transport address for the calling gatekeeper. It may be the RAS~~status & registration~~ address, or a dynamic address used only for this reply.

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

        requestSeqNum           INTEGER (1..65535),                        -
MANDATORY
        protocolIdentifier      OBJECT IDENTIFIER,                         -
MANDATORY
                                -- shall be set to the value
                                -- {itu recommendation h 323 version (0) 1}
        NonStandardData         NonStandardParameter,                      -
OPTIONAL
        gatekeeperIdentifier    OCTET STRING (SIZE(64)),                   - ???
        RASAddress              TransportAddress,                          -
MANDATORY
        callSignallingAddress   TransportAddress,                          -
MANDATORY
        ...
}
```

requestSeqNum -. shall be the value used in the guardian query requst.
protocolIdentifier - identifies the vintage of the responding terminal.

NonStandardData - if present, lists the non-standard capabilities of the responding terminal.
gatekeeperIdentifier - add procedures
RASAddress - this is the registration and status transport address for the gatekeeper of the called terminal. If the endpoint has no gatekeeper, it shall respond with the guardian query reject.
callSignallingAddress - of the gatekeeper being used by this terminal. ***{Editor's Note: Isn't this going to be confusing? It will get a message from terminal XX on a port it thought only a certain terminal was using???}***

```
Guardian Query Reject          ::=SEQUENCE -(GQRJ)
{
        requestSeqNum       INTEGER (1..65535),          - MANDATORY
        rejectReason        GuardianQueryRejectReason,   - MANDATORY
        ...
}
        GuardianQueryRejectReason          ENUMERATED
        {
                No Gatekeeper        (1),
                Invalid Revision     (5),
                Undefined Reason     (65535)
        }
```

requestSeqNum -. shall be the value used in the guardian query requst.
rejectReason - tells why the guery has failed; used to inform the requesting gatekeeper that the terminal is unregistered (has no gatekeeper).

7.16. Terminal to Gatekeeper Admission Messages

```
AdmissionRequest               ::=SEQUENCE --(ARQ)
{
        requestSeqNum       INTEGER (1..65535),                    -
MANDATORY
        callType            CallType,                              -
MANDATORY
        callModel           BOOLEAN,                               -
MANDATORY
        endpointIdentifier  OCTET STRING (SIZE(64))                -
MANDATORY
        destination         SEQUENCE of ExternalAddress,           -
MANDATORY
        CallSignalingAddress TransportAddress,                     -
OPTIONAL
        ~~replyAddress~~        ~~TransportAddress,~~                      -
~~MANDATORY~~
        bandWidth           INTEGER (1..4294967295),               -
MANDATORY
        ~~100 bit increments~~
        reserveRequest      BOOLEAN,                               -
MANDATORY
        ...


}

        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
        AnswerCall          (16),        -- answering a call
        ...
}
```

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

callType - Using this value, gatekeeper can attempt to determine 'real' bandwidth usage. The default value is (1) for all calls; it should be recognized that the call type may change dynamically during the call. and that the final call type may not be known when the ARQ is sent.

callModel - if (0) the endpoint is requesting the direct terminal to terminal call model. If (1) the endpoint is requesting the gatekeeper mediated model. The gatekeeper is not required to meet this request.

endpointIdentifier - This is an endpoint identifier that was assigned to the terminal by RCF, probably the E.163 address or H323_ID. It is used as a security measure to help ensure that this is a registered terminal within its zone.

destination~~Ext~~ - sequence of external addresses for the destination terminal, such as E.164 addresses or H323_IDs. ~~***{Editor's Note: Suppose they only have the transport address?? Where does it go???}***~~ ***The endpoint should be cautioned against supplying both a transport address(which might be out of date) and an E.164 address/H323_ID. Should we require only one or the other???}***

~~**replyAddress** - this is the transport address to which the **ACF**, or **ARJ** is to be sent. This may be the registration and status address of the requesting terminal, or it may be dynamic transport address.~~

bandWidth - the number of 100 bps requested for the bi-directional call. For example, a 128 Kbit/sec call would be signaled as a request for 256 kbit/sec. ***{Editor's Note: Does this value include only audio/video, or data + control as well? What is it's relationship to the SCN bandwidth?}***

reserveRequest - has the value 0 if the terminal is asking the gatekeeper to make a bandwidth reservation request for it but 1 if terminal is asking the gatekeeper to NOT make any bandwidth reservation for it; such a reservation may have already been made. Since the use of this bit is for further study, in this version of H.225.0 it shall be set to (1).

~~XXXX This is as far as I updated the address names to R&S, CallControl & H245.~~

```
Admission Confirmation        ::=SEQUENCE --(ACF)
{
        requestSeqNum         INTEGER (1..65535),              - MANDATORY
        bandWidth             INTEGER (1..4294967295),         - MANDATORYRYT
        callModel             BOOLEAN.                         - MANDATORY
        replyAddress          TransportAddress,
        ...
}
```

requestSeqNum - This shall be the same value that was passed in the **ARQ** by the caller.

bandWidth - the allowed maximum bandwidth for the call; may be less than that requested.

callModel - tells terminal whether call signaling sent on replyAddress goes to a gatekeeper or to a terminal. A value of (0) indicates that call signaling is being passed via the gatekeeper, while a (1) indicates that the endpoint-to-endpoint call mode is in use.

replyAddress - the transport address to send Q.931~~H.225~~.0 call signaling; uses the reliable well known(?) or dynamic port port, but may be an endpoint or gatekeeper address depending on the call model in use.

Admission Reject ::=SEQUENCE **--(ARJ)**

{

requestSeqNum	INTEGER (1..65535),	- MANDATORY
rejectReason	AdmissionRejectReason,	- MANDATORY

...

}

AdmissionRejectReason ENUMERATED ***{Editor's Note: requires revision; there has been a proposal for more codes}***

{

Not Bound	(1),	- isn't this really "not registered"???
Invalid Permission	(2),	- needs explanation
Request Denied	(4),	- no bandwidth available
Undefined Reason	(65535),	

...

}

requestSeqNum - This shall be the same value that was passed in the **ARQ** by the caller.	
rejectReason - reason the bandwidth request was denied	

7.17. Terminal to Gatekeeper Requests for Changes in Bandwidth

~~***{Editor's Note - the proposal on the table is that we rely on the open logical channel process to ensure that everyone in the conference has actually changed BW as needed; i.e. before responding to an open logical channel at a new rate, the terminal would the issue a BRQ and wait for a response from its gatekeepe, This seems to assure full coordination in all types of conferences, at the price of slowing things down a bit.}***~~

BandwidthRequest ::=SEQUENCE **--(BRQ)**

{

requestSeqNum	INTEGER (1..65535),	- MANDATORY
endpointIdentifier	OCTET STRING (SIZE(64)),	- MANDATORY
ConferenceID	INTEGER (1.. 4294967295),	-MANDATORY???
callType	CallType,	
~~replyAddress~~	~~TransportAddress,~~	
bandWidth	INTEGER (1..4294967295)	- MANDATORY

...

}

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

.endpointIdentifier - This is an endpoint identifier that was assigned to the terminal by RCF, probably the E.163 address or H323 ID. It is used as a security measure to help ensure that this is a registered terminal within its zone.
Conference ID - ID of the~~at~~ call that is to have the bandwidth changed. ***{Editor's Note: there certainly must be an ID here to allow association of the request with the call, however the details depend on how ad hoc conferencing is resolved, and also on what we determine the relatiionship of the CRV to the Conference ID is. It may be that they are one and the same. We may also wish to change this to call Id in accord with our new terminology.} This may seem odd, but suppose the terminal had two calls in the same conference. It is possible the BRQ should contain the CRV, not the conferenceID.***
callType - Using this value, gatekeeper can attempt to determine 'real' bandwidth usage.
~~**replyAddress** - this is the transport address to which the **BCF**, or **BRJ** is to be sent. It may be the registration and status address, or it may be a dynamic address.~~
bandWidth - the NEW number of 100 bps increments requested for the call

Bandwidth Confirmation ::=SEQUENCE ~~-~~-(**BCF)**

{

requestSeqNum	INTEGER (1..65535),	- MANDATORY
bandWidth	INTEGER (1..4294967295)	- MANDATORY

...

}

requestSeqNum - This should be the same value that was passed in the **BRQ** by the caller.
bandWidth - the maximum allowed at this time in increments of 100 bps.

Bandwidth Reject ::=SEQUENCE ~~-~~-(**BRJ)**

{

requestSeqNum	INTEGER (1..65535),	- MANDATORY
rejectReason	BandRejectReason,	- MANDATORY
allowedBandWidth	INTEGER (1..4294967295)	- MANDATORY

...

}

BandRejectReason ENUMERATED ***{Editor's Note: Requires expansion?}***

{

Not Bound	(1),	- binding permission has aged
Invalid ConferenceID	(2),	- possible revision
Invalid Permission	(3),	- true permission viiolation
Insufficient resources	(4),	- ~~why needed????~~
Invalid Revision	(5),	
Undefined Reason	(65535),	

...

}

requestSeqNum - This should be the same value that was passed in the **BRQ** by the caller.
rejectReason - the reason the change was rejected by the gatekeeper.

allowedBandWidth - the maximum allowed at this time in increments of 100 bps.

7.18. Disengage Messges

Add messages here to tell the GK that the call is being dropped. Note that all BW can be released but the call not dropped.

7.19. Status Request Messages

~~Support for (SRQ/SRR) is mandatory.~~

{Editor's Note: A conferenceID has been added to allow the request to be made for information about a particular conference. However, this raises questions about how SRQ is to be used. Options seem to be:

1)SRQ does not have a conferenceID(or when the ID is zero), the terminal responds with an SRR for each conference it is on.

2)We have a new message that requests a list of conferenceIDs, and then the Ids can be used for further queries.

In any case, although this is not a big issue, we should probably think about it a little more. Also, the conference id may become the call id, or otherwise be changed. OPEN ISSUE #27}

Add push from terminal for efficiency.(Toga idea).

Add a give me a list command to support spirit of message.

Add timestamp to the message so that the octets sent is meaningful.

Status Request ::=SEQUENCE –(SRQ)
{

requestSeqNum INTEGER (1..65535), - MANDATORY
conferenceID INTEGER (1..4294967295), - under consideration

}

requestSeqNum - this is a monotonically increasing number unique to the caller. It should be returned by the called in any messagePDUs associated with this specific messagePDU.
Conference idID - ID of that call that the query is about. ***{Editor's Note: there certainly must be an ID here to allow association of the request with the call, however the details depend on how ad hoc conferencing is resolved, and also on what we determine the relatiionship of the CRV to the Conference ID is. It may be that they are one and the same. We may also wish to change this to call id in accord with our new terminology.}***

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

requestSeqNum INTEGER (1..65535), - MANDATORY
EndpointType~~nodeType~~ EndpointType~~NodeType~~, - MANDATORY
conferenceID INTEGER (1..4294967295), - under consideration

(needs revision)

originator	BOOLEAN,	- MANDATORY
origAudioAddress	SEQUENCE OF TransportAddress,	- MANDATORY
origVideoAddress	SEQUENCE OF TransportAddress	- MANDATORY
origDataAddress	SEQUENCE OF TransportAddress	- MANDATORY
origH245Address	TransportAddress	- MANDATORY
origCallSignalingAddress	TransportAddress	- MANDATORY
origRASAddress	TransportAddress	- see definition
destAudioAddress	SEQUENCE OF TransportAddress,	- MANDATORY
destVideoAddress	SEQUENCE OF TransportAddress	- MANDATORY
destDataAddress	SEQUENCE OF TransportAddress	- MANDATORY
destH245Address	TransportAddress	- MANDATORY
destCallSignallingAddress	TransportAddress	- MANDATORY
destRASAddress	TransportAddress	- see definition
callType	CallType,	- MANDATORY
bandWidth	INTEGER (1..4294967295),	- MANDATORY
octetsSent	INTEGER (1..4294967295),	- needs comment
octetsRcvd	INTEGER (1..4294967295),	- needs comment

needs possible timestamp for octets sent/received

~~*{This has been added}*~~

...

}

requestSeqNum - this shall contain the sequence number from the SRR or zero for an unsolicited report to the gatekeeper.
EndpointType~~nodeType~~ - provides information about the endpoint.
Conference id~~ID~~ - ID of that call that the query is about. ***{Editor's Note: there certainly must be an ID here to allow association of the request with the call, however the details depend on how ad hoc conferencing is resolved, and also on what we determine the relatiionship of the CRV to the Conference ID is. It may be that they are one and the same. We may also wish to change this to call id in accord with our new terminology.}***
originator ~~-~~if 1~~0~~, the endpoint being queried was the call originator, if 0~~1~~ the endpoint was the call destination.
origAudioAddress - provides the transport addresses, which may differ only by TSAP identifier, of each audio stream to which the destination terminal is sending audio to the call originator.
origVideoAddress - provides the transport addresses, which may differ only by TSAP identifier, of each video stream to which the destination terminal is sending video to the call originator.
origDataAddress - provides the transport addresses, which may differ only by TSAP identifier, of each data stream to which the destination terminal is sending data to the call originator.
origH245Address - provides the transport address to which the destination terminal is sending H.245 control messages to the call originator. Note that this address may be on the gatekeeper or on the originating terminal.
origCallSignaling~~Control~~Address - provides the transport address to which the destination terminal is sending H.225.0 call control messages to the call originator (note that this address may be on the gatekeeper or on the originating terminal.
origRA&SAddress - provides the transport address to which the gatekeeper of the originating terminal is sending R&S messages; included only if this is the originating terminal.
destAudioAddress - provides the transport addresses, which may differ only by TSAP identifier, of each audio stream to which the originating terminal is sending audio to the destination terminal.
destVideoAddress - provides the transport addresses, which may differ only by TSAP identifier, of each video stream to which the originating terminal is sending video to the destination terminal.

destDataAddress - provides the transport addresses, which may differ only by TSAP identifier, of each data stream to which the originating terminal is sending data to the destination terminal.
destH245Address - provides the transport address to which the originating terminal is sending H.245 control messages to the destination terminal. Note that this address may be on the originating terminal's gatekeeper or on the destination terminal.
destCallSignaling~~Control~~Address - provides the transport address to which the originating terminal is sending H.225.0 call control messages to the destination terminal (note that this address may be on the originating terminal's gatekeeper or on the destination terminal.
destRA&SAddress - provides the transport address to which the gatekeeper of the destination terminal is sending R&S messages; included only if this is the destination terminal.
callType - provides information on call topology.
bandwidth - current usage in increments of 100 bps. ***{Editor's Note - we have to decide what this includes}***
octetsSent - needs explantion, needs timestamp???
octetsReceived - needs explanation

8. Mechanisms for maintaining QOS

~~{Editor's Note: It was agreed to do the following:~~

~~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

Transport QOS(Quality of service) on a LAN includes such characteristics as:

- Bit error rate.
- Packet loss rate.
- Delay.

Any transport 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 able 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.

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.
- A general rise in delay due to LAN congestion over time such that a feedback based mechanism is useful.

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.

Bit errors on a LAN generally are either corrected at a lower layer, or result in packet loss, so they are not considered further in this section.

Packet loss requires the receiver to be able to compensate for lost packets in a fashion that conceals errors to the maximum possible extent. For data and control, retransmission at the transport layer is used. For audio and video retransmission is optional.

A given level of transport QOS results in a level of user-perceived audio/video QOS that is a function in part of the effectiveness of the methods used to overcome transport QOS problems.

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 OPEN ISSUE #28}

Four~~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 Annex A~~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 8.4~~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 should~~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 Ipswic. Add Reid text.~~h~~}

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.

It should be noted that responding to interarrival jitter in a multi-router path where a large percentage of packets arrive out of order is difficult. It may be impossible to distinguish this source of jitter from other sources, or to base error recovery strategy on measured jitter. However, packet loss is quantifiable and unambiguous.

8.6 Echo Control

Control of acoustic echo is the responsiblity of the H.323 terminal. In general, given the delay involved in video/audio compression, its is assumed that all H.320/H.323/H.324 terminals have some form of echo control (cancellation or switching).

However, when the H.323 terminal is used to call a GSTN telephone, it is typically that case that the GSTN phone does not support echo control. Thus, the user of the H.323 terminal may hear acoustic echo return from the GSTN side. This acoustic echo return can be minimized by the use of a speakerphone with echo control, or the use of a handset or ear phones. Manuafacturers should(??) add loss (how much??) to the audio path when an H.323 teminal is connected to a GSTN POTS phone.

Control of hybrid (2 to 4 wire) echo is the resonsibility of the H.323 gateway.

~~*{Editor's Note: Is this what you had in mind? What's missing??}*~~

9. Annex A: RTP/RTCP(Normative)[See AVC 892]

10. Annex B: RTP Profile(Normative)[See AVC 894]

11. Annex C: H.261 Packetization(Normative)[See AVC 893]

12. Appendix A: RTP/RTCP(Informative)

See RFC 1889, "Real Time Transport Protocol" for the referenced informative material.

13. Appendix B: RTP Profile(Informative)

See RFC 1890, "RTP Profile for Audio and Video Conferences with Minimal Control" for the referenced informative material.

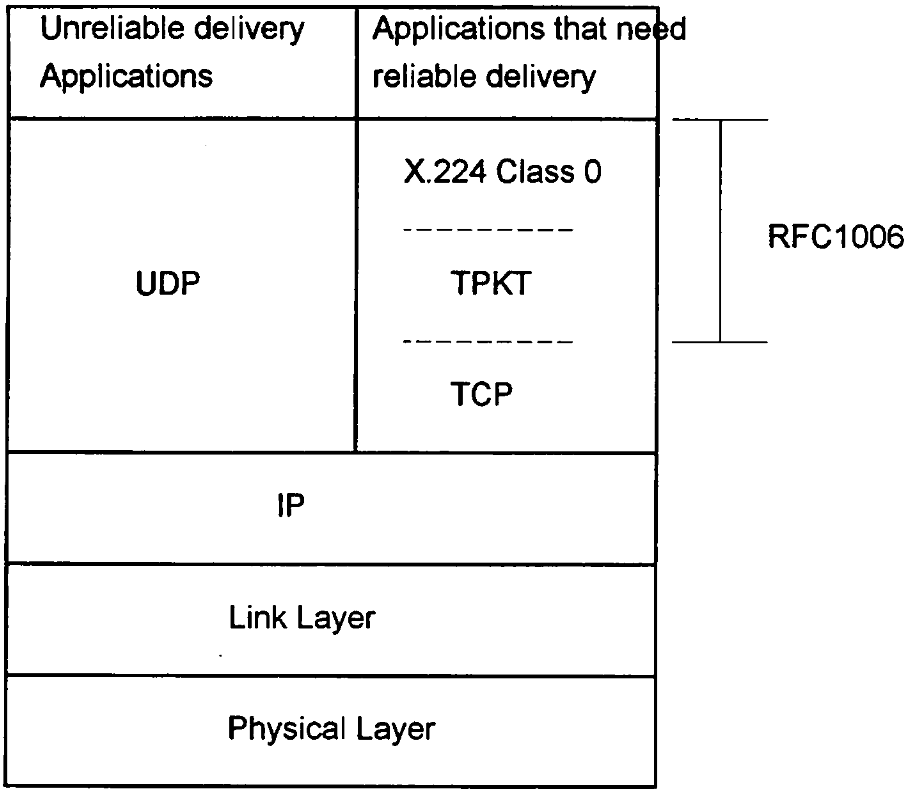
14. Appendix C: H.261 Packetization(Informative)

See RFC XXXX, "RTP Payload Format for H.261 Video Streams~~H.261 Packetization~~" for the referenced informative material.

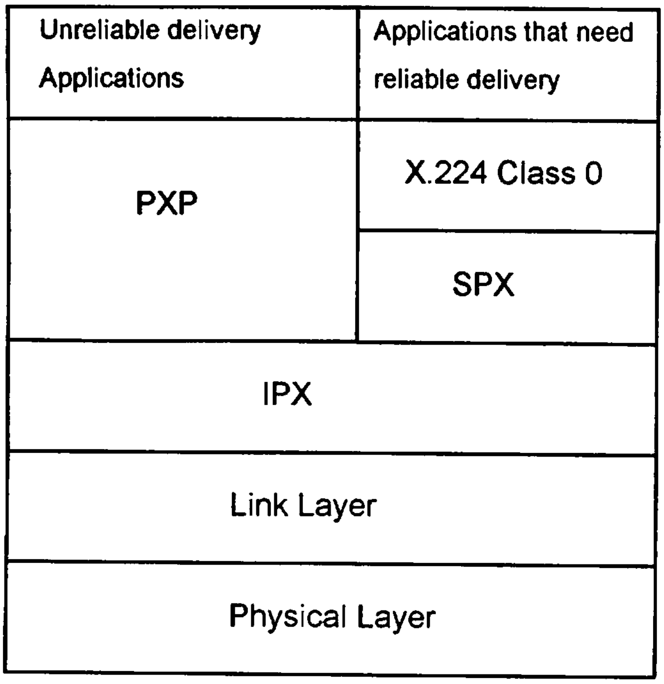
15. Appendix D (Informative)

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

15.1. TCP/IP/UDP


Unreliable delivery Applications
Applications that need reliable delivery
UDP
X.224 Class 0
TPKT
TCP
RFC1006
IP
Link Layer
Physical Layer

15.2. SPX/IPX



~~15.3. AppleTalk (Could someone provide this???)~~