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ATM and Other Network Environments

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STUDY GROUP 15
CONTRIBUTION

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Title: H.22Z, Media Stream Synchronization and Time Base recovery on Non-Guaranteed Quality of Service LANs

Date: October 12, 1995~~August 19, 1995~~

Summary: 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.

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 (rev 3) reflects changes from the July 31 version(rev 2 after Haninge).. It is the thirdversion to attempt to make use of RTP/RTCP more fully. This version reflects a considerable number of comments that have been received from many sources, and also attempts to incorporate the gatekeeper/gateway message sets from the PictureTel proposal. As mentioned in earlier mail, there appears to be one outstanding complaint with RTP/RTCP (other than its possible instabiltiy), and this is a call to allow for other similar protocols. Other comments have been generally supportive of the current direction, so pending a more complete proposal, I have continued to develop H.22Z in its current direction. Most of the material added in this version does not relate to RTP, and would be valuable even if we adopt this proposal to put RTP beneath a generalized interface.

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.22Z 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.

INTERNATIONAL TELECOMMUNICATION UNION

ITU-T

TELECOMMUNICATION
STANDARDIZATION SECTOR
OF ITU

DRAFT H.22Z

(August 19 95)

LINE TRANSMISSION OF NON-TELEPHONE SIGNALS

Media Stream Synchronization and Time Base recovery on Non-Guaranteed Quality of Service LANs

DRAFT ITU-T Recommendation H.22Z

Media Stream Synchronization and Time Base recovery 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.

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.22Z 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

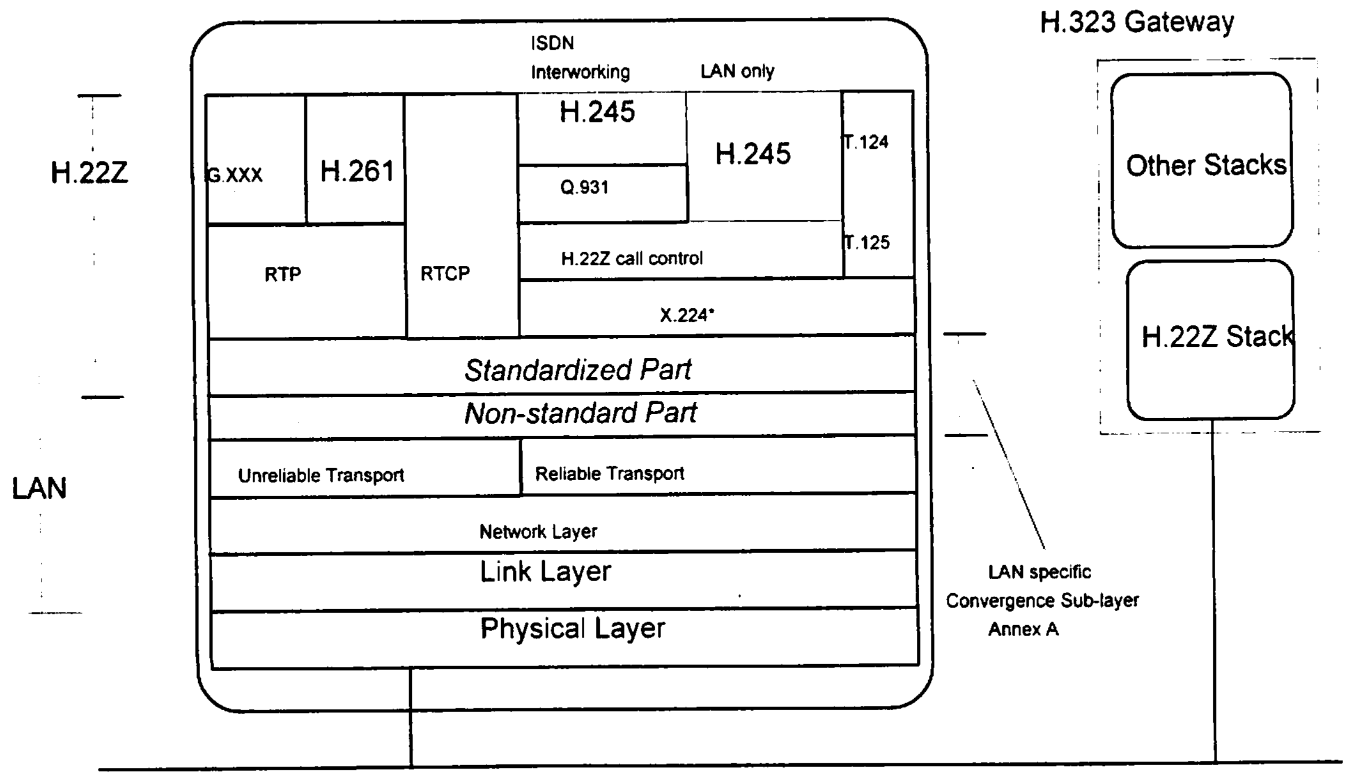
1)extend the transfer rate back to the H.323 terminal. For example, this implies signaling a 2*64 call differently than a 128 kbps call, since they differ in video rate by 1600 bits/sec, and requiring the H.323 terminal to follow whatever the H.320 rate is.

2)hide the differences in transfer rate on the H.320 side from the H.323 terminal. Thus, the gateway might perform a rate reduction or bit stuffing function.

Initiallly, I thought that having the gateway act as an H.244 CAU might be helpful, but at this moment, item (1) above seems simpler. Comments? }

H.22Z is designed so that, with an H.323 gateway, interpretability with H.320(1990), H.320(1993), and H.320(1996) terminals is possible. However, some features of H.22Z 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.

H.323 Protocol Stack



* I have received comments that X.224 does not belong here. What do you think??

Figure 1/H.22Z

{Comments on Figure 1: (I) 'Standardized Part and Non-standard Part' are not needed. (ii) 'X.224' is not needed here but it should be part of the IP and the IPX stack in Annex A. (iii) 'H.22z call control' is not needed because connection setup and teardown happens on the well-known unreliable channel and H.245 happens on the reliable logical channel 0. (iv)

1. Scope

This recommendation describes the means by which audio, video, data, and control are synchronized and passed between H.323 terminals on a non-guaranteed quality of service LAN, or between H.323 terminals and H.320, H.324, or H.310/H.321 terminals on N-ISDN, PSTN, or B-ISDN respectively via a LAN/WAN gateway. This gateway, terminal descriptions, and procedures are described in H.323 while H.22Z 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.22Z is intended to operate over a variety of different LANs, including IEEE 802.3, Token Ring, etc.. In this context, "LAN" includes the transport layer, such as TCP/IP/UDP, SPX/IPX, etc. Thus, H.22Z is defined as being above the transport layer. It is expected that a convergence sublayer would exist between H.22Z and the underlying real LAN. Many characteristics of this sub-layer are left to the manufacturer, but some elements require standardization for interoperability between H.323 terminals on the same LAN type. Specific profiles for particular LAN protocol suites are included in Annex A of this recommendation. Thus, the scope of H.22Z communication is between H.323 terminals and H.323 gateways on the same LAN, using the same convergence sub-layer. This LAN may be a single segment or ring, or it logically could be the entire Internet. 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, or on the Internet in general is beyond the scope of this recommendation. However, H.22Z 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.22Z are intended to extend H.320/H.221 conferences/connections onto the non-guaranteed QOS LAN environment. As such the primary conference model is one with size less than 1000 parties, strong admission control, and tight conference control. This is in contrast to various IETF (InterNet Engineering Task Force) protocols that focus on very large conferences with weak admission and conference control.

H.22Z makes use of the IETFs RTP/RTCP(Real Time Protocol/Real Time Control Protocol) for media steam packetization and synchronization for all underlying LANs. Please note that the usage of RTP/RTCP as specified in H.22Z is not tied in any way to the usage of TCP/IP/UDP. H.22Z assumes a call model where initial signaling on a non-RTP LAN port is used for call establishment and capability negotiation (see H.323 and H.245), followed by the establishment of one or more RTP connections. H.22Z contains details on the usage of RTP/RTCP~~, and also specifies any extensions to RTP/RTCP that are required for interworking with H.320~~.

Since H.22Z supports only single connection operation in the H.221 sense (i.e. any "channel" structure provided by the LAN is analogous to the "channels" of H.221, such as audio, video, LSD, etc. Thus, on the LAN side there are only 128 kbps calls, not 2*64 kbps calls); this rate matching function requires a little more thought. The primary rationale is to put complexity in the gateway rather than the terminal and to avoid extending onto the LAN features of H.320 that are tightly tied to ISDN unless this is absolutely necessary.

{Editors Note: There are two possible ways to deal with the rate matching problem:

4. ITU-T Recommendation H.221(1993)[1], *Frame Structure for a 64 to 1920 kbit/s channel in audiovisual teleservices.*

5. ITU-T Recommendation H.230(1993)[2], *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)[3], *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)[4], *Narrowband Visual Telephone Systems and Terminal Equipment.*

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

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

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

13. H.324.

14. H.321

15. H.322

16. H.310

17. RTP: A Transport Protocol for Real-Time Applications, IETF, March 21, 1995

18. RTP Profile for Audio and Video Conferences with Minimal Control, IETF, July 7, 1995

19. RTP Payload Format for H.261 Video Streams, IETF June 9, 1905

3. Definitions

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[1]Previously CCITT Recommendation
[2]Previously CCITT Recommendation
[3]Previously CCITT Recommendation
[4]Previously CCITT Recommendation

__'Combination of Q.931 and H.245' should be replaced with H.323 specific connection setup and teardown (AVC-8276 has the connection setup model and PDUs)__

Scope of H.22Z

The general approach of H.22Z is to provide a means of synchronizing packets that makes use of the underlying LAN facilities. H.22Z is not a "multiplex" and does not attempt to pack all media and control into a single stream, which is then packetized. The framing mechanisms of H.221 are not utilized for the following reasons:

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

2. References

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

{Editors Notes: References are included to RTP/RTCP and related documents to avoid duplication of text in H.22Z. Our current focus is on establishing a structure for the usage of RTP/RTCP. The issue of how to include the IETF documents is deferred for the time being. Options appear to be:

a)Include by reference - it is unclear that the ITU-T can include an IETF RFC as a normative document

~~***b)Include by reference but in a non-normative annex. This has the potential to lead to interoperability problems***~~

~~***c)Include the text of RTP in H.22Z, making RTP an ITU-T document***~~

~~***d)Create an H.RTP, H.RTCP, etc. as needed, and refer to these documents to avoid making H.22Z too large; note that the RTP related documents total 94 pages!.***~~

~~***e)Decide to not use RTP/RTCP after all.***~~

~~***Both (c) and (d) have the potential to lead to divergence between the ITU-T and IETF versions.}***~~

__Include RTP version 2 by reference. It is well thought out, light-weight, stable, and is already the standard in the computer industry. No proposal to this date (including our own work within Intel) has produced a better real time protocol. RTP will bridge the gap between the telephony and the computer industry.__

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

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

{Editor's Note: It is proposed that the GLI be removed. I have received some input that it is not useful. I am also tending toward the idea that it does not add any value that is not already in Annex A. Comments?

6.Synchronization Mechanism

6.1.General Approach

The general approach is to send call setup and call disconnect messages ~~(Q.931)~~ on a well known non-guaranteed channel and control(H.245) using an underlying guaranteed delivery means of transport . The impact of varying delay on these types of PDUs is not as strong as the impact on audio and video. These messages are contained in a control PDU structure described in section XXX below. They are used to first establish the connection between the gateway (and optionally the gatekeeper) and the terminal~~(Q.931~~), and then to exchange capabilities (H.245). In circumstances in which the gateway/gatekeeper are not required, this method is also used for point-to-point LAN call setup.
Once this control LAN port has been established, additional LAN ports for audio, video, and data may be established based on the outcome of the capability exchange. Also, the nature of the LAN side multi-media conference (centralized vs distributed/multicast) is negotiated on a per-conference~~media~~ basis. Note that control and data are centralized whereas video and audio can either be centralized or ~~may also potentially be~~ distributed/multicast.

Internet: Any system for the connection of LANs at the network layer. The operation of such systems is beyond the scope of H.22Z.

LAN: In the context of this document, a LAN(Local Area Network) is a mechanism for the switching of packet data over a limited area, including the physical, link, and transport layers. Thus, the "LAN" is beyond the scope of H.22Z for the most part, and H.22Z is intended for use with a variety of "LANs."

LAN Port: A destination on the LAN with a known address. May be multicast or unicast. Not to be confused with an MCU port. ***{Editor's note: Is there a better term?}***

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

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

4. Conventions

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

5. Abbreviations

BAS	Bit rate Allocation Signal
CIF	Common Intermediate Format
ECS	Encryption Control Signal
FAS	Frame Alignment Signal
FAW	Frame Alignment Word
GLI	General Lan Interface
GOB	Group of Blocks
H-MLP	High speed Multi-Layer Protocol
HSD	High Speed Data
IETF	Internet Engineering Task Force
IP	Add
LAN	Local Area Network
LD-CELP	Low Delay - Code Excited Linear Prediction
LSB	Least Significant Bit
LSD	Low Speed Data
MBE	Multi-Byte Extension
MCC	Multipoint Command Conference
MCN	Multipoint Command Negating
MCS	Multipoint Command Data

- ~~*Media capture timestamps used to restore media clocks and to associate audio and video. IN RTP.*~~
- ~~*Any media specific messages such as freeze picture release. IN RTP?*~~
- ~~*Specific rules for packetizing each type of media. IN RTP, T.123.*~~
- ~~*Methods for recovering from lost packets; the methods vary with media type. IN RTP; SOME ADDITIONAL METHODS ADDED HERE.*~~
- ~~*Per media sequence numbers used to detect lost packets. USE RTP SEQUENCE NUMBERS ON SEPARATE LAN PORTS.*~~

~~*I have included this list as a reminder of what was agreed, and will remove it once it is apparent that RTP is satisfactory to all.}*~~

In some cases signals carried by the BAS channel in H.221 are carried with the media stream in H.22Z. This is described below in section XXX, and also in H.323.

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 procedures of H.22Z are intended to support all allowed combinations. Thus, H.22Z provides for both point-to-point and multi-cast links per media, with the exception of T.120 data, for which only centralized operation is described; with distributed T.120 being for further study.

Audio and video are sent using RTP via a non-guaranteed delivery means to minimize delay. Error concealment or other recovery action must be applied to overcome lost packets; in general audio/video packets are not re-transmitted since this would result in excessive delay in the LAN environment. The assumption is that bit errors are detected in the lower layers, and errored packets are not sent up to H.22Z. Note that audio/video, ~~and~~ call setup, and ~~/~~control are never sent on the same LAN connection, and do not share a common PDU structure. Audio and video are sent to separate LAN ports using separate instances of RTP to allow for media-specific frame sequence numbers.

T.120 capabilities are negotiated via the H.245 control LAN port, and upon receipt of appropriate messages, T.125 sessions are established on LAN using the LAN stacks of T.123 as appropriate. There is no synchronization of T.120 data with the audio/video in RTP, or with the call setup/disconnect on the well-know port, and ~~Q.931~~/H.245 control LAN port. T.120 is conveyed over the LAN between endpoints, or between the endpoint and the gateway on a fourth LAN connection (T.120 may used up to 4 LAN ports, one for each MCS priority). Thus, the typical point-to-point, or point-to-gateway link can be expected to have at least six~~three~~ (and up to ten~~seven~~ if MLP is used) LAN ports.

1. Audio/RTP and Audio/RTCP

2. Video/RTP and Video/RTCP

3. Control on logical channel 0~~(Q.931/H.245)~~
4. Call setup/disconnect on the well-known port

4. Data (T.120)[Up to 4 LAN ports]

~~{Editor's Note: I have received more than one input that this lmitation is not needed since the higher layers have little control over LAN packet size in any case. It there is not disagreement, this rule will be removed}~~

~~{Editor's Notes: At the Haninge meeting, it was agreed to split H.22Z into two levels, and a list of needed functions was provided. I have included this list below, along with some comments on how they might (or might not) be met by RTP.~~

~~The functions of H.22Z are split into two levels. The lower level contains the following functions:~~

- ***~~Timestamp used to measure quality of service. These are numbered across all media types. INPUT FROM IETF SUGGESTS THAT WE CAN USE THE JITTER FEATURE OF RTP FOR THIS PURPOSE.~~***

- ***~~PDU structure that allows the association of audio, video, data, and control as being from a given endpoint. RTP ALLOWS ASSOCIATION VIA SOURCE IDENTIFIER; ASSOCIATION IS ALSO KNOWN SINCE ALL LAN PORTS ARE ESTABLISHED BASED ON INITIAL CONTROL LAN PORT SIGNALING. DATA IS NOT COORDINATED IN TIME, BUT IS ASSOCIATED AT THE CONTROL LEVEL.~~***

- ***~~PDU structure that allows the determination of media type. DETERMINED IN PART FROM LAN PORT USED~~***

~~The higher layer includes the following functions:~~

~~***conference information. Also, there is no function in H.243 that requires that the endpoint determine who the loudest talker is, or be aware of the current number of endpoints in the audio mix. The use of this field for distrubuted style conferences on the LAN requires further consideration.***~~

Payload Type(PT): Only ITU-T payload types such as (0)[PCMU], (8)[PCMA}, (9)[G722], and (15)[G728] shall be used. ***{Editor's Note: It is our hope that codepoints will be allocated for G.723 and G.dsvd so that we can make use of them. However, if this is not that case, we will rely on H.245 signaling and ignore the RTP payload type.}***

~~***{New Editors Note: It appears that the SSRC serves the valuable function of allowing different program streams from one source to be distinquished. It isespecially important if a terminal is listening to more than one conference, where the H.243 terminal number will no longer suffice. Thus there seems no reason to modify the IETF rules for its useage. However,***~~ ***H.323 terminals will still get and sent their terminal numbers via the H.245 control link, and should regard the SSRC as more of a program id than a terminal id***
~~***{Editor's Note: RTP-P requires a default 20 msec packet, with a requirement that the terminal be able to receive packets from 0 to 200 msec of audio. Is this acceptable to all?***~~ ***{Comment: 0 to 200 ms is an acceptable range.}*** ~~***Should we tighten up on this in some fashion? I have received input that the 0-200 msec requirement is not compatible with much currently used PC hardware. Thus, this appears to be one area WHERE H.22Z WILL CLEARLY DEVIATE FROM RTP. The current suggestion is that the endpoint should signal via caps the packet size they can handle using H.245. We will need to define a required size range - suggestions? 50-150 msec mandatory???}***~~

{Comment: This deleted paragraph should be part of the audio algorithm}~~If the underlying LAN indicates a bad packet, or if a interruption in sequence numbers is observed, the receiver shall repeat the most recent octet with a time constant such that the value of the repeated octet decays to silence in 10 msec. The choice of the time constant is left to the manufacturer.~~

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. ***{Editor's note: Is there any need for the H.323 terminal to know that PCM 48/56 is being used?}***

{Editor's note: We need to confirm that RTP can handle 48, 56, and 64 Kbps G.722 as well; there appears as for G.711, to be only one code point. The same sort of issues arise}
When using RTCP, both RR and SR packets should be sent periodically as described in RTP. Only the CNAME SDES PDU should be used, and instead of the canonical name mentioned in RTP, the H.243 terminal identifier shall be sent. ***{Editors note: We don't need CNAME since our SSRC is assured to be unique, but RTP says that CNAME is required in RTCP. Any other suggestions???Comment: The audio and video packets from the same source can have different SSRC identifiers. The CNAME is used to bind the audio and video so that the receivers can do, for example, lip synchronization.}***

6.2.2. Video PDUs

6.2. Use of RTP/RTCP

A separate LAN port will be established for audio and video. On each LAN port a separate instance of RTP will be used, one for audio and one for video. Additional LAN ports are needed for RTCP, one for each RTP LAN port. Using H.245 signaling, additional audio and video channels may be established if the terminal supports this capability.

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/RTP-P.

It should be kept in mind that the gateway may be acting as an RTP mixer, 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

The timestamp field of RTP for both audio and video packets should be accurate to within 1/1000~~1/64,000~~.
{Editor's Notes: I have received input that the RTP accuracy is too tight for many PC implemenatations. Comments on the 1/64,000 value??? As I understand it, this goal is easily met in UNIX systems, but difficult on PCs. Hence the use of "should."}

The H.323 LAN terminal, when engaged in a gateway mediated conference, whether point-to-point or multi-point, shall restrict the total bit rate averaged over a 5 second period (in other words, keep the bursts to a minimum!), to that signaled in the H.245 exchange. Thus, even though there may be a very large "limit" on the bit rate that can be transmitted on the LAN, the sender shall ensure that the sum of audio, video, and data on the LAN side does not exceed that sum of audio, video, and data on the WAN side as signaled by the gateway. The exceptions are

- 46/56 kbit/sec G.711, which is sent at the 64 kbit/sec rate on the LAN, and truncated or expanded in the H.323 gateway.
- Possibly G.722
- When the gateway is using a rate reducer; in this case the LAN side H.323 terminal shall matched the signaled rate, which will probably be less than the rate being sent over the WAN.

6.2.1. Audio

Version (V): RTP-P does not mention (apparently) which version number is to be used; perhaps it is intended to be implied. In any case, we should use (2) ***{Editor's Note: This will be removed as soon as we confirm that the change has been made in RTP. An issue with RTP may also exist in that it would be desirable to have a larger set of version numbers so that we can refer to a fixed point in time more easily. Comments, especially from IETFers?}.***

CSRC Count(CC): This shall be set to zero for the H.323 endpoint and for the gateway when the H.323 gateway is performing audio mixing. ~~***{Editor's note: What should the gateway set it to?? This seems problematic, as only 16 contributing sources are allowed, which is a lot less than the maximum H.231 conference size. Also, this seems like an awkward method for indicating who the loudest talker if is the gateway is acting as a mixer. One possibility is to require that the Gateway set CSRC to zero as well, and rely on H.245/H.243 to indicate***~~

Octet Number

Control Type (7 bits)	E	1
Message Body		2-N

Control Type: This 7 bit field has the values:

0	Reserved.
1	Q.931 Message
2	H.245 Message
3	H.22Z message
4	Reserved
5	Reserved for Q.2931 Messages
6-127	Reserved

E bit: When set to 1, indicates additional octets after the normal header. For this version of H.22Z, the E bit is zero.

{Editor's note: We have agreed that the use of a gatekeeper on a call is optional. However, it may be that support of terminal/gatekeeper messages should be required for all terminals. Consider the issue.}{Comment: Support of gatekeeper messages should be required for all terminals, MCs, and gateways. The gatekeeper is optional.}

7.2. Terminal and Gateway Registration PDUs

{Comment: AVC-827 introduces some new PDUs plus adds more information to the PDUs in Section 7.2.1 through 7.7.3 ~~6.2~~. We will update these sections soon with our comments.}

7.2.1. Registration Request - RRQ

Note: Need to add way of signaling non-standard features during registration. ***{Editor's Note: Shouldn't non-standard features be signaled call-by-call?? However, I strongly agree with the direction proposed here on this point}***

```
RegistrationRequest          ::=SEQUENCE --(RRQ)
{
        yearOfSpec           OCTET STRING (SIZE(4))
        requestSeqNum        INTEGER (1..65535),
        bindRequest          BOOLEAN,
        terminalIdentifier   OCTET STRING (SIZE(128)),
        ControlAddress       NetworkAddress,
        terminalType         NodeType,
        terminalExtNum       E.164Address
```

Version (V): RTP-P does not mention (apparently) which version number is to be used; perhaps it is intended to be implied. In any case, we should use (2). ***{ See earlier editor's note.}***

CSRC Count(CC): This shall be set to zero for the H.323 endpoint and the gateway.
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. Only the CNAME SDES PDU shall be used, 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 encoded using ASCII? T.61?. ***{Editors note: Since we are now potentially receiving from multiple sources in more than one conference, this is once again important. The H.243 terminal id, although unique in a conference is not unique across conferences, so we may need to use the RTP type domain name, or some other endpint unique name, such as a room number, etc. Comment: Two unrelated conferences from the same endpoint should have different CNAMES. Two related conferences from the same endpoint should have the same CNAME so you can use it to relate the same participant in each conference. }***

{Editor's note: The issue has been raised that MB packetization is not always desirable. Further views on this topic are solicited; noting that MB is what RTP calls for, I would suggest this as a requirement, and allow other options, but I believe that those concerned want to avoid having MB packetization be required. Comments??? Comment: MB provides flexibility in packetization, and resiliency. It is superior to GOB in the LAN environment.}

6.2.3.Data PDUs

There are no special data PDUs; T.120 is used on the LAN as per T.123. Centralized vs distributed data conferencing on the LAN is described in H.323, and is negotiated via H.245.

7. Control and Call Setup PDU Definitions

This section concerns the definition of PDUs for call setup, call control, and communications between the gateway, the gatekeeper, the VMC(Virtual MCU Controller), and the H.323 terminal. Procedures for the use of these PDUs are in H.323. { The H.323 document uses MC (Multipoint Controller) instead of VMC. This should be resolved.}

7.1. Control PDU Structure

{Comment: If call setup/disconnect PDUs are sent on the well-known unreliable channel and the H.245 control PDUs are sent on the reliable logical channel 0, the following header is not needed. This section (7.1) can be deleted.

The control PDU structure is as follows:

IP6Address SEQUENCE,
{
transport OCTET STRING (SIZE(16)),
port INTEGER(0..4294967295)
},
NetBios OCTET STRING (SIZE(16)),
}

NodeType ENUMERATED
{
Gatekeeper (1),
Gateway (2),
MC (4),
H323Terminal (8),
Undefined Node (268435456)
}

7.2.2. Registration Confirmation - RCF

~~Field~~	~~Field Size~~
~~Capability Definition~~	~~TBD~~
~~Source Identifier~~	~~CHAR(25)~~
~~Source Identifier Number~~	~~WORD~~
~~Year of Specification~~	~~CHAR(4)~~

RegistrationConfirmation ::=SEQUENCE --**(RCF)**
{
yearOfSpec OCTET STRING (SIZE(4))
requestSeqNum INTEGER (1..65535),
gatekeeperIdentifier OCTET STRING (SIZE(64)),
gatekeeperAddress NetworkAddress (SIZE(3)),
extensionCount INTEGER (0..65535)
}

7.2.3. Registration Rejection - RRJ

~~Field~~	~~Field Size~~
~~Rejection Reason~~	~~WORD~~
~~Year of Specification~~	~~CHAR(4)~~

Registration Reject ::=SEQUENCE --**(RRJ)**
{
yearOfSpec OCTET STRING (SIZE(4))
requestSeqNum INTEGER (1..65535),
rejectReason RejectReason,
}

Where:

- Rejection Reason is defined as:

RejectReason ENUMERATED
{
Not Bound Registration (1),
Duplicate Registration Request (2),

```
        extensionCount          INTEGER (0..65535)
}
```

~~Field~~ ~~Field Size~~

~~Field~~	~~Field Size~~
~~Source Network Address~~	~~BYTE(16)~~
~~Source Identifier~~	~~CHAR(25)~~
~~Source Identifier Number~~	~~WORD~~
~~Year of Specification~~	~~CHAR(4) e.g. "1995"~~

~~Where~~ ~~***{Editor's Note: this will be moved to Annex A since it is LAN specific}:***~~

- ~~Network address is defined for each network as follows:~~

~~For IP:~~

~~Field~~	~~Field Size~~
~~Network and Host Address~~	~~BYTE(4)~~
~~Socket Number~~	~~BYTE(2)~~
~~Subnet Mask~~	~~BYTE(4)~~
~~Reserved~~	~~BYTE(6)~~

~~For IPX:~~

~~Field~~	~~Field Size~~
~~Network Address~~	~~BYTE(4)~~
~~Node Address~~	~~BYTE(6)~~
~~Socket Number~~	~~BYTE(2)~~
~~Reserved~~	~~BYTE(4)~~

~~For Other Networks:~~

~~For Further Study.~~

~~Note: Subnet Mask is required for IP networks to locate terminals within the network topology.~~

- ~~Source Identifier is an optional text field for terminal or Gateway identification.~~

- ~~Source Identifier Number is an optional field to allow a terminal to associate a number with its address for identification purposes.~~

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

7.3. Gatekeeper to Gatekeeper Configuration Messages

Note that how a terminal obtains a directory address is beyond the scope of H.22Z/H.323.

~~7.3.1. Sub-Domain List Request~~

~~This message can be used only for gatekeeper to gatekeeper operation.~~

~~Field~~	~~Field Size~~
~~Terminal Type~~	~~BYTE~~
~~Source Network Address~~	~~BYTE(16)~~

- ~~Terminal Type is defined as:~~

~~Terminal Type~~	~~Code~~
~~H.323 Terminal~~	~~0~~
~~Gateway~~	~~1~~
~~Gatekeeper~~	~~2~~

~~7.3.2. Sub-domain Registered List~~

~~Field~~	~~Field Size~~
~~Number of Registered Terminals~~	~~WORD (16 bits)~~
~~Registered Terminal Record~~	~~Terminal Record(Number of Registered Terminals)~~

~~Where:~~

- ~~Terminal Records are defined as:~~

~~Terminal Record~~	~~Code~~
~~Terminal Network Address~~	~~NETWORK ADDRESS~~
~~Terminal Type~~	~~BYTE~~

7.3.1 Node List Request

This message may be sent from any node to a gatekeeper.

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

7.3.2 Node List Response

```
NodeResponseList             ::=SEQUENCE --(NRL)
{
        yearOfSpec           OCTET STRING (SIZE(4))
        requestSeqNum        INTEGER (1..65535),
        responseStatus       ResponseStatus
```

Invalid Ext Num (3),
Duplicate Bind Request (4),
Invalid Revision (5),
Invalid Network Address (6),
Undefined Reason (65535)
}

~~Rejection Reason~~	~~Code~~
~~Invalid Network Address~~	~~1~~
~~Duplicate Network Address~~	~~2~~
~~Others~~	~~?~~

7.2.4 Guardian Query Request - GQQ

Guardian Query Request ::=SEQUENCE --**(GQQ)**
{
yearOfSpec OCTET STRING (SIZE(4))
requestSeqNum INTEGER (1..65535),
replyAddress NetworkAddress,
extensionCount INTEGER (0..65535)
}

7.2.5 Guardian Query Response - GQRS

-**Guardian Query Response** ::=SEQUENCE --**(GQRS)**
{
yearOfSpec OCTET STRING (SIZE(4))
requestSeqNum INTEGER (1..65535),
gatekeeperIdentifier OCTET STRING (SIZE(64)),
gatekeeperAddress NetworkAddress,
extensionCount INTEGER (0..65535)
}

7.2.6 Guardian Query Reject - GQRJ

Guardian Query Reject ::=SEQUENCE --**(GQRJ)**
{
yearOfSpec OCTET STRING (SIZE(4))
requestSeqNum INTEGER (1..65535),
rejectReason GuardianQueryRejectReason,
}
Where:
- Rejection Reason is defined as:

GuardianQueryRejectReason ENUMERATED
{
No Gatekeeper (1),
Invalid Revision (5),
Undefined Reason (65535)
}

```
        destinationWanInfo    WanInfo,
        conferenceID          INTEGER(0..4294967295),
        callType              CallType,
        callMedia             CallMedia,
        bandWidth             INTEGER (1..4294967295),
        connectionID          INTEGER (1..4294967295),
        extensionCount        INTEGER (0..65535)
}
```

Where:

```
        WanInfo                         ::=SEQUENCE
        {
                E.164Numbers  SET SIZE(0..6) OF OCTET STRING (SIZE(16)),
                channelRate   INTEGER (1..4294967295),
                ????MORE INFO
        }

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

        CallMedia                       ENUMERATED
        {
                Data          (1),          -- note that these may be logically OR'd
                Audio         (2),
                Video         (4)
        }
```

- ~~Call Type is defined as:~~

~~Call Type~~	~~Code~~
~~Point to Point Call~~	~~1~~
~~Multicast call~~	~~2~~

- Bandwidth is defined as the maximum amount of bandwidth required for the call in Kbits/second (total bandwidth in both directions). This could be used by the Gatekeeper in its connection approval determination processing. ~~Note that Bandwidth is needed here, but is repated in the Q.931 messages.~~

{Editor's Note: Additional material will be added here on the usage of Q.931 messages for call progress, connection, etc. The messages will not be needed for a LAN to LAN connection, but will be needed if the gateway is involved in the call.}

```
        gatekeeperIdentifier    OCTET STRING (SIZE(64)).
        nodeList                SEQUENCE SIZE(1..256) OF NodeEntry.
        extensionCount          INTEGER (0..65535)
}
```

Where:

```
ResponseStatus              ENUMERATED
{
        Success             (0).
        Not Supported       (1).
        Unknown Node Type   (2).
        Invalid Revision    (5).
        Undefined Status    (65535)
}

NodeEntry           ::=SEQUENCE
{
        nodeIdentifier      OCTET STRING (SIZE(64)).
        nodeSeqNum          INTEGER (1..65535)).
        nodeAge             INTEGER (1.. 4294967295).
        nodeAddress         NetworkAddress (SIZE(3)).  -- one per transport
        nodeType            NodeType
}
```

~~7.3.3. Gatekeeper Termination Indication~~

~~No Parameters~~

7.4. Connection Request Messages

7.4.1. Connection Request - CRQ

~~Note: Need to add means of addressing multiple nodes.~~

This message is sent to a well known port on that does not use guaranteed delivery means of operation.

~~Field~~	~~Field Size~~
~~Dynamic Control and Data Network Address (guaranteed link)~~	~~Network Address~~
~~Call Type~~	~~BYTE~~
~~Destination Network Address~~	~~Network Address~~
~~Bandwidth~~	~~WORD~~

```
Connection Request          ::=SEQUENCE --(CRQ)
{
        yearOfSpec              OCTET STRING (SIZE(4)).
        requestSeqNum           INTEGER (1..65535).
        originatingID           OCTET STRING (SIZE(128)).
        originatingAddress      NetworkAddress.
        destinationAddress      NetworkAddress.
        originatingGatekeeper   NetworkAddress.
        destinationGatekeeper   NetworkAddress.
```

}

7.4.4 Connection In Progress - CIP

```
Connection In Progress        ::=SEQUENCE --(CIP)
{
    yearOfSpec                OCTET STRING (SIZE(4))
    requestSeqNum             INTEGER (1..65535),
    connectionID              INTEGER (1.. 4294967295),
    channelNum                INTEGER (0..65535),       -- ID of WAN channel #
    connection Status         ConnectionStatus
    extensionCount            INTEGER (0..65535)
}

    ConnectionStatus              ENUMERATED
    {
        Connected                 (0),
        No Connection             (1),
        Idle                      (2),
        Disconnecting             (3),
        Dialing                   (4),
        Connecting                (5),
        Ringing                   (6),
        Redirecting               (7),
        Undefined Status          (65535)
}
```

7.5. Disconnect Request PDU

The purpose of these messages is to inform the gatekeeper that a call is dropping; they are sent only in the optional gatekeeper mode of operation. ~~A mandatory Q.931 message will also be sent to the gateway.~~

7.5.1. Disconnect Request - DRQ

```
~~Field~~                     ~~Field Size~~
~~Disconnect Reason~~         ~~BYTE~~
DisconnectRequest             ::=SEQUENCE --(DRQ)
{
    yearOfSpec                OCTET STRING (SIZE(4))
    requestSeqNum                  INTEGER (1..65535),
    connectionID              INTEGER (1.. 4294967295),
    terminalIdentifier             OCTET STRING (SIZE(128)),
    gatekeeperIdentifier      OCTET STRING (SIZE(64)),
    disconnectReason          DisconnectReason,
    extProtocolType                ProtocolType,
    extReason                 INTEGER(0..65535)
    reasonString              OCTET STRING (SIZE(80)),
    delayTime                 INTEGER (0..65535),   - number of seconds
    extensionCount            INTEGER (0..65535)
}
```

7.4.2. Connection Confirmationt - CCF

```
Connection Confirmation              ::=SEQUENCE --(CCF)
{
        yearOfSpec                  OCTET STRING (SIZE(4))
        requestSeqNum               INTEGER (1..65535),
        originatingGatekeeper       NetworkAddress,
        destinationAddress          NetworkAddress,
        destinationGatekeeper       NetworkAddress,
        conferenceID                INTEGER(0..4294967295),
        connectionID                INTEGER (1.. 4294967295),
        extensionCount              INTEGER (0..65535)
}
```

7.4.3 Connection Reject - CRJ

```
Connection Rejection          ::=SEQUENCE --(CRJ)
{
        yearOfSpec                  OCTET STRING (SIZE(4))
        requestSeqNum                       INTEGER (0..65535),
        rejectReason                RejectReason,
        conferenceID                INTEGER(0..4294967295),
        connectionID                INTEGER (1.. 4294967295),
        extProtocolType                     ProtocolType,
        extReason                   INTEGER(0..65535)
        bandWidth                   INTEGER (1..4294967295)  -- measured in 1k bit increments
        extensionCount              INTEGER (0..65535)
}

Where:
        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
                Destination NoAnswer                (14),
                Undefined Reason                    (65535)
        }
        ProtocolType             ENUMERATED
        {
                Q.931                    (1),
                Undefined Protocol       (65535)
```

```
    originatingID             OCTET STRING (SIZE(128)),
    originatingAddress        NetworkAddress,
    destinationAddress        NetworkAddress,
    originatingGatekeeper     NetworkAddress,
    destinationGatekeeper     NetworkAddress,
    destinationWanInfo        WanInfo,
    callType                  CallType,
    callMedia                 CallMedia,
    bandWidth                 INTEGER (1..4294967295),
    bytesSent                 INTEGER (1..4294967295),
    bytesRcvd                 INTEGER (1..4294967295),
    extensionCount            INTEGER (0..65535)
}
```

- ~~For H.323 Terminals:~~

~~Field~~	~~Field Size~~	~~Field Description~~
~~Call State~~	~~WORD~~	~~State of the call.~~
~~Packets Sent~~	~~DWORD~~	~~Number of total packets sent~~
~~Packets Received~~	~~DWORD~~	~~Number of total packets received~~
~~Packets Missed~~	~~DWORD~~	~~Number of total packets missed~~
~~Packets Received Out of Order~~	~~DWORD~~	~~Number of total packets received out of order~~

- ~~For Gateways:~~

~~Field~~	~~Field Size~~	~~Field Description~~
~~Call State~~	~~WORD~~	~~State of the call.~~
~~Packets Sent~~	~~DWORD~~	~~Number of total packets sent~~
~~Packets Received~~	~~DWORD~~	~~Number of total packets received~~
~~Packets Missed~~	~~DWORD~~	~~Number of total packets missed~~
~~Packets Received Out of Order~~	~~DWORD~~	~~Number of total packets received out of order~~
~~Total Channel Rate~~	~~WORD~~	~~Audio Video and Data~~
~~Audio Receive Mode~~	~~WORD~~	~~H.320 audio receive mode~~
~~Audio Transmit Mode~~	~~WORD~~	~~H.320 audio transmit mode~~
~~Video Receive Mode~~	~~WORD~~	~~H.320 video receive mode~~
~~Video Transmit Mode~~	~~WORD~~	~~H.320 video transmit mode~~
~~Data Receive Mode~~	~~WORD~~	~~Data receive mode~~
~~Data Transmit Mode~~	~~WORD~~	~~Data transmit mode~~
~~Data Receive Rate~~	~~WORD~~	
~~Data Transmit Rate~~	~~WORD~~	
~~Multipoint Connect~~	~~WORD~~	

7.7. QOS Related PDUs

7.7.1. Bandwidth Request - BRQ

Where:

- Disconnect Reason is defined as:

DisconnectReason ENUMERATED
{
Hang Up (1),
Remote Hang Up (2),
Remote Abort (3),
Transfer (4),
Gatekeeper (5)
Undefined Reason (65535)
}

~~Disconnect Reason~~	Code
~~Hang Up~~	1
~~Remote Site Aborted Call~~	2
~~LAN Bandwidth Limit Exceeded~~	3

Note: The Disconnect reason between peers is usually "Hang Up", but in the event of a power failure or problem at one end of a link, the reason code when the disconnect is sent to the Gatekeeper will contain "Remote Abort ~~Remote Site Aborted Call~~"

~~7.5.2. Disconnect Confirmation - DCF~~

~~No Parameters.~~

7.6. Status Request Messages

7.6.1. Status Request - SRQ

~~No Parameters.~~

Status Request ::=SEQUENCE --**(SRQ)**
{
yearOfSpec OCTET STRING (SIZE(4))
requestSeqNum INTEGER (1..65535),
connectionID INTEGER (0..4294967295),
extensionCount INTEGER (0..65535)
}

7.6.2. Status Response - SRR

~~Field~~	~~Field Size~~
~~Terminal Type~~	~~BYTE~~
~~Terminal Status~~	~~Status based on Terminal Type~~

Status Report Response ::=SEQUENCE --**(SRR)**
{
yearOfSpec OCTET STRING (SIZE(4)),
requestSeqNum INTEGER (1..65535),
nodeType NodeType,
conferenceID INTEGER(0..4294967295),
connectionID INTEGER (1..4294967295),
callState ConnectionStatus,

Logically, there are two types of congestion related delay that might be measured:

- Short term increases in delay that will result in a perceptible but not annoying slowing of the frame rate. These bursts are less than 1 {Editor's note: what is the right figure; this is too short} second in duration.

- A general rise in delay due to LAN congestion over time such that a feedback based mechanism is useful. This rise in congestion is measured over 1 minute intervals{Editor's note: is this the right interval?}Essentially, short term bursts are approached by error concealment, and a longer term congestion is approached by reducing the multi-media load. The assumption is made that all LAN multimedia terminals are H.323 terminals, and all will attempt to reduce LAN usage as congestion rises rather than "steal" bandwidth from each other.

8.2. Use of RTCP in Measuring QOS

8.2.1. Sender Reports

{Editor's Note: It is recognized that this section needs more work}The sender report servers 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.22Z. Manufacturers may make use of the sender reports in other ways at their discretion.

The relevant field for ~~RTP~~ stream synchronization is the RTP timestamp and the NTP timestamp in the sender report of RTCP. The NTP timestamp ~~, which~~ gives "wall clock" time and corresponds to the RTP timestamp which has ~~with~~ the same units and random offset as the RTP capture timestamp in the media packets. The CNAME in RTCP binds the differrent SSRC identifiers from the same sender and can be used for synchronization. ~~Since both audio and video come from the same source, any inaccuracy or drift in the terminal "wall clock" is not important.~~ ***{Add more details on how this is used}***

~~When the difference between the audio and the video RTP timestamp exceeds 100 msec, indicating that audio and video are at least 100 msec out of synchronization, the receiver shall take recovery action by sending a ??? request to the sender. This message causes the sender to restart audio and video transmission in a synchronized fashion.~~ ~~***{Editor's note: At Haninge, it was agreed that a procedure of this sort was needed. Does this still seem to be the case? Is this the right value? The right procedure?}***~~

8.2.2. Receiver Reports

Three parts of the Receiver Reports are used in H.22Z to measure QOS:

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

```
BandwidthRequest              ::=SEQUENCE --(BRQ)
{
    yearOfSpec                OCTET STRING (SIZE(4))
    requestSeqNum             INTEGER (1..65535),
    terminalIdentifier        OCTET STRING (SIZE(128)),
    connectionID              INTEGER (1.. 4294967295),
    callType                  CallType,
    callMedia                 CallMedia,
    replyAddress              NetworkAddress,
    gatekeeperIdentifier      OCTET STRING (SIZE(64)),
    bandWidth                 INTEGER (1..4294967295)  -- measured in 1k bit increments
    extensionCount            INTEGER (0..65535)
}
```

7.7.2. Bandwidth Confirmation - BCF

```
Bandwidth Confirmation        ::=SEQUENCE --(BCF)
{
    yearOfSpec                OCTET STRING (SIZE(4))
    requestSeqNum             INTEGER (1..65535),
    bandWidth                 INTEGER (1..4294967295)  -- measured in 1k bit increments
    extensionCount            INTEGER (0..65535)
}
```

7.7.3. Bandwidth Reject - BRJ

```
Bandwidth Reject              ::=SEQUENCE --(BRJ)
{
    yearOfSpec                OCTET STRING (SIZE(4))
    requestSeqNum             INTEGER (1..65535),
    rejectReason              BandRejectReason,
}
```

Where:

- Reject Reason is defined as:

```
    BandRejectReason                ENUMERATED
    {
        Not Bound                   (1),
        Invalid ConnectionID        (2),
        Invalid Permission          (3),
        Request Denied              (4),
        Invalid Revision            (5),
        Undefined Reason            (65535)
    }
```

8. Mechanisms for maintaining QOS

8.1. General Approach and Assumptions

The Sender and Receiver Reports of RTCP shall be the means by which QOS will be assessed.

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

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.
- Turning off media of lessor importance (e.g. turning off video to allow a large amount of T.120 traffic).
- 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 LAN port.

{Editor's Note: Should these procedures be detailed here, or in H.323? Clearly H.22Z must at least supply the adaptive busy related messages, but the media control messages already exist in H.245}

4. Interarrival jitter

Items 2 and 3 are used to compute the number of packets lost since the previous receiver report. This can be taken as a long term measure of LAN congestion. See RTP section 6.3.4 for a sample computation. If this loss rate exceeds 1% ***{Editor's note: What is a good value?}*** the H.22Z terminal should reduce the media rates on the LAN side according to the procedures in section XXX below. If item 1 exceeds 1%, it may also be desirable to take corrective action. If the interval between receiver reports exceeds 5 minutes, H.323 terminals shall use item 1 as an indicator of serious congestion requiring media rate reduction on the LAN side.

Item 4 should be used as an indication of impending congestion. If interarrival 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. ~~GLI~~ QOS Metrics

{Editor's Note: In this section I was working towards some specification of the general relationship between total LAN bandwidth, LAN fundamental bit rate, desired media delay parameters, and suggested packet size for audio and video. The obvious notion is that packet size can increase as LAN bit rate increases. However, it may be that this sort of thing should be left to the manufacturer, as it will vary from LAN to LAN. Still, it seems that we need to have some kind of limit on packet size for audio and video. RTP suggests 20 msec audio packets, with 0-200 msec range support required. What are your views?}
{Comment: 20 ms audio packets with 0 to 200 ms range is acceptable.}

The following metrics are used to characterize the general LAN:

Maximum Bit Rate(MBR): The bit rate at which packets are sent on the LAN. For example, 10 Mbps for IEEE 802.3 is a common rate.

Instantaneous Packet Delay(IPD): The delay on a packet due to any arbitration or collisions that may take place. This delay varies from moment to moment, and often increases with load on the LAN.

Packet jitter(PJ): The variation in IPD from packet to packet that must be compensated for by buffering on the receive side. Generally, the higher the jitter, the more delay across the LAN.

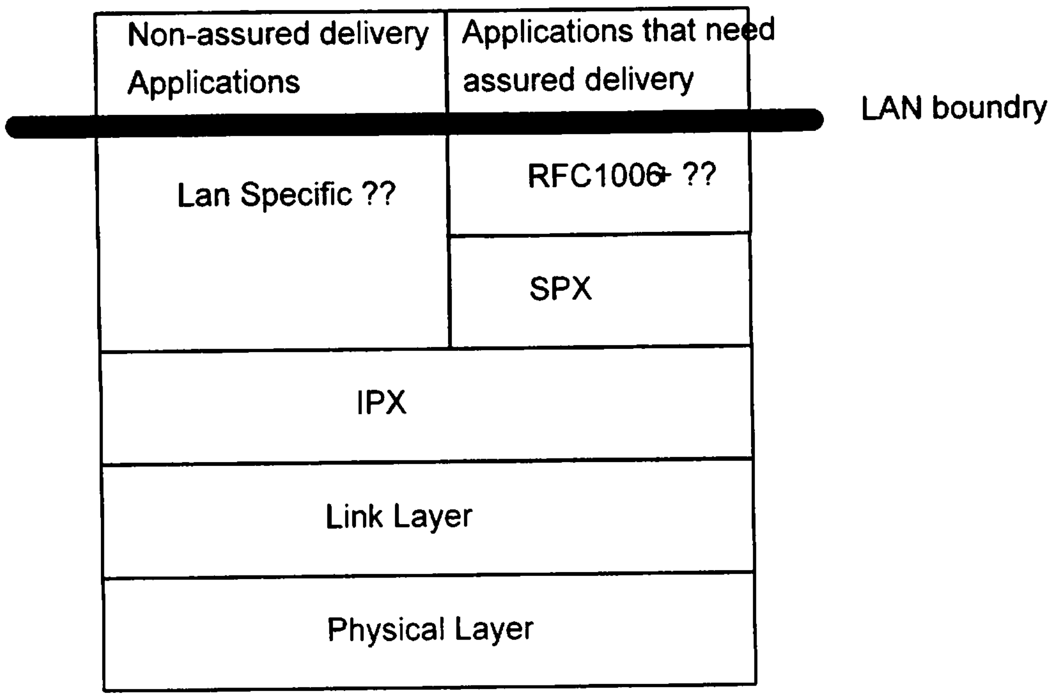
Media Bandwidth: On the LAN, the MBW is a product of the Packet size time the Packets/sec. For example, 64 kbit/sec PCM might be sent as 80 packets/sec of 100 bytes each.

Minimum Media Delay: On the LAN, the minimum delay that will be experienced is that needed to fill a packet plus the time needed for the packet to transit the LAN. Thus, to fill a 100 octet packet with PCM requires 12.5 msec. On a 10 Mbps LAN, sending the 800 bit packet requires about 80 usec. Whether we must double the packing delay to get overall delay depends on whether a checksum is computed on each packet

Actual Packet Delay: This is the sum of MMD(Minimum Media Delay) plus IPD(Instantaneous Packet Delay) plus Packet jitter (PJ). Note that jitter may raise or lower the total delay.

9.2. SPX/IPX

{Comments: RFC 1006 is the same as X.224 Class 0. It is needed to convert stream oriented SPX~~TCP~~ to packet oriented.}



9. Annex A

This annex provides additional details concerning the operation of H.22Z on various actual LAN protocol stacks. This annex is non-normative. ***{Editor's Note: We may split the annex into normative and non-normative parts}***

9.1. TCP/IP/UDP

{Comments: RFC 1006 is the same as X.224 Class 0. It is needed to convert stream oriented TCP to packet oriented.}