Experts Group for Video Coding and Systems in ATM and Other Network Environments

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STUDY GROUP 15 CONTRIBUTION

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1.0 Introduction

This document contains submissions on the following topics:

1.0 Introduction
2.0 Comments on the 12/28/95 H.323 Draft
- Specific Comments, Registration, MC Issues, Maximum Jitter Delay, Conference Models, Ad Hoc Conferencing, Call Forward and Transfer, Audio Processing, Asymmetric Operation, Bandwidth Changes

3.0 Comments on 12/28/95 H.225.0 Draft
- Specific Comments, PDU Review, Optional Messages

4.0 H.245 Additions
- H.225 Capability, Open Logical Channel ACK and Reject

2.0 Comments on 12/28/95 H.323 Draft

2.1 Specific Comments

- Section 1.1.1 - delete "speech and voice" in the list of things a terminal may provide as "speech only" should cover "speech and voice".
- Section 1.1.4 - The MP maybe the top MCS provider in a distributed multipoint conference. Change "centralized or hybrid multipoint conference" to "multipoint conference"
- Section 1.1.5 - Given that the simplest case of an MCU is an MC without an MP, isn't a MC a MCU and further, isn't the MC then callable?
- Section 1.3.2 - (On Editor's Note) How could this conference happen without an MC?
- Section 1.3.2.4 - Should we state "Mixed or Switched" video stream instead of just mixed? I would presume that switching would be the more common implementation.
- Section 1.3.2.5 Mixed Conference - Further Study? Given our current signaling procedures how is the conference created?
- Section 3 "Callable" - The simplest MCU is an MC which does not have audio MP. Callable should not list "plus the audio media stream" as a requirement.
- Section 5.2.4 Paragraph 5 - Asymmetric operation should not be mandatory (A QCIF only terminal is legal) so "shall" should be "may".
- Section 5.2.4.1 - We should note that the simultaneous capability signaled by any given terminal (a single terminal) in a conference where video is multicast should not limit the number of multicast video streams in that conference.
- Section 5.2.5.1 - We should note that the simultaneous capability signaled by any given terminal (a single terminal) in a conference where audio is multicast should not limit the number of transmitted audio streams in that conference.
- Section 5.2.6 - Replace paragraph 2 with the following: (this was proposed in PTEL14 and may have already been included)

  Intermediate processing points such as MCUs or interworking adapters may alter the audio and video time tag packet information, and shall transmit appropriately modified audio and video time tags and sequence numbers, reflecting their transmitted signals whenever the MCU is mixing audio. Receiving terminals may optionally use this information to add appropriate delay in the audio path to achieve lip synchronization.

- Section 5.2.8 - As stated in other submissions, when the MC functionality is in the Gatekeeper, the H.245 channel is still going to the MC. We should remove references such as "the terminal and a Gatekeeper"

- Page 25 - Gateway B - Is this a legal configuration for the H.320 side of the diagram?
- Section 5.4 Bullet 3 - Given that the admission request can be a null function, the Bandwidth request should be allowed to be null for consistency.
- Section 5.5 Paragraph 4 - See Section 2.3 of this document for recommendation.
- Section 5.6 - The MP maybe the MCS top provider in a distributed multipoint conference. See Section 2.3 for further comments.
- Section 5.8.2 - "The MC shall determine which terminal or terminals are actively multicasting video." How is this signaled? We have no command to halt and then restart video unless we consider flow control with value zero.
- Section 5.8.2 Paragraph 4 - See MC Section 2.3 for comments.
- Section 5.8.3 and 5.8.4 - Terminals should have the ability to setup a conference of the type they wish (e.g. Hybrid Centralized Audio). We currently have no way to signal this at call setup time as it is solely determined by the MC. Given that on a dynamic basis that best conference model to use is probably only known by the creator of the conference, we need to add a conference model type to the admission and setup message. See Section 2.3 for comments.
- Section 5.8.7 - Add the following text:

    An MP which is providing audio mixing in either the Centralized or Hybrid multipoint conference models must maintain audio and video synchronization by modifying the time tags of the audio and video streams to take into account its own time base. Further, when the MP mixes audio or switches video to generate a new stream sourced from the MP, the MP must generate its own sequence of sequence numbers in the audio and video packets.

    When mixing audio, the MCU must synchronize each of the incoming audio streams to its own timing, mix the audio streams, and then generate a new audio stream based on its own timing with its own sequence numbers. If the MP is also switching video, the switched stream must have its originating time stamp changed the match the MCU time-base to synchronize it with the mixed audio stream and must have a new sequence number representing the stream from the MP.

- Section 6.1 - Terminals must provide a Signal and RegistrationAndStatus (for lack of a better term) address. The E.164 and H.323 ID are optional but the E.164 address is required if calls are to be accepted from an inbound Gateway call. The Gatekeeper must translate E.164 addresses and H.323 IDs. (Note: Given that Paragraph 4 dictates that the H.323 ID must be unique, the Gatekeeper must support them) It should also be noted that the E.164 address must be unique within the Zone!
- Section 6.1 Paragraph 3 - Not sure what the access code text provides to the standard. It could be left out without any implications.
- Please refer to Section 2.2 of this document for comments on Registration.
- Section 6.3 Paragraph One: Change "reliable well known ports" to "the H.323 well known terminal signal port".
- Section 6.3 Paragraph Two: Change "unreliable well known port: to "well known Registration and Status port". Change "between reliable well known ports" to "on the Gatekeeper well known Signal port and the registered Endpoint Signal port using a reliable transport"
- In Section 6.3, Change "Each Gatekeeper" in item 1) to "Each Gatekeeper and terminal". Note: We need to specify the four well known ports for both IP and IPX which we plan to use for H.323.
- Section 6.3.1 - Note that these requests are on the Registration and Status ports and completed using an unreliable transport. Yes, it is the same as 7.4.2.
- Section 6.5 - "LAN transport of the endpoint containing the MC" - we need to add signaling so that endpoints and Gatekeepers can unambiguously determine what the LAN transport address of the MC is in any given conference. See Section 2.6 of this document
- Section 6.5 Paragraph 4 - "An endpoint receiving a Setup message with a non-zero CID is being asked to join an existing conference" In paragraph 2, when the call is routed through a Gatekeeper the initial CID seen by the called terminal is not zero. I think this is OK but designers should know that new conferences and invites are not distinguishable.

- Section 6.5 editors note - The CID is in the Status Response message.
- Section 7.1.1 Paragraph 1 - change “well known reliable port” to “well known Terminal Signal port”
- Section 7.1.1 - Message (9) should also be optional.
- Section 7.1.2 Paragraph 2 - “a reliable port that is on the Gatekeeper”. This needs to be a complete transport address to specify the Gatekeeper.
- Section 7.1.3 - The two call models cannot be mixed. Remove “If the Gatekeeper does not wish to route the call signaling, the Gatekeeper shall return the LAN transport address of end point 1 in the ACF.” First, the ARQ/ACF does not establish H.245 control ports, the Setup/Connect combination do; therefore the Gatekeeper does not know the H.245 address of endpoint one to return. Second, the Gatekeeper has no choice when it receives a “Request to answer a call” from a terminal, it must act in the “Non Routing” model as by receiving the request we know that the call is not being routed through the Gatekeeper cloud.
- Section 7.1.3 Paragraph 2 and Figure 16 should be deleted as they cannot be signaled as diagrammed. (See the previous bullet on Section 7.1.3)
- Section 7.1.4.1 - The note about the Gateway issuing Call Proceeding messages is only relevant to calls where one stage dialing is available.
- Section 7.1.4.1 Paragraph 2 - Are we stating that both DTMF and SBE numbers are mandatory? Should we define procedures for SBE numbers?
- Section 7.1.4.1 Paragraph 2 - Two stage dialing from H.324 should be via User Information which is mandatory for H.324 terminals. (Not DTMF)
- Section 7.1.4.2 Paragraph 1 - “Destination E.164 address” should be “Destination E.164 addresses” as numbers for each channel and the final E.164 extension must be provided.
- Section 7.4.1 Paragraph 1 & 3 - First, Terminals should not need to ask for permission to lower their bit rates, only to increase it. A terminal can slow the remote end of call down using the flow control command without the permission of a Gatekeeper. For increases, terminals should ask for permission as described, then close and reopen the Logical channel at the new rate. (This provides the ACK in Figure 17 and it allows the remote end of the link to ask permission from its own Gatekeeper). The statement “The H.245 flow control commands shall not exceed the bandwidth approved by the Gatekeeper” does not apply as the flow control commands should only be used to limit the bandwidth on any given logical channel from the maximum rate negotiated during the capability exchange (e.g. lower than requested in the ARQ). See Section 2.11 for more information.
- Section 7.4.1 - The Gatekeeper should never request an increase in bandwidth. It does not necessarily know the capabilities of the endpoints so it cannot request an increase. Further, what media would it be requesting the increase in?
- Section 7.4.1 - See Section 2.11 for proposed text.
- Section 7.4.3 - See Section 2.6 for comments.
- Section 7.5.3 (Editors Note) - Yes, this is sufficient.
- Section 9.1.3 - Why is the RTP header excluded from encryption?

2.2 Registration

There are only two legal types of Binding which can occur:

1) Auto - the mandatory way. Terminals broadcast a GRQ and then register with one of the answering Gatekeepers.
2) Deterministic - an option where terminals are configured to send their registration requests to a specific Gatekeeper.

<u>Terminals and or Gatekeepers configured for Manual binding and or registration are not interoperable with standard H.323 devices so this type should be eliminated from the text. The discussion in the 12/28/95 draft of the manual Binding and Registration procedures are very confusing and should be deleted.</u>

- Section 6.2 - "A Gatekeeper must be aware of the terminals ..... When a Gatekeeper is present, all endpoints shall register with it" . Note: Gatekeepers are only aware of those terminals who register with it. Terminals must attempt to auto bind even if there is no Gatekeeper on the network (So if one is introduced later, all terminals will eventually bind). Therefore, the first two sentences of this paragraph should be deleted.
- Section 6.2.1 - Auto Binding has been **agreed to be mandatory** and should be the rule; therefore the auto binding should be changed to reflect that it is mandatory.
- Registration is also mandatory. **The discussion of manual registration is a completely proprietary solution and is not interoperable with standard H.323 devices and should therefore be deleted from the text.**
- The following text is proposed for H.323:

6.2 Endpoint Binding, Registration and Unregistration

As part of their configuration, endpoints must be either configured to register with a specific Gatekeeper at program initialization or they must attempt to Auto Bind with a Gatekeeper at each program initialization. The Auto Bind procedure will guarantee that when a Gatekeeper is introduced onto a network, endpoints will find and register with it at their next program initialization.

6.2.1 Endpoint Auto Binding
If an endpoint is not configured to register with a specific Gatekeeper at start up it must attempt to Auto Bind with a Gatekeeper at each program initialization. Upon initialization, an endpoint will broadcast a Guardian Request (GRQ) on the H.323 Gatekeeper well known Registration and Status port which in effect asks "Who is my gatekeeper?" Any Gatekeeper which receives the GRQ and has not been programmed to ignore the request should respond with a Guardian Confirmation (GCF) "I can be your Gatekeeper". An endpoint may now select a Gatekeeper for registration by sending it a Registration Request (RRQ) and following the procedures as outlined in Section 6.2.2. Once bound to a Gatekeeper, endpoints should perform admission requests to the Gatekeeper prior to placing or answering subsequent calls. This procedure will ensure that an IT department can control H.323 endpoint equipment by introducing a Gatekeeper on to the network.

6.2.2 Endpoint Registration
As part of their configuration, if an endpoint finds a Gatekeeper through auto binding or is made aware of one through some static configuration, each terminal must register with the Gatekeeper.

Registration is the process by which an endpoint informs the Gatekeeper of its LAN transport address, E.164 address, and/or its H.323 ID.

Endpoints shall send a Registration Request (RRQ) to the selected Gatekeeper and include both a signal and Registration and Status address for subsequent communications. Endpoints registering with a Gatekeeper have the option of registering either the H.323 well known terminal Signal and Registration and Status ports or other dynamic ports of their choosing. The RRQ is sent via an unreliable transport to the Gatekeepers well known Registration and Status port. The Gatekeeper shall respond with either a Registration Confirmation (RCF) or a Registration Rejection (RRJ) to the Registration and Status port that the terminal included in the RRQ message. The RRQ may be repeated periodically (i.e., at terminal power up) so the Gatekeeper shall be able to handle multiple requests from the same endpoint. See Figure 7/H.323. Repeated requests to register (i.e. at terminal power up) should not set the Bind Request flag to BIND.

6.2.3 Endpoint Unregistration

During normal program termination, each endpoint shall send an Unregister Request (URQ) to its select Gatekeeper. The Gatekeeper shall respond with either a Unregister Confirmation (UCF) or an Unregister Rejection (URJ).

The unregistration message may not arrive when a terminal fails, so the Gatekeeper should have a mechanism to periodically remove inactive terminals.

The Gatekeeper shall assure that each E.164 address or H.323 ID translates uniquely to a single LAN transport address. Ambiguous registrations shall be rejected by the Gatekeeper. The combination of the unregistration and registration messages can be used by endpoints to change their E.164 or H.323 address or ID.

- Section 6.2.1 - The Autobind must be broadcast as it is mandatory function. Multicast capability is optional and cannot be providing on all network interface cards, network transports (IPX), and/or network configurations (router configurations, etc.). Note: Their is no such thing as a reliable broadcast.
- Section 6.2.2 - Endpoint Registration is mandatory given that a Gatekeeper is on the network. A terminal which is only capable of manual registration cannot be an H.323 terminal.
- The ports used in this section should be referred to as the "registration and status" port or by the name we chose to use in place of the registration and status port.

2.3 MC Issues and Proposed Changes

- Section 5.5 Paragraph 4 - "The choice of conference mode occurs after connection with the MC using H.245 signaling" - Endpoints need to be able to signal the type of conference they would like to setup. See Section 2.6.
- Section 5.8.2 Paragraph 1 - "The terminal shall have the capability to multicast" must be changed to "may". The ability of a terminal to multicast is dependent on the network interface card, the network transport, and the network configuration in use.
- Section 5.8.2 Paragraph 3 - What command does the MC use to stop an endpoint from multicasting video?
- MCs can exist in endpoints and Gatekeepers. Currently their is no way for an endpoint to request or not request an MC on the call and further given that an MC is desired on a call, their is no information available to the Gatekeeper to decide which one of many MCs it may know about to swap on to the call. I propose the following:
    - During Registration, MC's should be split into four terminal types: 1) MC no MP, 2) MC data MP, 3) MC audio/data MP, and 4) MC with audio/video/data MP.
    - In the Admission Request, endpoints be allowed to signal a request flag for requesting an MC on the call, the conference type as defined in H.323 (centralized, distributed, hybrid central audio and hybrid central video) and an estimate of the total of number of nodes to be on the call. This will provide the Gatekeeper enough information to make an intelligent decision about which MC, assuming their are multiple, to swap on the call. Most importantly, if a terminal has an MC in it, it can prevent the Gatekeeper from swapping in an external MC when the terminal does not want that to occur.
    - Given that MCs can have different types of MP's an endpoint should be able to request MC type in the admission request.
    - The Admission Reject reason code should have the following reject reasons added: MC not on network, and MC type not available on network.
- This still leaves locating the MC on a call as an issue. Please refer to section 2.6 for a proposal.

2.4 Maximum Jitter Delay

Given, that there is no way to bound jitter on a non-QoS guaranteed LAN without help from a protocol such as RSVP, bounding the maximum delay jitter of the H.323 transmitter will have limited use.

However, given that RSVP and other QoS protocols will be used to specify a maximum jitter delay, H.323 should specify a way for a transmitter to further refine its jitter specifications. The following text is adapted from H.324. An associated change in H.245 is proposed in Section 4.1 of this document.

For H.323 (Section 5.2.5)

Audio packets shall be transmitted periodically at an interval determined by the audio codec Recommendation in use (audio frame interval). The transmission of each audio packet shall commence no later than 10 milliseconds after a whole multiple of the audio frame interval, measured from transmission of the first audio frame (audio delay jitter). Transmitters capable of further limiting their audio delay jitter may so signal using the H.245 **maximumDelayJitter** parameter of the **H225Capability** message, so that receivers may optionally reduce their jitter delay buffers.

2.5 Conference Models

The proposed models for multipoint in the current draft are a great simplification! Please disregard the hybrid models proposed in ptel14.doc.

2.6 Ad hoc Conferencing

One of the main advantages of a conference on the LAN is the ability for three or more terminals to conference together in an ad hoc way without having to worry about who calls who and what the conference ID is. The current draft only specifies a invite mechanism for creating a conference and contains a number of problems which are listed below:

- The text equates the Direct Endpoint signaling with an MC in a terminal and Gatekeeper Routed calls with an MC in the Gatekeeper. The specification should allow for active MC's in an endpoint even though the call model is Gatekeeper routed and vice-a-versa.
- 7.4.3.1 Item 3) - Note that the MC is within the terminal and that the H.245 channel is not moved. How the Endpoint 2 attaches the terminals to the MC is beyond the scope of the recommendation.
- In 7.4.3.1 (Second Example) Item 4, the terminal knows that it is Direct Call Model and it lost the M/S determination and therefore assumes that the MC is the signal address of the other terminal. (See Section 2.6.2).
- In 7.4.3.2 (First Example) Item 4, the terminal knows it is Gatekeeper Routed Model and it lost the M/S determination and therefore assumes that the MC is the signal address of the Gatekeeper. (See Section 2.6.2)
- In 7.4.3.2 (Second Example) Item 4, the terminal knows it is Gatekeeper Routed Model and it lost the M/S determination. In this case the terminal (and the Gatekeeper) do not know where the signal address of the MC truly is. The endpoint only knows to send the setup message to the Gatekeeper and the Gatekeeper does not know who won the M/S determination procedure.
- We need to add procedures for a terminal to join a conference as described below.
- If Endpoints 1 and 2 are in a call, endpoint 3 should be able to call endpoint and if the MC is configured to allow it, be joined into the call without the user having to prearrange or be invited.
- If a Gatekeeper uses the Direct Endpoint Call model it cannot swap an MC onto a call.

Our Solution for ad hoc conferencing should offer the following:

1. Support for conferences where one of the terminals is an MC. (7.4.3.1 in H.323)
2. Support for conferences where there is an MC on the network which is not collocated with one of the terminals. (7.4.3.2 in H.323)

3. Support for a third terminal to join a conference automatically (if approved by the MC of the conference). (See Proposal Below)
4. All endpoints and Gatekeepers need to be able to locate the MC on a call. Currently it is ambiguous where the MC on calls signaled with Gatekeeper Routing. (See Proposal Below)

To Solve the Issues we need to:
1. Provide the Gatekeeper with a way to determine whether or not to swap an MC (and what kind of MC) onto a call at call "setup" . See Section 2.3.
2. Endpoints do not know where to send call setup messages as the MC is swapped into the call without their knowledge. To solve this, we need to add an H.245 command which can be sent by the MC identifying the signal address to use to reach the MC.

The proposed H.245 command would be:

```
MCLocationCommand     ::==SEQUENCE
{
        signalAddress          OCTET STRING
        portNumber             INTEGER(0..65535)
}
```

The following Categories and Scenarios for ad hoc calls exist:

Categories:

1. MC in a terminal
    - Direct Endpoint Signaling
    - Gatekeeper Routed Call Signaling
2. MC Swapped in by Gatekeeper (either within the Gatekeeper or elsewhere on the network)
    - Direct Endpoint Signaling
    - Gatekeeper Routed Call Signaling (Routed H.245)
    - Gatekeeper Routed Call Signaling (non Routed H.245)

Scenarios

1. MC invites an endpoint into call.
2. A non MC endpoint invites another endpoint into the call.
3. Another endpoint calls the MC to join the call.
4. Another endpoint calls a non MC endpoint to join the call.

The proposed H.323 changes are then:

7.4.3.1 MC in Terminal Ad Hoc Conferencing

Scenarios A though C apply to both Direct Endpoint and Gatekeeper Routed Call Signaling.

Scenario A
In an Ad Hoc Multipoint call having direct endpoint or Gatekeeper Routed call signaling, if the endpoint containing the MC wishes to add another endpoint to the conference, the following procedure is used:

1. Endpoint 1 calls endpoint 2 with CID=0 according to the procedure in Section 7.1 and endpoint 2 responds to endpoint 1 with the CID=N.
2. Using H.245 master/slave determination procedure, it is determined that Endpoint 2 is the master, and hence has the MC.

3. Endpoint 1 and endpoint 2 may be attached to the MC now, or when the user initiates the multipoint conference function, at the choice of the manufacturer.
4. *The MC (Endpoint 2), sends the MCLocationCommand to Endpoint 1.*
5. Endpoint 2 calls endpoint 3 with CID=N according to the procedures in Section 7.1.
6. Endpoint 2 attaches endpoint 3's H.245 control channel to its MC . All negotiation of multicast vs uni-cast is done via H.245 .
7. The MC may chose to send **multipointModeCommand** at this time.

Scenario B

In an Ad Hoc Multipoint call having direct endpoint or Gatekeeper Routed call signaling, if the endpoint that does not contain the MC wishes to add another endpoint to the conference, the following procedure is used:

1. 1) through *4)* are the same as Scenario A.
2. Endpoint 1 sends a Setup message to the MC indicating a call to endpoint 3 with CID=N.
3. The MC calls endpoint 3 with CID=N according to the procedures in Section 7.1. Thus, the MC remains at the center of the conference; in effect it is acting as a dial-out MCU.
4. The MC attaches endpoint 3's H.245 control channel to its MC . All negotiation of multicast vs uni-cast is done via H.245 .
5. The MC may chose to send **multipointModeCommand** at this time.

It should be noted that the call is ended by a failure of the endpoint that is providing the MC.

Scenario C

In an Ad Hoc Multipoint call having direct endpoint or Gatekeeper Routed call signaling, if another endpoint calls either endpoint in the call to join the call the following procedure is used:

1. 1) through *4)* are the same as Scenario A.
2. Endpoint 3 calls endpoint 1 or 2 with CID=0 according to the procedures in Section 7.1.
3. Endpoint 1 or 2 sends a release complete to Endpoint 3, and if it is configured to provide conference information include the CID of the current call and the signal address of the MC of the conference.
4. Endpoint 3, if it is configured to do so, can now place a call to the MC of the conference with the proper CID using the procedures in Section 7.1.
5. If the MC allows the terminal to join. the MC attaches endpoint 3's H.245 control channel to its own . All negotiation of multicast vs uni-cast is done via H.245 .
6. The MC may chose to send **multipointModeCommand** at this time.

7.4.3.2 MC swapped on to the call by the Gatekeeper

A Gatekeeper cannot swap an MC into the call using Direct Call Signaling so the following procedures apply only to Gatekeeper Routed Call Signaling Scenarios:

Scenario D

In an Ad Hoc Multipoint call having Gatekeeper Routed call signaling and Gatekeeper routed H.245 control channel, if either endpoint wishes to add another endpoint to the conference, the following procedure is used:

1. Endpoint 1 calls endpoint 2 via the Gatekeeper. endpoint 1 sends CID=0, the Gatekeeper sends CID=N to endpoint 2, and return CID=N to endpoint 1, according to the procedures in Section 7.1.

2. Using H.245 master/slave determination procedure, it is determined that the Gatekeeper is the master, and hence has the MC (or has access to it).
3. The Gatekeeper may route the H.245 control channels from endpoint 1 and 2 to an MC at this time, or later when endpoint 3 is connected . Whether endpoint 1 or 2 has an MC, the Gatekeeper by definition shall always win the master/slave determination process .
4. *The MC (The Gatekeeper), sends the MCLocationCommand to Endpoint 1 and Endpoint 2.*
5. Endpoint 1 sends a Setup message to the MC indicating a call to endpoint 3 with CID=N.
6. The MC calls endpoint 3 with CID=N according to the procedures in Section 7.1. Thus, the MC continues to route the H.245 control channels.
7. The MC routes endpoint 3's H.245 control channel to the MC . All negotiation of multicast vs uni-cast is done vie H.245 .
8. The MC may chose to send **multipointModeCommand** at this time.

<u>Scenario E</u>
In an Ad Hoc Multipoint call having Gatekeeper Routed call signaling and Gatekeeper routed H.245 control channel, if either endpoint receives a request to join the conference from an endpoint outside the call, the following procedure is used:

1. 1) through *4)* are the same as Scenario D.
2. Endpoint 3 calls endpoint 1 or 2 with CID=0 according to the procedures in Section 7.1.
3. Endpoint 1 or 2 sends a release complete to Endpoint 3, and if it is configured to provide conference information include the CID of the current call and the signal address of the MC of the conference.
4. Endpoint 3, if it is configured to do so, can now place a call to the MC of the conference with the proper CID using the procedures in Section 7.1.
5. If the MC allows the terminal to join. the MC attaches endpoint 3's H.245 control channel to its own . All negotiation of multicast vs uni-cast is done via H.245 .
6. The MC may chose to send **multipointModeCommand** at this time.

2.7 Call Forward and Call Transfer

Re: H.323 ISDN Supplementary services

We can add two services to the specification today that do not require extra PDU's, add considerable value to the current specification, and are simple enough that translating them at a Gateway to a Q.xxx specification should be trivial. These services are:

1. Call Forward - The ability to forward all calls to another terminal.
2. Call Transfer - The ability to transfer an active call to another terminal.

<u>We need these functions in the first release of the specification for the following reason:</u>
When dialing inbound via a Gateway, the far end system may not be able to produce the desired SBE codes, have DTMF capability, or have the access to any secondary dialing capability. In those cases, it would be desirable for the Gateway to route all calls to a single site and have that site transfer the calls (i.e. an operator) to the appropriate node. This requires a call transfer function.

To add call forward and call transfer we need to add the following fields to the user information in the Release Complete Message in 7.11 of H.225.

- Release Complete Reason - Add the following to codes
  Call Forwarded (14)
  Call Transferred (15)

Add a destinationAddress type NetworkAddress to the Release Complete user field.

2.8 Audio Processing

Add Section from H.324 to H.323 Section 5.2.5.2 (Audio Frame Size (e.g. H.324 Section 6.7.3)

All H.323 terminals offering audio communication shall support the G.711 codec using H.225. For all frame-oriented audio codecs, 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, 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.

2.9 Asymmetric Operation

In H.323 Section 5.2.5, the statement "The H.323 terminal shall be capable of asymmetric operation for all audio capabilities it has declared, e.g. it shall be able to send G.711 and receive G.728 if it is capable of both."

There is no precedent for this statement in other H.series documents. H.245 allows for a node to support G.728 and G.711 without being able to do so asymmetrically. Therefore, the "shall" should be changed to a "may".

2.10 Bandwidth Change Procedures

The following text is proposed for Bandwidth Change procedure in H.323:

Call bandwidth is initially established during the admissions exchange. The bandwidth number in the admissions request should represent the best guess at the total transmit and receive audio and video traffic in kbps. This estimate should not include data which is bursty and typical of other LAN traffic. At any time during a conference, the endpoints may request an increase or decrease in the call bandwidth and the Gatekeeper may request a decrease in call bandwidth. Any bandwidth change shall be in accordance with the declared capability of the endpoints.

An endpoint which wishes to decrease its bandwidth can do so without permission. If the endpoint wishes to reduce its bandwidth and further release the bandwidth it has reserved via the admission request it can send a BRQ to the Gatekeeper to decrease its bandwidth. An endpoint wishing to increase its transmitted bit rate sends a BRQ(1) message to the Gatekeeper. The Gatekeeper determines if the request is acceptable using a criteria which for this determination is outside the scope of this Recommendation. If the Gatekeeper determines that the request is not acceptable, it returns a BRJ(2) message to endpoint. If the Gatekeeper determines that the request is acceptable, it returns a BCF(2) message. The endpoint may now change its transmitted bit rate by closing the appropriate logical channel and then reopening a new one at the new bit rate.

An endpoint wishing to decrease the transmitted bit rate of another endpoint in a conference, should use the H.245 flow control commands. To request an increase in transmitted bit rate, a terminal should use the H.245 Request Mode command.

A Gatekeeper wishing to decrease the transmitted bit rate of an endpoint sends a BRQ(1) message to the endpoint. The endpoint shall always comply with this request from the Gatekeeper and returns a BCF(2). The Gatekeeper shall not request an increase in bandwidth.

If the Gatekeeper is managing the H.245 control channel, and the call is point-to-point, the endpoints shall still send the BRQ. Note that the endpoint is not aware of whether the H.245 channel is managed by the Gatekeeper.

If the call is multipoint, and the endpoints are in receipt of **multipointModeIndicate**, they shall not issue BRQs; it is assumed that the Gatekeeper has reserved sufficient bandwidth for all its requests.

3.0 Comments on 12/28/95 H.225.0 Draft

3.1 Specific Changes

- Figure 1 - H.22Z references should be updated to H.225
- Section 6.1 Item 2 - The option to close the channel if a Gateway is not on the call is a good compromise and should be retained. If the Gatekeeper is so low on resources that it cannot reopen the port later in the call (for the H.320 node), it certainly should not be holding that port open for every calls that does not contain a Gateway just in case a H.320 terminal is added!
- Section 6.1 Paragraph 7 - Reference H.245 for mode switching procedures.
- Section 6.2.2 - Editors Note - As discussed on an audio call, the MTU for Ethernet is 1500 bytes. Sending packets larger then the network MTU does not minimize the packet rate.
- Table 3-1 - Per the latest H.323, Release should be made mandatory to support call clearing with the Gatekeeper. Connect Acknowledge should be optional.
- Section 7.10 - Release should be retained.
- Section 7.11 - Update text to reflect use of Release command.
- Section 7.16 - If Bandwidth refers to the total for the call, then the bandIncDec is redundant information.

3.2 PDU Review

3.2.1 General Comments

1. Change the address definitions to match the port proposal: "RegStatus" for Registration and Status, "Signal" for Signaling, and "Control" for the H.245 channel.
2. Mixed Q.931 and ASN.1 PDU definitions are very confusing. We should adapt the Q.931 messages to an ASN.1 definition within H.225 so that call setup on the LAN is completely defined within H.225. We could even leave out all of the shaded Q.931 fields which do not apply to the LAN. The current specification which contains only half of the information needed to implement call setup for an H.323 terminal will lead to confusion, misunderstanding, and interoperability problems.
3. Make the Connect Acknowledgment and Setup Acknowledgment optional. They do not provide any information given that we are using TCP.
4. The Progress message defined in the text is not listed in Table 3-1 on page 18.
5. Editors Notes in Table 3-1 - One reason to keep an acknowledgment scheme for call disconnect is to insure that the far end (the end being hung up on) does not confuse an aborted call with a call which has terminated normally. We should keep an acknowledgment scheme.
6. The Resume and Suspend messages included in table 3-1 are not described. We should either include them and describe what a LAN terminal should do if it receives one of them or we should remove

them. The same comments go for the Congestion Control, Facility, Information, Notify, Status and Status inquiry messages.

7. A Signal and a Registration and Status port are need for communications. These additions are proposed below. The definition for the addresses in the text boxes for each PDU should be described as follows:
    - SignalAddress - this is an array of transport addresses; one for each transport to use when signaling Q.931 PDU's. This address includes the port information.
    - RegStatusAddress - this is an array of transport addresses; one for each transport type to use when signaling non-Q.931 H.225 PDU's. This address includes the port information.

3.2.2 Specific Comments to Q.931 PDUs

1. Connect
    - The destination address should be called controlAddress for consistency. It is the address the terminal receiving the Connect message should attempt to establish logical channel 0 for H.245 communications.
2. Setup
    - Missing information needs to be added for further review.
    - Editor's Note "Calling number is in Q.931" - we want to provide all of the H.323 identifier's to the user at the far end of the call, not just the Q.931 number. We should add the H.323 ID and any other identifier we chose to carry in this PDU.
    - Editor's Note: on "the terminal is sending to this address so there is no need to include it" - if this refers to the destination address - if the call model is through the Gatekeeper and the callee is unbound, the Gatekeeper does not know where to send the connect request unless it is in this packet.
    - The caller's H.245 Control Address needs to be added to this PDU.
    - The destination WAN information needs to be included in its entirety. Gateways need the numbers of all channels when calling an existing H.320 system, the Gatekeeper or Gateway will need the channel rate information to select a Gateway or to dial in the way prescribed by the user (e.g. dial restricted because you are calling a system connected to ACCUNET).
    - We need to specify all of the channel numbers to dial and then another E.164 number for a potential LAN node when dialing via two Gateways to another H.323 terminal.
    - The conference ID determination and setting as specified in AVC-827 has been dropped, this needs to be added to allow for adhoc conferencing.
3. Release Complete
    - Reason - Add the following codes:
      Call Forwarded (14)
      Call Transferred (15)
      OngoingConference (16)
      PrivateConferenceInProgress(17)
    - Add a destinationAddress type NetworkAddress to the Release Complete user field. (where to forward or transfer)
    - Add a MCSignalAddress type NetworkAddress (provides MC of ongoing conference, who to contact to join)
    - Add ConferenceID as specified in AVC-827 (provides Conference ID of an on going conference)

3.2.3 Specific Comments to non Q.931 PDUs

1. Network Address
    - The Subnet mask should be included to allow the IP network number to be extracted from the full IP address (net number + host ID). The extracted number uniquely identifies on which physical segment a terminal resides. (Subnet mask same format as IP address)

2. Node Type
    - Do we need to specify what type of MP an MCU has associated with it? (MC no MP, MC Audio/Data MP, MC Audio/Data/Video MP).
3. GuardianRequest
    - Change terminalAddress to signalAddress
    - Add RegStatusAddress type NetworkAddress.
4. GuardianConfirmation
    - Remove YearOfSpec
    - Change gatekeeperAddress to signalAddress
    - Add RegStatusAddress type NetworkAddress
    - Change extensionCount to propExtension.
5. Registration Request
    - Add RegStatusAddress type Network Address.
6. Registration Confirmation
    - Change gatekeeperAddress to signalAddress
    - Add RegStatusAddress type NetworkAddress
    - The gatekeeperAddress description (now signalAddress) is for non Q.931 H.225 messages sent from a terminal, not an address "that the gatekeeper will respond to".
7. Registration Reject
    - Add duplicate Ext Number as a reject reason. A single Gatekeeper should not allow two terminals with different network addresses and identical E.164 extensions to register.
8. Add UnRegistration Request (URQ)

```
UnregistrationRequest        ::==SEQUENCE --(URQ)
{
    terminalIdentifier              OCTET STRING(SIZE(128)),
    terminalExtNum           E.164Address
    propExtension            ToBeAdded
}
```

9. Add UnRegistration Confirmation (UCF)
    No parameters
10. Add UnRegistration Reject (URJ)

```
UnregistrationRejection      ::==SEQUENCE --(URJ)
{
    rejectReason      UnregistrationRejectReason,
}

UnregistrationRejectReason   ENUMERATED
{
    Call In Progress  (1),
    Not Currently Registered  (2),
}
```

11. Guardian Query Response
    - Change gatekeeperAddress to signalAddress
    - Add RegStatusAddress type NetworkAddress.
12. Admission Request

As documented on the IMTC audio call on December 7th, the ARQ message and response need to include complete destination (H.323 ID + network address + WAN information (number of channels requested or WAN channel rate) in their PDUs. To summarize, this information is needed for the following reasons:

- The Gatekeeper is managing the location of Gateways. Upon receiving an ARQ the Gatekeeper must first, be able to recognize that the terminal is requesting an outbound call and second, be able to return the address of a Gateway which can complete the call with the desired resources (channel rate).
- The Gatekeeper needs the end point terminal network address to allow for calling terminals which do not have Gatekeepers or terminals managed by another manufacturer's Gatekeeper.
- In the call model where the terminal calls directly but asks his Gatekeeper for permission, the destination information is the only information that allows the Gatekeeper to make an informed decision on whether it should allow the call for a particular network segment.
- In H.323 Section 6.1 it states that "When there are no Gatekeepers in the system, the calling terminal may address the called terminal directly using the network address of the called terminal". For the call model where each end point talks to its own Gatekeeper, the terminal must still be able to provide the destination network address of the called terminal so that if the two systems have different Gatekeepers or if one terminals does not have a Gatekeeper, the call can still be placed.

- This message is sent to a Gatekeeper from a terminal who is registered to it. The replyAddress was previously included in the Registration message and can be removed from this PDU.
- Add complete destination information (E.164 extension, H.323 ID, and all WAN dialing numbers and channel rate).
- Under Call Type, add the option "Request to Answer Call" for the admission request sent to a Gatekeeper by a terminal which is answering a call. Depending on the call model the Gatekeeper has implemented this information is required.
- Add CID to allow for conference additions (invites and joins) to be distinguished from new conferences.
- Add MCRequest for endpoint to request an MC be swapped in by the Gatekeeper
- Add MCTypeRequest (Node Type) to provide which type of MC is required
- Add Conference Type
- Add Conference Size Estimate

13. Add the following error codes to Admission Reject
    - MC not on net, MC type not on net
14. Bandwidth Request
    - Same comment on replyAddress.
15. Bandwidth Confirmation
    - Given that the bandwidth returned is not guaranteed to be there when requested, we should remove it. How would this be used?
16. Bandwidth Reject
    - Change reason "RequestDenied" which is redundant to "Insufficient Bandwidth".

3.3 Optional Messages

We should consider the following optional messages for inclusion within H.225 to promote some level of interoperability within the Gatekeeper cloud. These messages would allow inbound calls through a Gateway to reach any node within the Gatekeeper cloud as setup in any installation. Further it could allow for the rejection of duplicate E.164 terminal addressees during the registration process.

The following three messages would allow for the setup of a hierarchy of Gatekeepers at any given site to allow for interoperation in the Gatekeeper cloud. It will further allow for in bound Gateway calls to be able to reach any terminal at a site regardless of which (assuming there are multiple) Gatekeeper they are registered with. All of these messages are sent on the Gatekeeper well known Registration and Status port are proposed to be optional.

```
GatekeeperTerminalQuery        ::==SEQUENCE --(GTQ)
{
        requestSeqNum   INTEGER (1..65535),
        replyAddress            NetworkAddress,
        terminalExtNum  E.164Address,
        propExtension           ToBeAdded
}

GatekeeperTerminalConfirmation        ::==SEQUENCE -- (GTC)
{
        requestSeqNum   INTEGER (1..65535),
        terminalExtNum  E.164Address,
        SignalAddress           NetworkAddress,
        propExtension           ToBeAdded
}

GatekeeperTerminalReject       ::==SEQUENCE -- (GTR)
{
        requestSeqNum   INTEGER (1..65535),
        terminalExtNum  E.164Address,
        propExtension           ToBeAdded
}
```

4.0 H.245 Additions

4.1 H.225 Capability

The following H.225 capability should be added to our proposed list of H.245 changes to handle the following situations:

1. We need a way for a terminal to specify to the far end the maximum size of its receive buffers so that transmitters do not overflow them. An example of where this is needed is the desire of some manufacturer's to send video packets much larger than the network MTU. In this case, some limit must be negotiated to ensure the receiver can accept those packets.
2. In cases where network jitter can be bounded or guaranteed, it is desirable that transmitters be able to further bound their jitter specifications so that the receiver can reduce its jitter queue appropriately.

Add h225Capability to MultiplexCapability:

Where H225Capability includes:

```
{
        maximumDelayJitter          INTEGER (0..1023)      -- units in milliseconds
        maximumAudioPacketSize      INTEGER (1..x) -- units in octets
        maximumVideoPacketSize              INTEGER(1..x)  -- units in octets
}
```

And:

H225Capability: indicates capabilities specific to the H.225 Media Stream and Packetization Service.

maximumDelayJitter indicates the maximum peak-to-peak transmission jitter that the transmitter shall cause. It is measured in milliseconds. Transmission jitter is defined as the difference in time of delivery of each audio packet to the network compared to when it would be delivered at a constant bit rate without packetization.

maximumAudioVideoPacketSize indicate the maximum packet size (including all headers) the receiver can handle given its buffer capability.

4.2 Open Logical Channel ACK

Add a h225LogicalChannelAckParameters of type H225LogicalChannelAckParameters instead of LANAddress at the highest level as an optional parameter. This will allow the H.225 parameters to be optional for those H.series documents other than H.323 but indicate that this information is mandatory for someone using H.225.

Then add:

```
H225LogicalChannelAckParameter                ::==SEQUENCE
{
        LANAddress              CHOICE
        {
                unicastPortNumber       as in TD38
                multicastChannel        as in TD38
        }
}
```

4.3 Open Logical Channel Reject

Add "Insufficient Bandwidth" as a cause.