ATIS-0100522.2000(S2019)

Quality of Service for Business Multimedia Conferencing

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ATIS-0100522.2000(S2019), *Quality of Service for Business Multimedia Conferencing*

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American National Standard
for Telecommunications –

# Quality of Service for Business Multimedia Conferencing

1   Scope, Purpose and Application

This document specifies classes of Quality of Service (QOS) sufficient to support Business Multimedia Conferencing on Internet Protocol (IP) networks, defined as equivalent to legacy conference system performance (e.g., H.320 at Basic Rate Interface (BRI) rates).

It also specifies the threshold of perceptible impairment for some user interface parameters. This standard applies to communications between a subset of multimedia end-points, namely Video Teleconference room systems and Desktop systems.

The current scope is limited to H.323 point-to-point communication (Multipoint communication is for further study).

It applies to cases where IP is utilized between the end-points (e.g., IP/Circuit-Switched Hybrid Network connections are for further study, e.g. networks supporting H.320-compliant terminals).
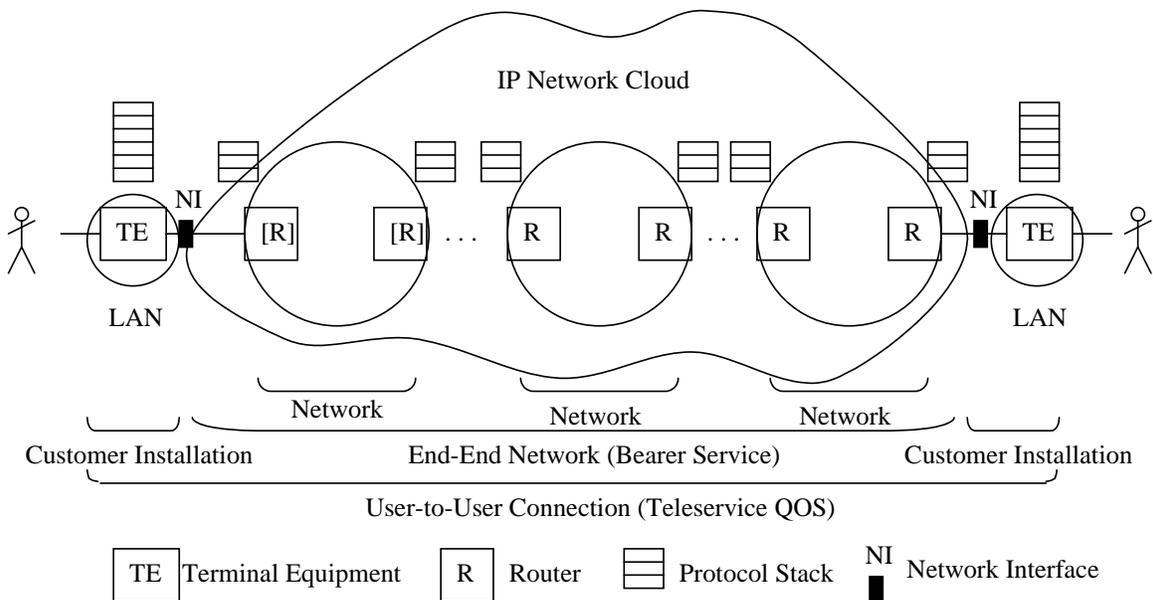
Figure 1 - User-to-User Connection on an IP Network

Figure 1 illustrates some of the terminology of this clause. A user-to-user connection has components on the Customer Installation, and components of one or more network service providers between the network interfaces. The Customer Installation includes a multimedia

communication system (Terminal Equipment, TE), and may contain a Local Area Network (LAN), routers, etc.

Figure 1 includes equivalent ITU-T terms in parentheses (Teleservice and Bearer Service). A Teleservice provides a complete user-to-user communications capability, while Bearer Services provide the transport of signals between network interfaces, and do not include the customer's terminal equipment or local network. Furthermore, the Network Interface shown in Figure 1 is equivalent to the ITU-T user-network interface.

All networks in the Figure are not necessarily identical, especially in regards to the position of functions such as authentication and billing. Resources (e.g., capacity) may be reserved on some networks comprising the connection, and not on others. Some networks may not be IP-routed networks, e.g., dedicated lines, frame relay networks, and other networks where routers are not present within the administrative boundaries (shown as [R] in Figure 1).

The user-to-user QOS is dependent on the quality and performance of all components that work to establish, or comprise the media connection.

This standard will identify objectives for the user-to-user connection, and allocate portions to the Customer Installation (TE and LAN) and end-end network components. There are different suppliers for terminal equipment and network services, and the allocation will assure that the intended user-to-user performance levels are achieved. It will not allocate portions to individual network providers.

The following are intended uses for this standard:

1. Network planning.

2. Network and Customer Installation component development.

3. User-to-User connection planning.

In connection with User-to-User connection planning activities (item 3), it is necessary to stipulate the desired system type(s). Further, there are cases where certain measurement procedures must be agreed to in advance, since the standardized measurement coverage was incomplete at the time this standard was developed. Although this standard specifies IP network performance parameters, some Non-IP services procured must be evaluated using alternate parameter sets. Draft ITU-T Recommendation I.381, in development at the time this standard was prepared, provides a mapping between the IP layer and lower layer parameters. Another consideration is whether the Acceptable Level specified in this standard is the goal.

When referenced for procurement purposes, the purchaser should specify which values of tables 5 through 12 apply.

2   Normative References

The following standards contain provisions which, through reference in this text, constitute provisions of this American National Standard. At the time of publication, the editions indicated were valid. All standards are subject to revision, and parties to agreements based on this American National Standard are encouraged to investigate the possibility of applying the most recent editions of the standards below.

T1.401-1993, *Telecommunications – Interface Between Carriers and Customer Installations – Analog Voicegrade Switched Access Lines Using Loop-Start and Ground-Start Signaling*[1]

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[1] For electronic copies of some standards, visit ANSI's Electronic Standards Store (ESS) at www.ansi.org. For printed versions of all these standards, contact Global Engineering Documents, 15 Inverness Way East, Englewood, CO 80112-5704, (800) 854-7179.

T1.801.01-1995, *Telecommunications - Digital Transport of Video Teleconferencing/Video Telephony Signals - Video Test Scenes for Subjective and Objective Performance Assessment*[1]

T1.801.04-1997, *Telecommunications - Multimedia Synchronization, Delay, and Frame Rate Measurement*[1]

ITU-T Recommendation G.114 (02/96) - *One-way transmission time*[1]

ITU-T Recommendation H.320 (07/97) - *Narrow-band visual telephone systems and terminal equipment*[1]

ITU-T Recommendation H.323 (02/98) - *Packet-based multimedia communications systems*[1]

ITU-T Recommendation I.380 - *Internet Protocol Data Communication Service - IP Packet Transfer and Availability Performance Parameters*[1]

ITU-T Recommendation P.931 (12/98) - *Multimedia communications delay, synchronization and frame rate measurement*[1]

RFC 2330 - *Framework for IP Performance Metrics*[2]

RFC 2679 - *A One-way Delay Metric for IPPM*[2]

RFC 2680 - *A One-way Packet Loss Metric for IPPM*[2]

RFC 2681 - *A Round-trip Delay Metric for IPPM*[2]

3   Definitions for MM Conferencing QOS Parameters

The following tables summarize the performance parameters of interest at the user and network interfaces.

Table 1 - Quality of Service Parameters - MM User Interface

	Quality Criteria		
Communications Function	Speed	Accuracy	Availability and Reliability
Connection Establishment	Set-up Time Transfer Time	Mis-directed	Accessibility Ratio (among media) Connection Failure
User Information Transfer	Delay (Spontaneity)* Delay variation Contention Resolution	Media Quality Media Synchronization*	Dropped Connection
Connection Release	Take-down Time		Release Failure

* = parameter specified in this standard

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[2] Internet Request For Comments (RFC) are available from the Internet Engineering Task Force (www.ietf.org) or from the RFC Editor (www.rfc-editor.org).

Table 2 - Quality of Service Parameters - End-End Network Interface

Communications Function	Quality Criteria		
	Speed	Accuracy	Availability and Reliability
Connection Establishment	Set-up Time Transfer Time	Mis-directed	Accessibility* Connection Failure
User Information (Packet) Transfer	Delay* (Network Latency) Delay variation* (within a single media stream and between streams) Information Bit Rate (Committed Bit Rate* Delivered Bit Rate)	Lost Transport Packet Rate* (combines IP Packet defects, such as Errored & Lost Packets)	Dropped Connection* (IP Availability)
Connection Release	Take-down Time		Release Failure

* = parameter specified in this standard

Table 3 - Mapping Between Interfaces

MM Network	MM User
Latency	Spontaneity
Delay Variation	Spontaneity, Media Quality, Media Sync
Info Rate	Media Quality, Media Sync
Loss Ratio	Media Quality, Media Sync

3.1 User-to-User Connection (Teleservice [I.112])

A type of telecommunication service that provides the complete capability, including terminal equipment functions, for communication between users according to protocols established by agreement between Administrations or recognized operating agencies.

3.2 End-End Network (Bearer Service [I.112])

A type of telecommunication service that provides the capability for the transmission of signals between user-network interfaces.

3.3   Network Interface (user-network interface [I.112])

The interface between the terminal equipment and a network termination where the access protocols apply.

3.4   Customer Installation [T1.401-1993]

All telecommunication equipment and wiring on the customer side of the network demarcation point.

3.5   Frame [T1.801.04]

The smallest division of a media stream (defined by the measurement system) that conveys an independent, self-contained unit of content.

3.6   Active Frame (Non-Repeated Video Frame [T1.801.04])

An output video frame that is distinguishable from its preceding frame(s) in the sequence (when the corresponding input sequence frames possess distinguishable differences). An Active Frame is assumed to have traversed the channel from input to output, and its delay may be included in the visual channel delay distribution. Since Active Frames convey new visual stimulus, they are the basis for calculation of frame inter-arrival time (and subsequently elementary frame rate).

3.7   Connection Set-up Time

The elapsed time from the connection request to notification of remote alerting (or other response, such as station busy) measured at the originating terminal.

3.8   Connection Transfer Time

The elapsed time from the transfer request to notification of remote alerting (or other response, such as station busy), measured at the requesting terminal.

3.9   Mis-directed Connection

A defect where a connection is established with an undesired destination.

3.10   Connection Accessibility

Ability to establish a connection for each requested medium between capable end-points.

3.11   Accessibility Ratio (among requested media)

This is the ratio of Successful Media Accesses to Media Access attempts during a single call.

Sources of Access failure include terminal, network, and rejection of individual media.

Maintenance/cause codes will be deployed to determine the responsibility for failure.

3.12   Connection Failure (Blocking)

The denial of a connection request.

3.13   Spontaneity (Delay [T1.801.04])

The time a particular frame takes to traverse the transmission channel from system input to output (one-way). The average delay may be calculated from a set of individual delay measurements.   Classical summary statistics may also be supplied, as applicable.

3.14   Delay Variation (Media Stream Delay Distribution [T1.801.04])

A range of one-way delays experienced among successive frames.

3.15   Contention Resolution Time

The elapsed time from the request for a shared resource to remote response, such as "success" or "busy," measured at the requesting terminal.

3.16 Media Quality

3.16.1 Frame Interarrival Time [T1.801.04]

The time between successive Active Frames at the output of the channel.

3.16.2 Frame Rate [T1.801.04]

The reciprocal of the Active Frame Inter-arrival Time for the present Active Frame. The Elementary Frame Rate is equal to 1 divided by the difference between the arrival times of the present and previous Active Frames.

3.16.3 Frame Skipping Ratio [P.931]

The ratio of input to output inter-arrival times (or elementary frame rates) for a pair of matching frames. A measure of the change between the input and output Active Frame rates.

Users of this standard may want to consult T1.801.03 for a list of additional end-to-end video quality parameters that can be used to verify and maintain the operational readiness of video transport systems used for business multimedia conferencing. Users of this standard may want to consult T1.518 for an additional end-to-end parameter that can provide an overall indication of speech quality, or can be used to verify and maintain the operational readiness of speech transport systems used for business multimedia conferencing.

3.17 Media Synchronization

3.17.1 Lip-Sync [T1.801.02]

Operation to provide the feeling that the speaking motion of the displayed person is synchronized with that person's voice. The minimization of the relative delay between the visual display of a person speaking and the audio presentation of that person's voice. The objective is to achieve a natural relationship between the visual image and the aural message for the viewer/listener.

3.17.2 Time Skew [T1.801.04]

The time skew between associated audio and video frames at the output, introduced by the transmission system channel, is the offset between the two output frames minus any offset between the corresponding input frames. Following the convention of positive time delay, synchronization time lag is a positive value. If the second channel is leading, the parameter has a negative value.

3.18 Dropped Connection (Cut-off)

A defect where an established connection is prematurely terminated other than by the actions of the calling or called party.

3.19 Take-Down Time

The elapsed time from the disconnect request to notification that the connection is cleared, measured at the requesting terminal.

3.20 Release Failure

The inability to take down a connection between capable end-points upon request.

3.21 Committed Bit Rate

The minimum acceptable bit rate for transmission (does not include the overhead of packet headers).

3.22 Delivered Bit Rate

The sum of information bits delivered at the receiver, divided by the evaluation interval, and expressed in units of bits per second.

4   Baseline Media Quality Models

The objective for the network (and CPE) performance specifications given here is to define levels necessary to achieve conference quality similar to that experienced by users of circuit switched systems today.

In order for the network parameters to have more specific ranges, we also define two baseline conferencing systems with typical applications, below:

Table 4 - System Types with Applications

H.323 System Types	Tier 1 Desktop PC System	Tier 2 Group Conference Room Systems
Description (Typical attributes)	Informal, possibly spontaneous calling between two individuals.	Scheduled/organized meeting with multiple participants at each system, and having much longer duration than Tier 1 (e.g, 1 hour).
Approximate Capacity*	128kbps	384kbps

*Committed Bit Rate will be specified as a performance level

Video
Compression Codec   H.261 (H.263)
Resolution/Format   CIF
Frame Rate (Avg.)   15 fps for Class A and B Scenes (T1.801.01)

Key assumption: small transmission errors and packet losses will be concealed by the receiving terminal and translate into image (frame) repetition or temporary frame freeze condition, as opposed to various forms of block distortion. It is the consensus that this represents a reasonable state of the art for MM terminal equipment. H.323 supports this capability with the video Fast Update request messages defined in H.245. Further studies are needed to assess the comparative performance of packet and circuit-switched video under error conditions.

Audio
Codec   G.711, (G.729)
Processing   Echo Control on Full Duplex System
Level   Set by each user - Terminals are equipped with volume controls

Data
Transport Reliability   Data for application-sharing, file transfer, white-board, etc., employs
                                 reliable byte stream delivery (TCP/IP).

Miscellaneous
Sufficient buffering is present in Terminal Equipment to accommodate minimal oscillator offsets, and terminals employ dynamic buffer adaptation techniques when overflow or underflow is evident.

5   Parameter Values

This clause tabulates the specified user-to-user values for each performance parameter, in terms of the perceptible level, and a lesser acceptable level. The perceptible level is where the maximum benefit of improving this parameter has been achieved, and the need for further improvement is questionable for the specific application. Acceptable is a level where there is evidence that users can still communicate effectively. We also allocate the acceptable user-to-user values between TE guidelines and requirements for Networks. Note that the TE allocation includes the LAN, if present (see Figure 1).

Parameter Definitions for End-End Network Interface Parameters are given in ITU-T Rec. I.380 for the IP layer, unless otherwise indicated. The IETF also specifies methods of measurement for IP Packet performance which are given in [RFC 2678, RFC 2679, RFC 2680, RFC 2681]. Parameters and methods of measurement for user-to-user connections shall be as specified in T1.801.04-1997 or ITU-T Rec. P.931, unless otherwise indicated.

5.1   Delay Performance

Table 5 - Allocation of One-Way Delay for Audio-Visual Teleservices

Audio & Video User-User 1-way Delay	Perceptible Level	150 ms		
	Acceptable Level	400 ms		
		Max TE Delay=250ms	Max Network Delay = 400ms - Max TE Delay = 150ms	
				Max Network Delay Variation = 40ms (Tier 1)
				33 ms (Tier 2)

Note that there are different requirements for Graphic Teleservices, see Table 6.

The Network Delay shall be measured as specified in RFC 2679. Packet insertion time (length of a packet in bits / NI bit rate) is part of the network delay, as indicated in ITU-T Rec. I.380, and RFC 2330.

TE manufacturers and Network Providers are encouraged to minimize their contributions to user-to-user delay.

TE delay shall be evaluated using T1.801.01 Class A and B scenes, and measured according to T1.801.04-1997 or ITU-T Rec. P.931.

The allocation of Acceptable One-way Delay Level reflects the consensus on feasible performance with current standard-compliant technology. Alternate decompositions of the Acceptable Level are possible. The Max TE Delay allocation is to accommodate H.261 or H.263 Discrete Cosine Transform video codecs and buffering needed for constant bit rate channels. Alternate coding methods (e.g., Intra-frame only) and use of variable bit rate channels may reduce TE delay, but at the expense of other parameters such as frame rate or channel capacity. Also, TE video codecs may require lower delay when using higher bit rate channels (e.g., 768 kbps or 1536 kbps, that are for further study).

User-to-user delay values in Table 5 shall be used for general business conference planning purposes. However, it is recognized that in some exceptional cases this limit might be exceeded (see ITU-T Rec. G.114). All connections with existing video terminals, including those that employ satellite or digital cellular facilities, or connections having great geographical distance between users, will have significant user-to-user delays. Thus, use of appropriate echo control and media synchronization are essential.

The values for Delay Variation were derived from a view of packet insertion times on an ISDN H0 access at 384kbps and the additional delay variation in actual packet transfer time (Max MTU at 384kbps requires 32 ms insertion time). Out-of-Sequence video packets are deemed lost. When agreeable to all parties, the informative procedures in Appendix B of ITU-T Rec. I.380 may be used to assess delay variation. In this case, the reference packet delay shall be defined such that the maximum number of absolute packet transfer delay measurements fall within the allowable variation.

The measurement interval for Delay Variation shall be 10 seconds.

> NOTE - This interval was chosen to produce a sufficient sample of packets for evaluation, and because of its loose correspondence to speech segment lengths. The delay variation for a set of frames in a measurement interval is calculated as the maximum of the set minus the minimum of the set.

Table 6 - Allocation of Round-Trip Delay for Graphic Teleservices

Graphic User-User Round-Trip Delay	Perceptible Level	100 ms	
	Acceptable Level	200 ms	
	Max TE RT Delay=50ms	Max Network RT Delay = 150 ms	
			Max Network Delay Variation = For Further Study

The Network RT Delay shall be measured as specified in RFC 2681.

The graphic application considered here assumes a form of remote application sharing. One User types characters on his or her local keyboard, but the characters appear on the local display after their representation has traversed the network twice (to the remote application hosted on the other user's TE and back). It is usually preferable for TE to provide a local character display, thus avoiding network delays, and in this case the Perceptible Level (100ms) is a suitable TE objective.

User-to-user delay values in Table 6 shall be used for general business conference planning purposes. However, it is recognized that in some exceptional cases this limit might be exceeded (see ITU-T Rec. G.114). All connections with existing video terminals, including those that employ satellite or digital cellular facilities, or connections having great geographical distance between users, will have significant user-to-user delays.

The acceptable level is derived from the Shur and Morton study (see annex C).

> NOTE - The performance levels in this clause may be difficult to achieve with ITU-T Rec. T.120 application sharing methods, as they are characterized by the remote host echo described above.

5.2   Media Quality as Affected by Transient Impairments

5.2.1   Packet Loss Performance

Packet loss performance is specified at the transport layer (TCP or UDP). Packet Defects at the IP Layer, such as Lost Packets, Errored Packets, and packets whose arrival is beyond the maximum delay or beyond the tolerance range for delay variation, and video packets that arrive out of sequence are mapped to lost packets at the transport layer. IP Packet Loss shall be measured according to RFC 2680. When agreeable to all parties, the definition of errored packets in ITU-T Rec. I.380 may be used in measurements. Arrival sequence may be assessed through examination of RTP packet header information.

This parameter is intended to constrain infrequent randomly distributed packet losses (loss probabilities are Independent). Losses occurring in bursts are constrained by the Availability and Reliability parameters.

5.2.1.1   Video Packet Loss-free Interval

At the user interface (display) the acceptable interval between visible transient impairments (e.g., freeze-frame, sometimes called repeated frame) is 1 hour in a formal Tier 2 group conferencing setting.

Accurate INTRA-mode coded frame reception is essential to the decoding of all INTER-mode coded frames that follow. Assume that 50% of packets contain INTRA-Mode Macroblocks. Then 1 out of 2 lost packets would cause a temporary freeze-frame condition.

Also, periods of scene motion, where the freeze-frame concealment is more easily detected, are infrequent (especially in desktop conferencing). Assume that only 10% of freeze-frame conditions will be detected by users, due to lack of sufficient motion in the displayed video or other concealment techniques. In this case, assuming one noticeable freeze-frame per hour is acceptable in Tier 2 Group conferencing, 20 random packet losses per hour will result in acceptable distortion.

Table 7 - Allocation of Transient Impairments in
Video Streams - Tier 2 Group Conferencing

Video User-User Impairment-free Interval	Perceptible Level	For Further Study	
	Acceptable Level	1 hour (allows 1 defect per hour)	
	Max TE Defects < 1 per 6 hours	Network Defects = Packet Loss at Transport Layer	
		Transport Layer Packet Loss $\leq$ 20 per hour	
		Sum of IP Layer Defects $\leq$ 20 per hour	
		(e.g., Packet Loss, Error, Excessive Delay)	

In the case of Tier 1 desktop conferencing (usually between only two users and in a less-formal setting than group conferences), freeze-frames may occur more frequently than Tier 2. Here, one perceived freeze-frame over the average length PSTN call yields the requirements in table 8.

Table 8 - Allocation of Transient Impairments in Video Streams -Tier 1 Desktop
Conferencing

Video User- User Impairment- free Interval	Perceptible Level	(Not needed for Tier 1)	
	Acceptable Level	3 minutes (allows 1 defect per 3 minutes)	
	Max TE Defects < 1 per 6 hours	Network Defects = Packet Loss at Transport Layer	
		Transport Layer Packet Loss $\leq$ 200 per hour	
		Sum of IP Layer Defects $\leq$ 200 per hour	
		(e.g., Packet Loss, Error, Excessive Delay)	

This is a Defect to total Packet ratio of 0.60%, with assumptions as described in annex B.

5.2.1.2   Audio Packet Loss-free Interval

When an audio channel looses transport stream packets, several outcomes are possible including drop outs (lost speech segments) and audible distortion (clicks, pops, warbling noises, etc.). Some audio decoders attempt to conceal packet losses, and may be quite effective depending on the similarity between the preceding speech segments and lost segment(s).

Note: To make the mapping between packet loss and user interface impairment more clear, this section makes the following assumptions:

1. Packet size (in terms of the amount of media contained in each packet) is in the range of 1 to 5 audio frames of 10ms each. Assume 20ms audio per packet (applies with G.711 and G.728 coders).

2. Packet loss concealment is assumed present in the decoder. Audio concealment can be very effective (therefore, probability of packet loss detection is fairly small).

Table 9 - Allocation of Transient Impairments in Audio Streams

Audio User- User Impairment- free Interval	Perceptible Level	For Further Study	
	Acceptable Level	5 minutes (MMCF basic QOS)	
	Max TE Defects < 1 per 6 hours	Network Defects = Packet Loss at Transport Layer	
		Transport Layer Packet Loss Ratio $\leq$ 1%	
		Sum of IP Layer Defects $\leq$ 1%	
		(e.g., Packet Loss, Error, Excessive Delay)	

The mapping of acceptable level to network defects is derived from the Cox and Perkins (AT&T) study. Additional studies supporting the IP Packet Loss requirement are contributions by Cermak (GTE) and by Thorpe and Avellaneda (NortelNetworks) (see annex C).

5.2.1.3   Data Packet Loss-free Interval

There are two primary causes of transport packet loss when Transmission Control Protocol (TCP) provides reliable byte stream delivery and flow control for Data channels. Packets may be lost in networks when queues overflow, or when packets collide on LANs. Another source of packet

loss is TCP's continuous probing of the connection capacity, increasing it's flow control window by one packet (or segment of the byte stream, in true TCP parlance) until it exceeds available capacity. Packet losses from either source cause the TCP sender to reduce its sending rate (e.g., cutting the window by half), and then to begin to probe again. See T1A1.3/98-015 for background on TCP (annex C).

The main point of this discussion is that packet losses cannot be avoided with typical TCP connections and implementations. Since this complexity exists for TCP Data channels, this standard does not specify a Packet Loss-free Interval in this category. For cases where the Data channel has a Committed Bit Rate, the Delivered Bit Rate will be a sufficient measure of performance. The continuous probing of advanced TCP implementations yields a delivered bit rate of about 75% of the reserved channel capacity, and the Tolerance (used in 5.4) needs to be 50% of the Committed Bit Rate to account for window flow control.

5.3 Committed 1-way Bit Rates on IP network

The Committed Bit Rate is agreed between Network and calling user prior to or during connection establishment.

When a TE requests Committed Bit Rate using an H.323 Admissions Request message (ARQ), overhead bits associated with headers are excluded. (Note that the ARQ message requires the bi-directional capacity, or twice the Committed Bit Rate.) Subsequent (per channel) requests for capacity may need to include the overhead.

Overhead (OH) is approximately 0.3 times payload bit rate.

The network should provide 384kbps * (1+OH) bit rate $\approx$ 500kbps (this is the sum of bit rates for all unidirectional media channels requested by the TE) for Tier 2 Group Conferencing (Room System) Applications.

The network should provide 128kbps * (1+OH) bit rate $\approx$ 168kbps, for Tier 1 Desktop Conferencing. The overhead precludes the use of BRI access for Tier 1 Desktop Conferencing.

The minimum NI bit rate for Tier 1 shall be 384kbps, and 768kbps for Tier 2, when there is best effort and real-time traffic combined.

A sender shall not supply packets at a rate greater than the Committed Bit Rate. A sender should re-negotiate the Committed Bit Rate when they do not desire to supply packets at that rate for a period of time.

5.4 Availability/Reliability Measurements (during Information Transfer)

This subclause specifies availability and reliability in terms of Delivered Bit Rate. The Committed Bit Rate, or a lower rate supplied by the sender, reduced by a Tolerance that accounts for permissible levels of packet defects specified earlier, is the basis for the evaluation of Delivered Bit Rate. This Threshold Bit Rate associates Unavailability with intervals of correlated, or bursty packet losses. Random, or independent packet losses are constrained by the requirements of error-free intervals and packet loss ratios.

If a connection delivers Bit Rate $\geq$ Threshold Bit Rate for one second, then the connection shall be considered in the Available state.

If a connection delivers Bit Rate < Threshold Bit Rate for one second, then there is a failure in connectivity and the connection shall be considered in the Degraded state.

If a connection delivers Bit Rate < Threshold Bit Rate for ten consecutive seconds, then there is a failure in connectivity and the connection shall be considered in the Unavailable (Dropped) state. For a call in progress, this defines the measurement of a Dropped Connection.

Specification of the return from Dropped state isn't necessary in this case, because the session is assumed terminated when the Dropped state is entered, and a new session would need to be established to continue communications.

The following Table illustrates this requirement.

Table 10 - Availability Specification Derived from Delivered Bit Rate

Consecutive Seconds	Delivered Bit Rate	
	>/= Threshold	< Threshold
1		
2		
3	Available state	Degraded state
…		
10 and greater		Unavailable (Dropped)

Delivered Bit Rate shall be measured at 1 second intervals.

> NOTE - Test systems may arbitrarily establish the start of measurement intervals, and in this case independent systems measuring at the same NI may produce slightly different results.

As an example calculation of the Threshold Bit Rate, consider a multimedia connection comprised of one video channel (whose packets have **video_payload** length information fields) and one audio channel (with **audio_payload** information fields). In any specific measurement interval, both an audio and video packet may be lost (adjacent intervals may need to deliver all packets to meet the longer term packet loss requirements).

In this case, the Tolerance = **video_payload** + **audio_payload** and this value in bits will be subtracted from the Committed Bit Rate (or the bit rate supplied) to determine the Threshold Bit Rate.

If a user requests a lesser Committed Bit Rate with a Bandwidth Change Request (BRQ), or accepts a lesser Committed Bit Rate in response to a Gatekeeper's BRQ, then the new Committed Bit Rate will be the basis for evaluating availability. Renegotiated rates should be responsive to the considerations of 5.3.

5.5   Connection Set-Up Time

Connection set-up time for all requested media should be less than 30 seconds. This interval is allocated as 10 seconds maximum for each network connection and the remainder divided equally between calling and called terminals.

5.6   Connection Accessibility

The guideline for connection accessibility is 99% successful connections (with Committed bandwidth requests/reservations consistent with the Tier, as in 5.3). This parameter applies to the extent of services agreed between User and End-End Network Provider and is measured at the Network Interface. (For example, this would exclude cases of called terminal busy).

5.7   Connection Reliability (frequency of Dropped Connection, or Call Cut-off)

The guideline is that 99.9% of reference calls are sustained to completion (where a reference call is 10 minutes in length). Note that all resources, TE, LAN, and Networks between NIs contribute to this performance. The guideline is not allocated among contributors here.

5.8 Media Synchronization

For this parameter, it is possible to specify both the perceptible and acceptable quality threshold levels.

Table 11 - Audio - Video Synchronization

User-User A/V Sync Tolerance	Perceptible Level		+20ms to -40ms	
	Acceptable Level	+80ms to -80ms		
Max A/V Skew		TE +80ms to -80ms		Network Not specified

The Perceptible level is derived from the traditional requirement for broadcast television synchronization. The Acceptable level is derived from the Steinmetz study (see annex C), where less than 40% of subjects detected error for experiments with T1.801.01 Class A and B composition.

Table 12 - Audio - Cursor Motion Synchronization

User-User A/Cursor Sync Tolerance	Perceptible Level	For Further Study
	Acceptable Level	+750ms to -500ms
Max A/Cursor Skew		TE +750ms to -500ms	Network Not specified

The Acceptable level is derived from the Steinmetz study (see annex C), where less than 20% of subjects detected error for experiments with T1.801.01 Class D composition.

Annex A
(informative)

Notes on Network-TE Interactions during H.323 Call Set-up with RSVP

This annex illuminates the topic of requests for network capacity in calls between H.323 TE. There are two types of requests, one for bi-directional aggregate audio and video capacity and another request for unidirectional channel capacity for each individual media stream.

Assuming Gatekeeper Routed Signaling, the following information will be exchanged on the H.225.0 Registration, Admission, and Status (RAS) channel (Section references are for H.323 02/98 unless otherwise specified):

1. Endpoint (or TE) registers with an assigned or discovered Gatekeeper.

2. Endpoint sends an Admissions Request (ARQ) specifying the Call Bandwidth (**bandWidth** field). From 7.2.4 " This is an upper limit on the aggregate bit rate for all transmitted and received audio and video channels excluding any RTP headers, RTP payload headers, network headers, and other overhead. Data and control channels are not included in this limit. The Gatekeeper may reduce the requested Call Bandwidth in the Admissions Confirm (ACF) message. An endpoint shall assure that the aggregate bit rate, averaged over one second, for all transmitted and received, audio and video channels is at or below the Call Bandwidth. An endpoint or the Gatekeeper may attempt to modify the Call Bandwidth during a call using the Bandwidth Change Request (BRQ) message." Note that Call Bandwidth is a bi-directional specification (e.g., an endpoint desiring 128kbps in each direction must request 256kbps). Also, an endpoint indicates its ability to reserve resources in the ARQ **TransportQOS** field (Appendix II.2).

3. Call Channel and H.245 Control Channel Signaling follow.

When Call Signaling and Capabilities Exchange are complete, an endpoint "A" may begin to send media streams and must open unidirectional H.245 logical channels (for each stream) to the other endpoint "B".

a)   From Appendix II.3, "RSVP signaling for resource reservation should be a part of the opening logical channel procedure. Endpoint A would cause RSVP *Path* messages to be sent out to B. These *Path* messages go through routers and leave "state" on their way tracing towards B. *Path* messages contain the complete source and destination addresses of the stream and a characterization of the traffic that the source will send." RFC 2210 describes the usage and contents of the SENDER_TSPEC, which characterizes the traffic endpoint A intends to send. SENDER_TSPEC includes a peak data rate and token bucket rate and size.

b)   From Appendix II.3, "Endpoint B would use the information from the *Path* to make the RSVP *Resv* request for the full length of the path. *Resv* messages contain the actual reservation and will generally be the same as the traffic specification in the *Path* message." RFC 2210 describes the usage and contents of the FLOWSPEC, which characterizes the traffic endpoint B intends to reserve.

c)   Appendix II.5 describes the **OpenLogicalChannel** and RSVP procedures in detail. When endpoint B receives the RSVP *ResvConf* message, its reservation is confirmed.

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Annex B
(informative)

H.323 Packet size assumptions

H.323 Packet Loss Work Sheet

Payload of a Video Packet in H.323 typical use	1500.00	bytes/packet
	12000.00	bits/packet
Aggregate Info Rate of H.323 Desktop Conference	128000.00	bits/s
Rate for audio codec	16000.00	bits/s
Remaining rate for video codec	112000.00	bits/s
Packet Rate for video	9.33	Packets/second
	560.00	Packets/minute
	33600.00	Packets/hour
IF the acceptable Error-Free Interval is	0.33	Defect/minute
AND the Prob of Defect Detection is	0.10	
THEN the allowable Packet Defect Rate is	3.33	Defect/minute
	200.00	Defect/hour
This is a Packet Defect Ratio of	0.60%	Defects/Packet

Annex C
(informative)

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