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Title: H.323 on ATM AAL5

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1. Introduction

The International Telecommunications Union - Telecommunications Standardization Sector (ITU-T) produces recommendations for building interoperable audiovisual terminal systems and equipment. One of these recommendations is H.323, which is titled Visual Telephone Systems and Equipment for Local Area Networks Which Provide a Non-guaranteed Quality of Service.

The purpose of this paper is to describe the details necessary to implement H.323 on Asynchronous Transfer Mode (ATM) using ATM Adaptation Layer Five (AAL5) as described in ITU-T I.363. The characteristics of ATM are such that ATM virtual channels can operate in modes that provide guaranteed or non-guaranteed Quality of Service. This paper will describe the methods needed to build an interoperable H.323 implementation that uses ATM AAL5 as the network protocols. Although the use of a guaranteed quality of service network is not specified in H.323, it should be obvious to one familiar with ATM how this would be accommodated.

2. Problem Statement

The ITU has specified H.310 and H.321 as the audiovisual terminals for use with ATM. The main drawback with implementation of these recommendations is that they do not fully encompass the use of ATM AAL5. ATM AAL5 is being provided for use on most ATM network interface cards for personal computers. It is desired to have a audiovisual terminal specification that will be interoperable on ATM AAL5.

The ITU specifications for H.310 and H.321 both call for the use of a multiplex for the packaging of the audio and video streams being placed on the network. They both allow the use of H.221, a circuit based multiplex designed for use on ISDN Bearer channels. H.310 also allows the use of H.222, which is the MPEG transport stream. It is desirable to use separate transmission virtual circuits for audio and video. This is the method described in H.225.0, the media stream packetization protocol designed for use with H.323. The overall architecture of H.323 is very desirable for network based A/V communications. The use of ATM AAL5 is an example of network based communications. It is therefore desired to specify how H.323 can be used on ATM AAL5.

Currently H.323 does not specifically accommodate the use of ATM. This is because it was written to accommodate only networks that provide a non-guaranteed quality of service. One could specify the use of H.323 on IP over ATM, but this would be less efficient than using AAL5 directly for the transport of the audio and video streams of H.323. This paper specifies the solution to taking this approach.

2.1. H.323 Interoperability on ATM-AAL5

Interoperability can be specified by ensuring that the endpoint can create a connection to a peer terminal and then use that connection to conduct an audiovisual conference. In the case of specifying interoperability over ATM AAL5, this paper will specify the methods that are either not specified, or are not plainly evident to one reading the H.323 specification.

3. Overview of H.323

An example of an H.323 terminal is shown in Figure 1 below. The diagram shows the user equipment interfaces, video codec, audio codec, telematic equipment, H.225.0 layer, system control functions, and the interface to the LAN. All H.323 terminals have a System Control Unit, H.225.0 layer, Network Interface, and an Audio Codec Unit. The Video Codec Unit and User Data Applications are optional.

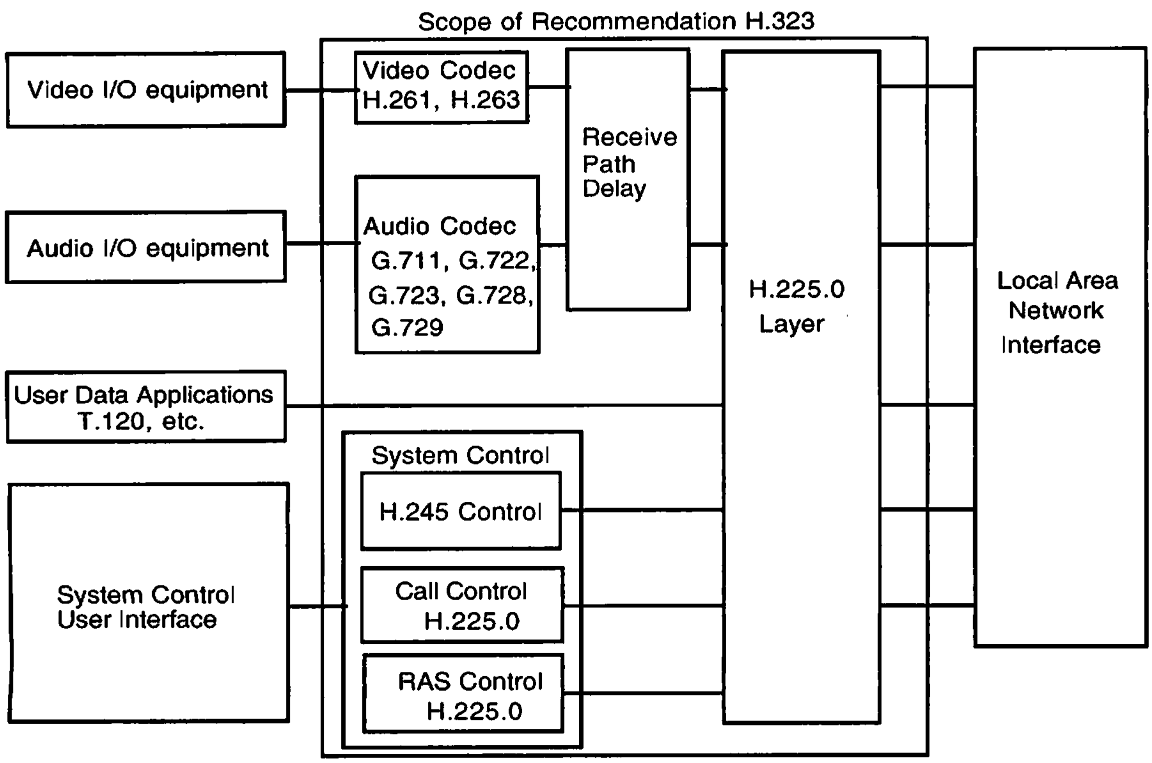


Figure 1 - H.323 Terminal Equipment

Only those items that are in the dotted box are in the scope of the H.323 recommendation. The network interface is not specified by H.323 but the following characteristics are given in H.323:

> The LAN interface shall provide the services described in Recommendation H.225.0. This includes the following: Reliable end-to-end service is mandatory for the H.245 Control Channel, the Data Channels, and the Call Signaling Channel. Unreliable end-to end service is mandatory for the Audio Channels, the Video Channels, and the RAS Channel. These services may be duplex or simplex, unicast or multicast depending on the application, the capabilities of the terminals, and the configuration of the LAN.

3.1. H.225.0

The general scope of ITU-T H.225.0 is shown in below. H.225.0 generally deals with the issues of Call Signaling, Terminal to Gateway Signaling, and the use of RTP/RTCP.

ITU-T H.225.0 describes the means by which audio, video, data, and control are associated, coded, and packetized for transport between H.323 terminals or gateways on a non-guaranteed quality of service LAN. H.225.0 is intended to operate over a variety of different LANs. Thus, H.225.0 is defined as being above the transport layer. The scope of H.225.0 communication is between H.323 terminals and H.323 gateways on the same LAN, using the same transport protocol. This LAN may be a single segment or ring, or it logically could be an enterprise data network comprising multiple LANs bridged or routed to create one interconnected network.

H.225.0 makes use of RTP/RTCP(Real-time Transport Protocol/Real-Time Transport Control Protocol) for media stream packetization and synchronization for all underlying LANs. The usage of RTP/RTCP as specified in H.225.0 is not tied in any way to the usage of TCP/IP/UDP. H.225.0 assumes a call model where initial signaling on a non-RTP transport address is used for call establishment and capability negotiation (see H.323 and H.245), followed by the establishment of one or more RTP/RTCP connections. H.225.0 contains details on the usage of RTP/RTCP.

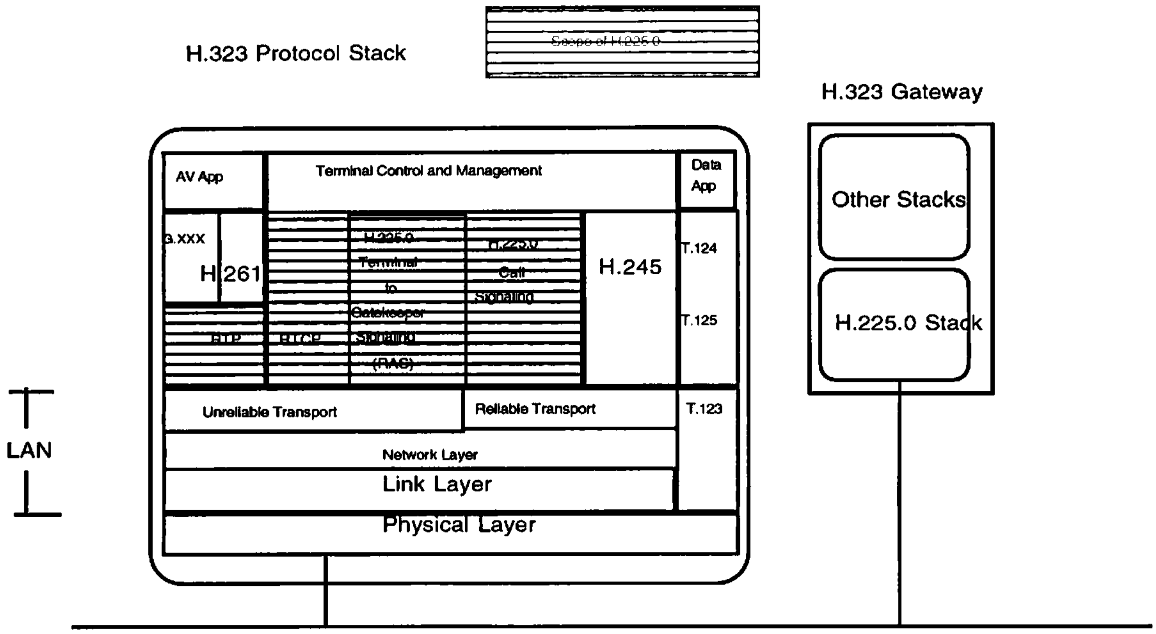


Figure 2 - Scope of H.225.0

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

3.2. H.245

ITU-T H.245 is used in H.323 for conference control after initial call setup of ITU H.225.0. H.245 specifies syntax and semantics of terminal information messages as well as procedures to use them for in-band negotiation at the start of, or during communication. The messages cover receiving and transmitting capabilities as well as mode preference from the receiving end, logical channel signaling, and Control & Indication. Acknowledged signaling procedures are specified to ensure reliable audiovisual and data communication.

H.245 is used to establish data, audio, and video logical channels during the H.323 conference.

3.3. RTP/RTCP

The Real-Time Transport Protocol (RTP) and RTP Control Protocol (RTCP) are based on an IETF draft. It is contained in IETF RFC 1889. The pertinent parts of RTP and RTCP are repeated in an appendix of H.225.0. RTP is basically a data stream header that can provide the information to reconstruct a real-time stream, such as an audio or video stream. RTCP is used to allow the application to monitor network conditions and possibly adjust for these changes.

3.3.1. RTP Overview

To better understand the function of RTP in H.323 we can adapt some of the text from the H.225.0 recommendation:

> RTP provides end-to-end delivery services for data with real-time characteristics, such as interactive audio and video. Those services include payload type identification, sequence numbering, timestamping and delivery monitoring. Applications typically run RTP on top of UDP to make use of its multiplexing and checksum services; both protocols contribute parts of the transport protocol functionality. However, RTP may be used with other suitable underlying network or transport protocols. RTP supports data transfer to multiple destinations using multicast distribution if provided by the underlying network.
>
> RTP itself does not provide any mechanism to ensure timely delivery or provide other quality-of-service guarantees, but relies on lower-layer services to do so. It does not guarantee delivery or prevent out-of-order delivery, nor does it assume that the

underlying network provides reliability or in-order delivery of packets. The sequence numbers included in RTP allow the receiver to reconstruct the sender's packet sequence, but sequence numbers might also be used to determine the proper location of a packet, for example in video decoding, without necessarily decoding packets in sequence.

While RTP is primarily designed to satisfy the needs of multi-participant multimedia conferences, it is not limited to that particular application.

3.3.2. RTCP Overview

The RTP control protocol (RTCP) used in H.323 allows the monitoring of the quality of service of the underlying network. If both audio and video media are used in a conference, they are transmitted as separate RTP sessions, RTCP packets are transmitted for each medium using two different UDP port pairs and/or multicast addresses. There is no direct coupling at the RTP level between the audio and video sessions, except that a user participating in both sessions should use the same distinguished (canonical) name in the RTCP packets for both so that the sessions can be associated. H.245 does provide a specific association field in the protocol for associating two related streams together when operating in H.323.

One motivation for this separation is to allow some participants in the conference to receive only one medium if they choose. Despite the separation, synchronized playback of a source's audio and video can be achieved using timing information carried in the RTCP packets for both sessions.

4. Overview of ATM-AAL5

ATM moves information at the ATM layer in cells that are 53 bytes in length. Five of those bytes are for the ATM cell header, and 48 bytes are the ATM service data unit (ATM-SDU). An ATM Adaptation Layer (AAL) enhances the service provided by the ATM layer to support functions required by the next higher layer. The AAL performs specific functions that are required by the user, control and management planes and supports the mapping between the ATM layer and the next higher layer. The AAL receives information from the ATM layer in one or more 48 byte ATM-SDU. It also passes information to the ATM layer in the form of a 48 byte ATM-SDU. An AAL uses a specific combination of a segmentation and reassembly (SAR) sublayer and convergence sublayer (CS) to provide different service access points (SAPs) to the layer above the AAL.

AAL5 has the CS subdivided into two parts, the Common Part CS (CPCS) and the Service Specific CS (SSCS) as shown in Figure 3 below.

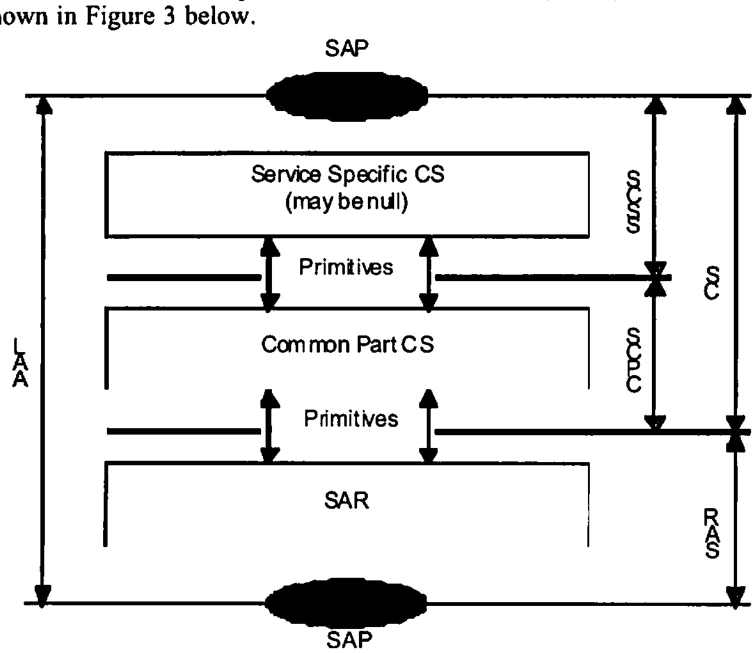


Figure 3 - Structure of AAL5

AAL5 provides the capabilities to transfer the AAL-SDU from one AAL user to another AAL user through the ATM network. The Service Access Point (SAP) above the CS is known as the AAL-SAP.

The SAP below the SAR is called the ATM-SAP. There is a one-to-one relationship between an AAL-SAP and an ATM-SAP on a per-virtual circuit basis.

AAL5 provides the capabilities to transfer the AAL-SDU from one AAL-SAP to one other AAL-SAP through the ATM network in a point-to-point connection. AAL5 users will have the capability to select a given AAL-SAP associated with the QOS required to transport the AAL-SDU (for example, delay and loss sensitive QOS). This is shown in section a of Figure 4 below.

AAL5 in non-assured operation provides the capability to transfer the AAL-SDUs from one AAL-SAP to more than one AAL-SAP through the ATM network in point-to-multipoint connections. This is shown in section b of Figure 4 below.

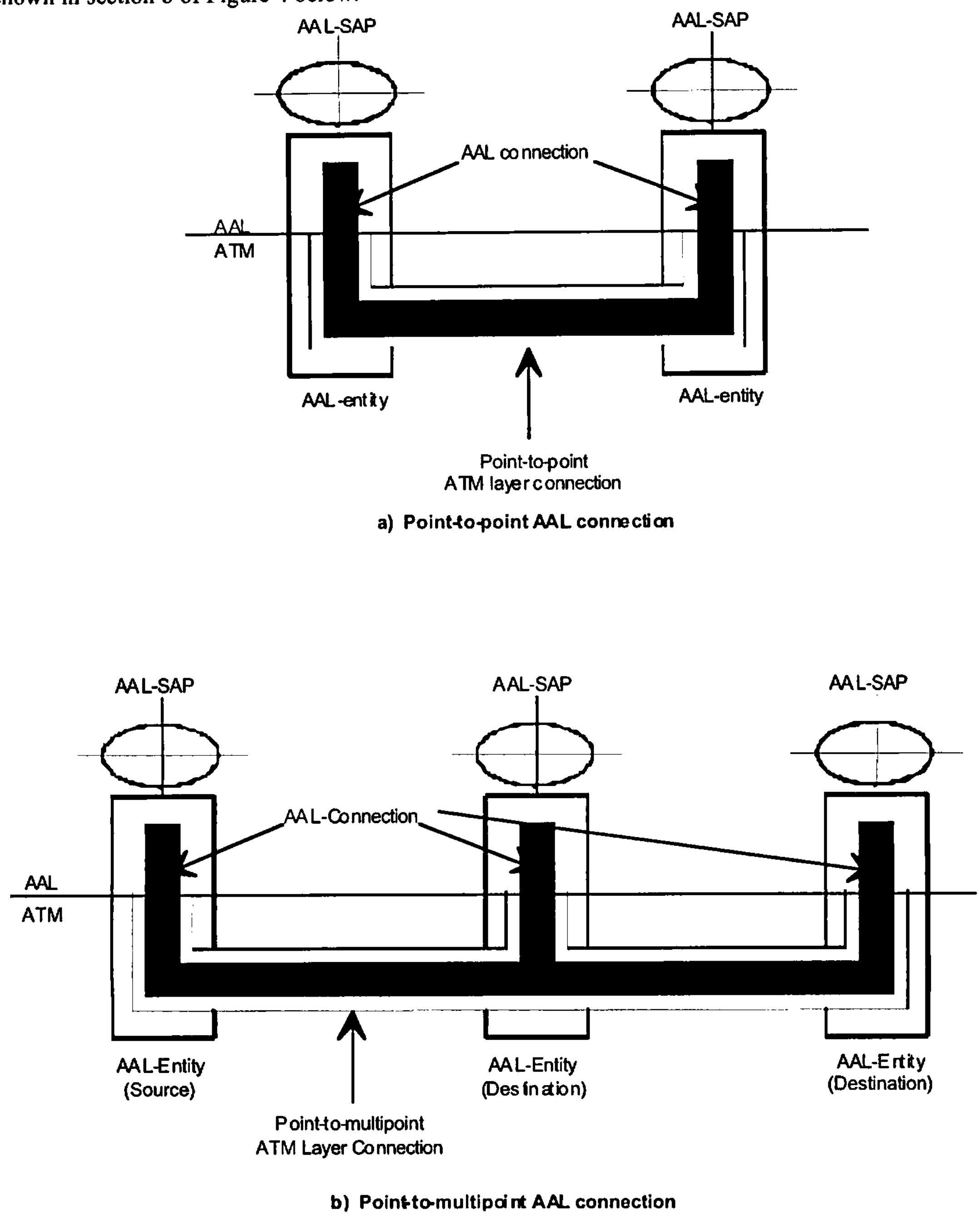


Figure 4 - AAL5 Connection types.

The AAL type 5 makes use of the service provided by the underlying ATM layer. Multiple AAL connections may be associated with a single ATM layer connection, allowing multiplexing at the

AAL; however, if multiplexing is used in the AAL, it occurs in the SSCS. The AAL user selects the QOS provided by the AAL through the choice of the AAL-SAP used for data transfer.

4.1. Mode Support

Two modes of service are defined for AAL5 in ITU-T I.363. These are Message Mode and Streaming Mode. The ATM Forum UNI 3.1 only specifies support for message mode. Most common ATM network interface cards (NICs) only provide Message mode in their AAL5 support. Therefore we will only address the use of message mode on AAL5 in this paper.

4.1.1. Message Mode Description

The characteristics of Message Mode service is given below as adapted from ITU-T I.363:

> *Message Mode service* – The AAL Service Data Unit is passed across the AAL interface in exactly one AAL Interface Data Unit (AAL-IDU). This service provides the transport of fixed size or variable length AAL-SDUs.
>
> i) In case of small fixed size AAL-SDUs, an internal blocking/deblocking function in the SSCS may be applied; it provides the transport of one or more fixed size AAL-SDUs in one SSCS-PDU.
>
> ii) In case of variable length AAL-SDUs, an internal AAL-SDU message segmentation/reassembling function in the SSCS may be applied. In this case a single AAL-SDU is transferred in *one or more* SSCS-PDUs.
>
> iii) Where the above options are not used, a single AAL-SDU is transferred in one SSCS-PDU. When the SSCS is null, the AAL-SDU is mapped to one CPCS-SDU.

4.2. Data Rate Support (UBR, CBR, VBR)

An ATM network simultaneously transports a wide variety of network traffic -- voice, data, image, and video. It provides users with a guaranteed quality of service (QOS) on a per Virtual Channel (VC) basis. ATM provides for three levels of QOS for bearer traffic delivery of ATM-SDUs on a VC. These are Constant Bit Rate (CBR), Variable Bit Rate (VBR), and Unspecified Bit Rate (UBR). The QOS for a VC is assigned at the creation of the VC.

All ATM equipment supports UBR. This is effectively best effort delivery, not providing any true QOS. Nearly all ATM equipment supports CBR. This requires the specification of a peak bandwidth (PB) that will be used on the VC. Any data sent at a rate greater than the PB is not guaranteed to be delivered. Any buffering that is required for data rate adjustment must be done at the endpoints of the ATM SAP. Some ATM equipment supports VBR. It allows the rate of delivery at the ATM SAP to vary.

Only non-assured services (no QOS) are supported on point-to-multipoint ATM connections. Therefore the use of point-to-multipoint is left for further study.

4.3. IP on ATM

Initial uses of ATM networks in the enterprise are for traditional LAN traffic. To aid in the rapid adoption of ATM, it was desired to allow the ATM network to be used for IP traffic. To this end two proposals have gained popularity. These are firstly LAN Emulation (LANE), proposed by the ATM Forum, and secondly Classical IP over ATM, proposed by the IETF. It is unfortunate that the two methods cannot communicate directly with each other on a shared IP segment. There must be a bridging router to enable two endpoints, using different IP on ATM approaches, to communicate with each other.

4.3.1. LAN Emulation - ATM Forum

LAN Emulation allows traditional LAN protocols, including IP based protocols, e.g. IPX, SPX, TCP, UDP, etc., to be used on ATM. It actually emulates an Ethernet LAN to the software on the computer using the network. It effectively hides the fact that ATM is being used for the network traffic. A serious drawback to this method is that it is done in such a way that the LAN traffic cannot use the QOS capabilities of the ATM layer. Since the LAN traffic cannot embrace QOS, it cannot run RSVP on the IP routers.

4.3.2. Classical IP over ATM - IETF

Classical IP over ATM is designed to allow IP protocols, e.g. TCP,UDP, to be used on ATM. This method shares the serious drawback with LANE, that it is done in such a way that the LAN traffic cannot use the QOS capabilities of the ATM layer. It allows use of a classical IP router infrastructure to mange traffic on the LAN, but does not use QOS in the ATM layer. It also cannot embrace RSVP. ATM VC's are used at the lowest layer to move data between endpoints and routers.

5. Architecture

The basic protocol architecture of the system is shown in Figure 5 below. It uses IP on ATM for reliable delivery of the H.225.0 and H.245 messages. It uses RTP/RTCP on AAL5 directly for the audio and video streams.

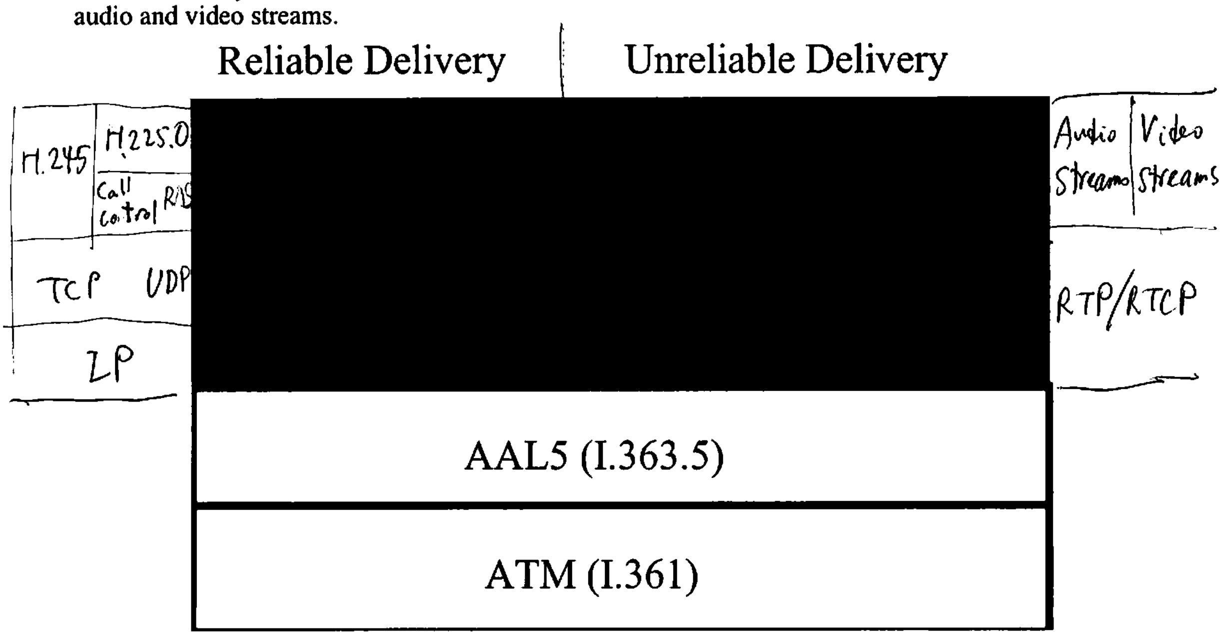


Figure 5 - Architecture for H.323 on ATM-AAL5

5.1. Overview of System

The system architecture is designed to make use of H.323, and its component protocols, as they are specified. It is further designed to use commonly available services of ATM-AAL5 on currently available ATM equipment.

5.2. H.225.0 and Q.931 on TCP/IP on ATM-AAL5

The H.225.0 and Q.931 is used on TCP/IP on ATM-AAL5 using one of the two IP on ATM methods mentioned above. The endpoint should listen on the well known TCP ports identified in H.225.0. If the endpoint is being used on a LAN with a Gatekeeper, the endpoint should use the methods described in H.225.0 to discover and register with the Gatekeeper. This requires the support of UDP multicast. If multicast is not available on the network, the endpoint will have to be pre-configured with the Gatekeeper(s) address(es).

The methods outlined in H.225.0, combined with an ATM over IP method, should be used to establish the H.245 control channel on TCP/IP on ATM-AAL5. The H.245 control channel will be used to setup all additional logical channels for the media streams to be used in the H.323 conference.

It should be noted that endpoints on a shared IP segment must be using the same IP on ATM methods in order to receive call notifications on the well known ports. If two endpoints are using different IP on ATM methods, they must use a bridging IP router to deliver the message.

If the ATM NIC has support for both IP on ATM methods, the preferred method shall be Classical IP over ATM as specified in IETF RFC 1577 because of its efficient handling of IP data.

5.3. H.245 on TCP/IP on ATM-AAL5

Once the reliable H.245 control channel has been established using methods described in H.225.0, additional channels for audio, video, and data are established based on the outcome of the capability exchange using H.245 open logical channel procedures.

H.323 has the capability for the audio and video streams to be established on a different machine address that the H.245 control channels. This is fortunate since the TCP/IP channels are established on the machines IP address, and the audio and video are to be sent on RTP/RTCP on AAL5 directly. The following text is adapted from H.225.0:

> Although the transport address for, say, audio and video, may share the same LAN address and differ only by TSAP identifier, some manufacturers may choose to use different LAN addresses for audio and video. The only requirement is that the convention of Annex A/B should be followed in the numbering of TSAP identifiers in the RTP session.

For the purposes of implementing H.323 on ATM-AAL5, we will use this capability to effectively establish the A/V streams separate from the H.245 control channel.

5.3.1. Addressing for A/V Streams

H.245 specifies the fields for the address PDU's to contain an machine address and a port number. According to procedures outlined in H.225.0, for each logical channel opened for an audio or video stream, there is a corresponding RTP/RTCP port pair. The addressing is outlined in the sections that follow.

5.3.1.1.ATM Machine Address

The ATM machine address should be given in the **H2250LogicalChannelParameters** of the H.245 **OpenLogicalChannel**. The **mediaChannel** field should be filled with the 20byte using the NSAP-style ATM End System Address BCD encoding, as defined in the ATM Forum UNI 3.1. The actual H.245 PDU format shown in H.225.0 should be used, as the NSAP field was inadvertently left out of the current **H245ncm6.doc** version of the H.245 recommendation. The RTP and the RTCP virtual circuits are opened as a single logical channel as described in H.225.0 using the same NSAP machine address. The format is specified in the protocol section below.

The use of E.164 for the address is not currently specified in H.245 or H.323/H.225.0. Therefore, its use is for further study.

5.3.1.2.BHLI Specification

The Broadband High Layer Information is specified in the **portNumber** field of the **OpenLogicalChannel** message as described in H.245. The definition in H.225.0 should be clearly defined such that the nature of the ATM addressing in this way is understood to be supported. It is sufficient to use a user specified BHLI that contains the RTP port to be used for the specific audio or video stream, even though this is not required. The format of the BHLI primitive for ATM Forum UNI 3.1 is specified in the protocol section below.

5.3.1.3.BLLI Specification

The Broadband Lower Layer Information field is not used. This must be signaled by setting the appropriate fields in the ATMF UNI 3.1 primitive to the value **SAP_FIELD_ABSENT**. The format for specifying this is shown in the protocol section below.

5.3.2. Capability extensions for ATM Services

The H.245 capabilities exchange for H.225.0 will need to be extended to fully encompass ATM QOS support and relevant network capabilities. This exercise is left for further study in a future revision of this document.

5.3.3. AAL5 Information Elements

The following text should be used to set the AAL5 parameters, specifying the information elements used for H.323.

```
AAL_TYPE =  AALTYPE_5
ForwardMaxCPCSSDUSize = 1536
```

```
BackwardMaxCPCSSDUSize = 1536
AAL5_MODE  = AAL5_MODE_MESSAGE  (only used for UNI 3.0)
AAL5_SSCS = AAL5_SSCS_NULL
```

The format for specifying this in ATMF UNI 3.1 is shown in the AAL5 Information Elements protocol section below.

5.4. A/V-RTP on AAL5 Message Mode

The ATM virtual circuits connections are established prior to sending any data on them. The connection establishment is triggered by servicing the **OpenLogicalChannel** primitive in H.245. A channel pair will be opened for each direction. This requires the H.323 terminal to use a different RTP/RTCP port ID pair for each direction. This is further explained below. The audio and video streams are then sent to the AAL5 SAP. The size of the Network Transmission Unit (NTU) must be known on both sides. The current suggested size in H.323 of maximum of 1536 bytes should be used.

5.4.1. Unidirectional VC's

ITU-T H.245 usage in H.323 has no current required method to allow the use of the backward channel of bi-directional VC's. Therefore the usage of only the forward channel is currently specified. The use of both directions of a ATM VC is for further study.

5.4.2. NTU Size

The suggested NTU size in H.323 is maximum of 1536 bytes as described above. The maximum NTU for ATM AAL5 is 64Kilobytes. The suggested size should only be changed if it is known to be more efficient network usage. The NTU size is specified in the capabilities exchange in H.245 setup.

5.4.3. RTCP on AAL5

RTCP is used on AAL5 in the same manner that the audio and video streams are supported, except that a QOS ATM VC is not necessary for the RTCP stream, even if one is used for the A/V stream. The RTCP messages should conform to the usage outlined in H.225.0 for RTCP. The RTCP streams are used to monitor and report on network conditions, so that possible adjustments can be made to support efficient bandwidth usage.

5.4.4. RTP/RTCP Port ID Numbering

RTP allows only a single RTP session per transport port ID. We must use ATM bi-directional channels in a unidirectional usage for the RTP media streams on ATM. Therefore, each side of the H.323 connection must use a different port ID pair for the RTP/RTCP logical channels. RTP states that a single session cannot span different ports at an endpoint. This means that each side of a point-to-point connection must use a different H.245 **SessionId** in the H.245 **OpenLogicalChannel** request for the RTP/RTCP channels. This is a departure from the H.323 convention of both sides using the **SessionId's** of 1 and 2 for audio and video respectively. The H.323 endpoint will specify the RTP session ID based on the Mater/Slave procedures outlined in H.245. The Master endpoint will use ports 1and 2 as conventional. The Slave endpoint will use the ports ID of the zero when sending OpenLogicalChannel requests to the Master. The Master will then specify a new session id that it will associate with the Slave's RTP media channel in the **OpenLogicalChannelAck.** The following table illustrates the proper port numbering.

Terminal	RTP/RTCP Port ID pair numbers.
Master	Conventional Numbering (1 and 2)
Slave	Received from Master in **OpenLogicalChannelAck.**

Both endpoints must locally do any necessary implementations to

5.5. QOS Considerations (Optional)

ITU-T H.323 states support only for non-QOS LANs. Therefore the direct use of QOS in the underlying network is outside the scope of the recommendation. This means there is no way to signal the desired use of QOS channels in the in-band messaging of H.323. This paper will show how QOS-based ATM virtual circuits can be used for the audio and video streams of the H.323 connection.

5.5.1. Unidirectional VC's

ITU-T H.245 usage in H.323 has no current required method to allow the use of the backward channel of bi-directional VC's. Therefore the usage of only the forward channel is currently specified. The use of both directions of a ATM VC is for further study.

5.5.2. Opening of Virtual Circuits

Support for QOS in the ATM virtual circuit is indicated at the time the virtual circuit is opened. If successful, the ATM network provides a guaranteed QOS for the lifetime of the opened virtual circuit. The actual level of QOS is determined by the data rate support, as explained in the overview of ATM AAL5 above. The most likely available candidate for QOS support for A/V streams is the use of a CBR channel. This is opened by using the proper Q.2931 Information Elements (IE) and the proper ATM Traffic Descriptor IE. The format for the proper primitives is given in the ATM QOS Support (Optional) protocol section below.

5.5.3. Use of CBR

The most likely available usage of QOS circuits is to use Constant Bit Rate (CBR) ATM VC's for the audio and video RTP streams flowing on AAL5. The use of CBR can be accomplished by using the proper **TrafficType** field in the ATM Broadband Bearer Capability IE. The primitive is given in the QOS protocol section below.

5.5.4. Setting the Proper Bandwidth

When a QOS ATM VC is used, it is important to set the proper parameters in the ATM Traffic Descriptor for the forward VC. The **PeakCellRate** and **SustainableCellRate** parameters can be derived from the capabilities exchange parameters. For video, the **maxBitRate** field can be used from the H261VideoCapability or the H263VideoCapability to determine the ATM Cell rate. For audio, the audio capability chosen implies the bit rate to be used. For example, the use of **g711Ulaw64k** suggests the use of a 64 Kilobit per second audio channel. Another example is that the use of **g728** indicates the use of a 16 Kilobyte channel.

To convert bit rates to Cell rates, take the bit rate and divide it by 384(8x48).

6. Protocol Section

6.1. ATM UNI 3.1 Usage

6.1.1. ATM Address

The ATM address is **ATM_NSAP** in a 20 bytes format in the NSAP-style ATM Endsystem Address scheme. Using ATM Forum UNI 3.1 the address will be specified as follows:

Field Name	Value	Meaning in ATMF UNI 3.1
AddressType	ATM_NSAP	NSAP-style ATM Endsystem Address scheme.
NumofDigits	any	field is ignored for ATM_NSAP case.
Addr	ATM Addr	BCD coding of NSAP-style AESA as in ATMF UNI 3.1.

Use of E.164 addresses is currently not supported.

6.1.2. ATM BHLI

The BHLI is user specified and contains the RTP port identification number. Using ATM Forum UNI 3.1 the primitive will be as follows:

Field Name	Value	Meaning in ATMF UNI 3.1
HighLayerInfoType	BHLI_UserSpecific	User Specified BHLI.
HighLayerInfoLength	0x08	Next field is 8 bytes long.
HighLayerInfo	RTP Port	Signals RTP port to be used for current Logical Channel.

6.1.3. ATM BLLI

The BLLI is not used. This is specified by setting the appropriate fields in the BLLI primitive to the value to indicate **SAP_FIELD_ABSENT**. Using ATM Forum UNI 3.1 the primitive will be as follows:

Field Name	Value	Meaning in ATMF UNI 3.1
Layer2Protocol	SAP_FIELD_ABSENT	BLLI Layer 2 is not used.
Layer2UserSpecifiedProtocol	any	field not used.
Layer3Protocol	SAP_FIELD_ABSENT	BLLI Layer 3 is not used.
Layer3UserSpecifiedProtocol	any	field is not used.
Layer3IPI	any	field is not used.
SnapID	any	field is not used.

Unused fields should be set to all zeros.

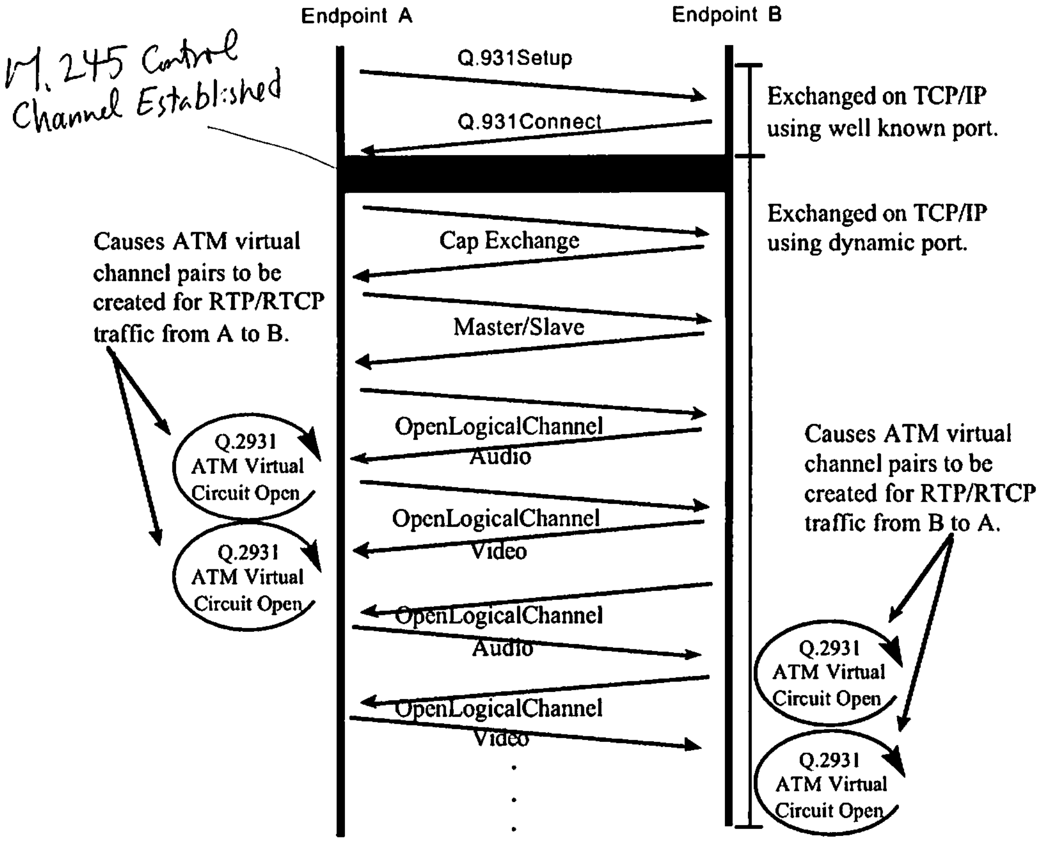
6.1.4. AAL5 Information Elements

The AAL5 parameters information elements are described in Section 5.3.3 -AAL5 Information Elements. Using ATM Forum UNI 3.1 the primitive to specify the AAL_PARAMETERS_IE will be as follows:

Field Name	Value	Meaning in ATMF UNI 3.1
AALType	AALTYPE_5	AAL5 is to be used.
ForwardMaxCPCSSDUSize	1536	Max NTU is 1536 decimal.
BackwardMaxCPCSSDUSize	1536	Max NTU is 1536 decimal.
Mode	any	field is not used.
SSCSType	AAL5_SSCS_NULL	SSCS layer is a null layer.

6.2. Q.931/H.245 Usage

The establishment of a H.323 call using ATM AAL5 is done in a manner similar to using H.323 totally on IP. The difference is that the completed **OpenLogicalChannel** exchange in H.245 should result in a virtual circuit pair being established on the ATM AAL5 connection. This means that the H.245 acknowledged **OpenLogicalChannel** should cause a explicit connection to be made at network layer. This is illustrated in below.



Endpoint A calling Endpoint B

Figure 6 - H.323 call establishment showing ATM effect.

Those that are familiar with H.323 will see that this in no way departs from expect behavior of H.323 call establishment. The addition of an explicit connection establishment, is different from the use of UDP/IP which is a connectionless model, requiring no explicit connection to be established.

6.3. RTP Usage

Usage of RTP on H.323 is outlined in Annex a of H.225.0. The use of RTCP is currently required for all H.323 connections, and therefore is required even when using a CBR virtual circuit.

6.4. ATM QOS Support (Optional)

The following protocol should be used if the audio and video RTP streams are going to flow on QOS based ATM VC's.

6.4.1. Q.2931 Information Elements for QOS Support

If the RTP streams are to be established on a QOS ATM VC, the following Q.2931 IE structure must be used in the list of concatenated Q.2931 IE structures. If no QOS is to be used the structure is not included. The format for the ATMF UNI 3.1 primitive for the Q.2931 IE is as follows:

Field Name	Value	Meaning in ATMF UNI 3.1
IEType	IE_QOSClass	QOS is to be used.
IELength	size of PDU	Size of all fields of the Q.2931 IE.
IE (array)	AAL5 IE	as listed above.

6.4.2. ATM Broadband Bearer Capability Information Element

If the RTP streams are to be established on a QOS ATM VC, the TrafficType in the ATM Broadband Bearer Capability IE structure must be used to indicate the QOS desired. The possible values are shown below:

TrafficType	Meaning in ATMF UNI 3.1
TT NOIND	No indication of traffic type (no QOS or 'best effort').
TT CBR	A CBR QOS channel is to be used.
TT VBR	A VBR QOS channel is to be used.

The use of VBR ATM VC's are left for further study.

7. Windows specific parameters.

7.1. Winsock 2 (WS2) Usage

Implementers can refer to Winsock 2 ATM Specification, and the ATM Annex in particular, for details on using ATM in a Windows environment. Using the parameter information above, it should be clear the proper parameters for WS2 channel establishment. The use of WS2 can simplify the implementation of the ATM connection establishment. A call to **WSAConnect()/WSPConnect()** for point-to-point connections, or **WSAJoinLeaf()/WSPJoinLeaf()** for point-to-multipoint connections, is all that is required.