

IP MultiMedia over Satellite (IMMSAT) - Study Report



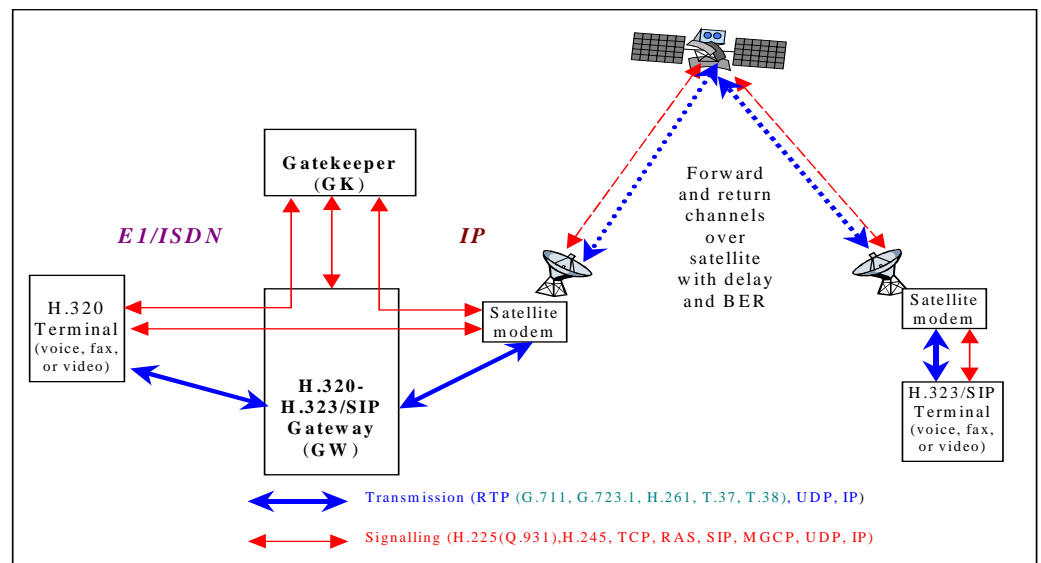
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ANVENDT DATAFORSKNING

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Norwegian Computing Center/Applied Research and Development

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Sammendrag/Abstract:
This report presents the results of a study of the technical aspects of “Internet Protocol based MultiMedia Services over a Satellite Network”. The work is part of the IMMSAT project between Norsk Regnesentral and Nera SatCom AS, partially funded by Norsk Romsenter.

The study starts with an overview of today's terrestrial IP based signalling and transmission protocols and standards, and in particular the H.323 protocol suite. A technical assessment of the applicability of such protocols and standards to a GEO satellite environment is then performed, with special focus on the quality of service achievable. Finally, a summary of the results of an H.323 based prototype simulating the satellite network is presented.

The main conclusion, both derived from the study and confirmed in the testing, is that an H.323 VoIP service can operate quite well in GEO satellite based IP networks.

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1. INTRODUCTION

1.1 Scope and Field of Application

This report presents the results of a study of the technical aspects of “Internet Protocol based MultiMedia Services over a Satellite Network”.

Since the IP network in this study is satellite based, the interworking with and the interfaces to terrestrial networks such as the ISDN/PSTN are of primary interest. The major network components and protocols/standards for such a network are shown in Figure 1.

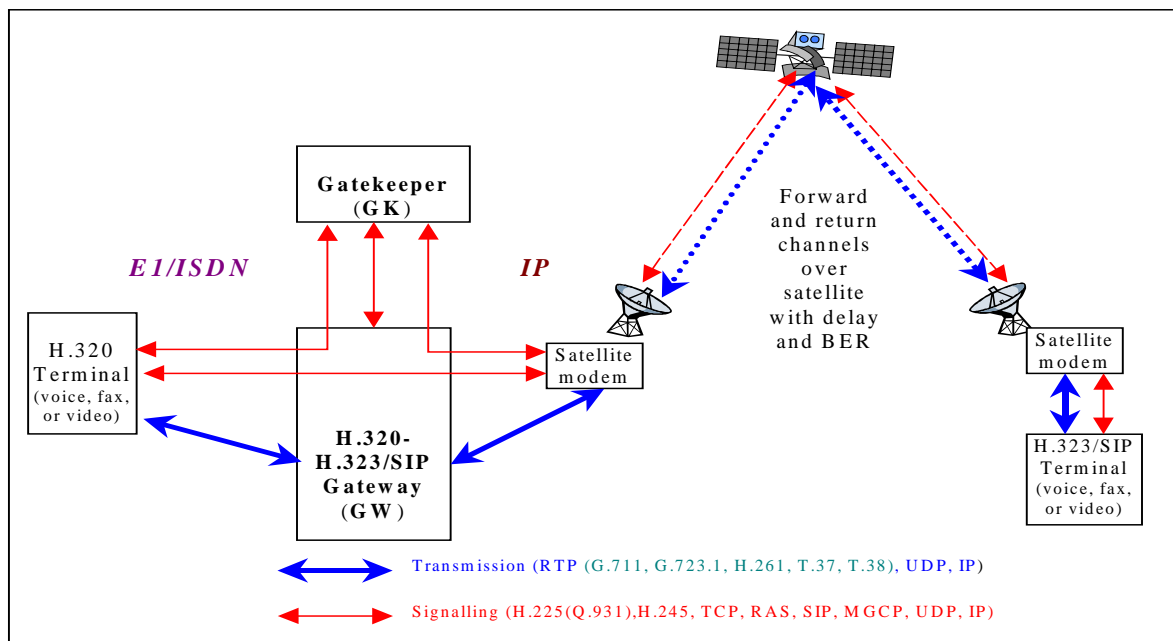


Figure 1: View of Network Components to support IP MultiMedia over Satellite

Some of the other IMMSAT project deliverables such as requirement specification, design description, etc. are not within the scope of this document.

However, there are references in this study report to the prototype/testing documentation.

1.2 Purpose

The purpose of this document is to present the results of the IP MultiMedia over Satellite (IMMSAT) study as well as to identify topics for further study in possible subsequent projects.

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1.3 Definitions and Abbreviations

AAA	Authentication, Authorisation and Accounting
ACK	Acknowledgement
ARP	Address Resolution Protocol
ARQ	Admission Request
ARQ	Automatic Repeat Request
BER	Bit Error Rate
BPSK	Binary Phase Shift Keying
BRI	Basic Rate Interface (ISDN service level)
CDMA	Code Division Multiple Access
CDR	Call Detail Record
COTS	Commercial Off-The-Shelf
CRC	Cyclic Redundancy Check
CT	Computer Telephony
DHCP	Dynamic Host Configuration Protocol
DNS	Domain Name System
FAX	Facsimile
FCAPS	Fault, Configuration, Accounting, Performance and Security
FoIP	Fax over Internet Protocol
FTP	File Transfer Protocol
GK	H.323 Gatekeeper
GEO	Geosynchronous
GW	H.323 Gateway
HDLC	High Level Data-link Control
HLR	Home Location Register (SS#7 - GSM)
HTML	Hypertext Mark Up Language
HTTP	Hypertext Transfer Protocol
HW	Hardware
IAF	Internet Aware Fax
IETF	Internet Engineering Task Force
IMMSAT	IP over MultiMedia Satellite
IP	Internet Protocol
IPCP	Internet Protocol Control Protocol
ISDN	Integrated Services Digital Network
ISP	Internet Service Provider
ISUP	ISDN User Part
ITU	International Telecommunication Union
Kbps	Kilobit per second
Kbit/s	

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LAN	Local Area Network
LLC	Logical Link Control
MAC	Media Access Control
Mbps, Mbit/s	Megabit per second
MCU	Multipoint Controller Unit (H.323)
MIB	Management Information Base
MTP	Message Transfer Part
MUX	Multiplexer
NA	Not Applicable
NAT	Network Address Translation
NCP	Network Control Protocol
NHRP	Next Hop Routing Protocol
NMS	Network Management System
NNI	Network to Network Interface Network Node Interface
PC	Personal Computer
PDU	Protocol Data Unit
PEP	Performance-Enhancing Proxies
PPP	Point-to-Point Protocol
PRA	Primary Rate Access
PRC	Program Reference Clock
PRI	Primary Rate Interface (ISDN service level)
PSTN	Public Switched Telephone Network (standard telephony)
QA	Quality Assurance
QoS	Quality of Service
R&D	Research and Development
RADIUS	Remote Authentication Dial-In User Service
RAS	Registration, Admission, Status
RFC	Request for Comment
RIP	Routing Information Protocol
RRM	Radio Resource Management
RS	Requirements Specification
RTCP	Real-time Transport Control Protocol (multimedia transport control in IP suite)
RTP	Real-time Transport Protocol (multimedia transport in IP suite)
RTSP	RealTime Streaming Protocol: IETF RFC 2326 (video server control)
Rx	Reception
SAP	Session Announcement Protocol: IETF internet draft
SCPC	Single Channel Per Carrier
SDL	Specification and Description Language
SDP	Session Description Protocol: IETF RFC 2327 (Used by SIP, SAP, RTSP)

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SIP	Session Initiation Protocol: IETF RFC 2543bis [21]. (Internet telephony signalling)
SMS	Short Message Service (SS#7- GSM)
SNMP	Simple Network Management Protocol
SP	Service Provider
SS#7	Signalling System Number 7
SSL	Secure Socket Layer
SU	Signalling Unit
SW	Software
TBD	To Be Defined
TCP	Transmission Control Protocol
ToS	Type of Service (IP header)
TUP	Telephony User Part (SS#7)
Tx	Transmission
UAC	User Agent Client (SIP [21] internet telephony)
UAS	User Agent Server (SIP [21] internet telephony)
UDP	User Datagram Protocol
UNI	User to Network Interface
URL	Uniform Resource Locator (various IETF internet standards)
UTC	Universal Time Co-ordinated (formerly GMT)
UTF	US-ASCII compatible multibyte international character set
VoIP	Voice over Internet Protocol
VPN	Virtual Private Network
WAN	Wide Area Network
WWW	World Wide Web

1.4 Study Project Definition

The main objective of the study project is to study, create a prototype, and test new technologies that give maximum utilisation of multimedia services over an IP based satellite network. A related goal is that the participating Norwegian companies develop expertise in the “IP based MultiMedia” technologies such that they can be competitive.

The study will focus on quality of services, signalling protocols, authentication, authorisation, and accounting for multimedia services. Interworking between a satellite-based IP network and circuit switched terrestrial networks is one of the most important issues concerning these topics.

1.5 Participating Partners and Contributors

Norsk Regnesentral and Nera SatCom AS (main contractor) are the participating partners in this IMMSAT project that is partially sponsored by Norsk Romsenter.

The contributors to this study report are:

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2. TECHNICAL BACKGROUND FOR THE STUDY

The sections below consist of some background information and explanations for the study and for the information that is presented in the later sections of this study report.

2.1 ISDN

The term Integrated Services Digital Network or the abbreviation ISDN refer to a number of different things related to end-to-end digital communication including networks, internationally-adopted standards, protocol layers, protocols, and services.

The ISDN is sometimes referred to as a circuit-switched digital network separate from the PSTN. In practice, ISDN is an overlay or segment of the PSTN in many countries since the two are integrated together. Calls between ISDN telephones are often routed through switches and PABXs that are part of the infrastructure used for analog services in the PSTN.

Among the international standard bodies that have issued standards and recommendations on ISDN are ITU (international) and ETSI for Euro ISDN. ITU, which was called CCITT in 1984 when it published the first ISDN Recommendations, tried to establish a common ISDN standard for all countries. However, these same recommendations allow for national variants that include nation-specific information elements in the ISDN data frames through the use of the codeset mechanism[31].

The three protocol layers specified by ITU for ISDN according to the OSI model are physical, data link, and network layers. The physical layer is derived from earlier recommendations since one objective was to carry ISDN service over existing telephone infrastructure. The ISDN interface to the end is based on 2 wire twisted pair copper cable which is the same that is used in the existing analog PSTN and private networks.

The physical layer or layer 1 covers the electrical, formation, and protocol aspects of the interface in order to transmit and receive information over transmission media. The ITU-T I.4xx recommendations are concerned with how ISDN uses the physical media defined in the ITU G.xxx recommendations[32].

The data link layer or layer 2 addresses how to provide error control including detection and re-transmission and flow control of data transmission over the physical link. The protocol used for ISDN layer 2 is LAPD that is defined in ITU-T recommendation Q.921. LAPD is derived from LAPB, which is used in X.25.

The network layer or layer 3 defines the call control and routing/directing of the data flow to the desired destination. The network layer and ISDN layer 3 protocol are defined in ITU-T recommendations Q.930, Q.931, and Q.932.

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Among the services provided by ISDN are transmission and multiple service types including voice, data, fax, and video. In addition, the ITU-T I.2xx and Q.7xx recommendations define ISDN Supplementary Services that include such services as calling line identification/restriction, connected line identification/restriction, call forwarding, call transfer, call waiting, three party service, advice of charge, etc.

ISDN provides end-to-end compatibility for voice and fax services. Calls can be made between analog and ISDN devices anywhere in the world and ISDN services can be carried over the existing telephone network infrastructure[30].

The two ISDN interface standards that are in use today are Basic Rate Interface (BRI) and Primary Rate Interface (PRI).

The Basic Rate Interface defines a digital communications line consisting of three logical TDM channels often referred to as 2B+D. It consists of two Bearer (or B) channels operating at a rate of 64 Kilobits per second and one Data (or D) channel at 16 Kilobits per second. The term ISDN or ISDN line, which is the connection to the user or UNI, is often used synonymously with the Basic Rate Interface.

The Primary Rate Interface is a higher-level network interface defined at the rate of 2.048 (Europe) or 1.544 Mbps (North America, Japan, Taiwan, etc.). The two rates are divided into logical channels 30 B channels and 1 D channel and 23 B channels and 1 D channel respectively. All channels can operate at a rate of 64 Kbps.

The three types of logical digital communication channels on an ISDN interface are:

- 1) B-Channel is the bearer channel for transfer of digital data, video, fax and voice.
- 2) D-Channel is the data channel for transfer of signalling and supervisory information
- 3) H-Channel is a higher bandwidth bearer channel

Normally, the two B channels are treated independently by the network and allow for two simultaneous connections; each carrying voice, fax, data, or video. The B channels can be bonded or linked together to provide an aggregate 128 Kbps channel.

The D channel can also be used to carry packet-mode data over an X.25 network.

2.2 Voice Over IP

“Voice over IP” is a service for providing transmission of voice/audio between two or more users. VoIP service may be provided by an H.323 based system, a SIP based system, or a harmonised system (in the future) that supports both “standards”.

The H.323 or SIP “system entities” control and supervise the setup of, transmission during, and release of “voice calls” over an IP based network.

The users of VoIP service may have terminals that connect directly to an IP network, which is the case for the H.323 based or SIP based systems. Users also may access VoIP services indirectly through an ISDN/PSTN interface to an H.323 Gateway into an IP based network.

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The primary difference between IP telephony and traditional telephony is their switching architectures. While the Internet uses dynamic routing (based on non-geographic addressing), traditional telephony networks use static switching (based on geographic telephone numbering). Circuit switched networks dedicate a fixed amount of bandwidth for each conversation, and thus quality is guaranteed. The Internet, on the other hand, is a packet switched network where all packets are treated independently and routed according to information in their packet header.

IP packets carrying voice are subject to delay, loss, and retransmissions like IP packets carrying other type of data. Therefore, IP based voice packets may need special treatment in networks that have limited bandwidth in order to avoid high delay, jitter and loss.

The basic steps for transmitting voice over IP network are listed below.

1. Audio from microphone or line input is converted from analog to digital samples at the audio input device (telephone or sound card).
2. The samples are copied into memory buffer in blocks of frame length.
3. The IP telephony application estimates the energy levels of the block of samples if IP packets are to be suppressed in “quiet periods” between speech.
4. The block is coded with the selected algorithm.
5. Some header information is added to the block.
6. The block is transferred over a physical network in an IP packet and received by the peer.
7. The header information is removed, the block of audio is decoded based on the same algorithm that was used for encoding, and the samples are written into a buffer.
8. The block of samples is copied from the buffer to the audio output device.
9. The audio output device (telephone or sound card) converts the samples from digital to analog and outputs them to the receiver/headset/speaker.

IP telephony requires extended signalling compared to telephony through circuit-switched networks, since the endpoints can have multitude of capabilities (different bandwidth requirements, audio and video codecs, data capabilities, etc.).

While SS7, the signalling protocol used in all modern circuit-switched networks, alone enables the exchange of messages throughout the network (e.g. for routing, resource reservation, call admission, call establishment, call management and billing), this has to be done by a number of protocols for VoIP in H.323 based systems.

The two standards for IP telephony that exist today are:

- ITU-T Recommendation H.323 [17]
- IETF draft SIP [21]

These standards are described and summarised later in this study report.

2.3 Fax Over IP

“Fax over IP” is a service for providing transmission of text and graphics between two users using FAX machines at the two ends or a Fax machine at one end and an e-mail application at the other end.

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FoIP service is provided by an H.323 based system.

The H.323 “system entities” control and supervise the setup of, transmission during, and release of Fax calls over an IP based network.

Traditionally, fax documents containing text and images have been transferred over the telephone network, as defined in the two ITU-T specifications T.30 [26] and T.4 [37]. The T.30 protocol describes the formatting of non-page data, such as messages that are used for capability negotiation, while the T.4 protocol describes formatting and compression of page image data. A Fax over Packet application enables the interworking of standard fax machines with packet networks. It accomplishes this by extracting the fax image from an analog signal and carrying it as digital data over the packet network.

The reference model for fax transmission over IP networks is shown in Figure 2 with three scenarios.

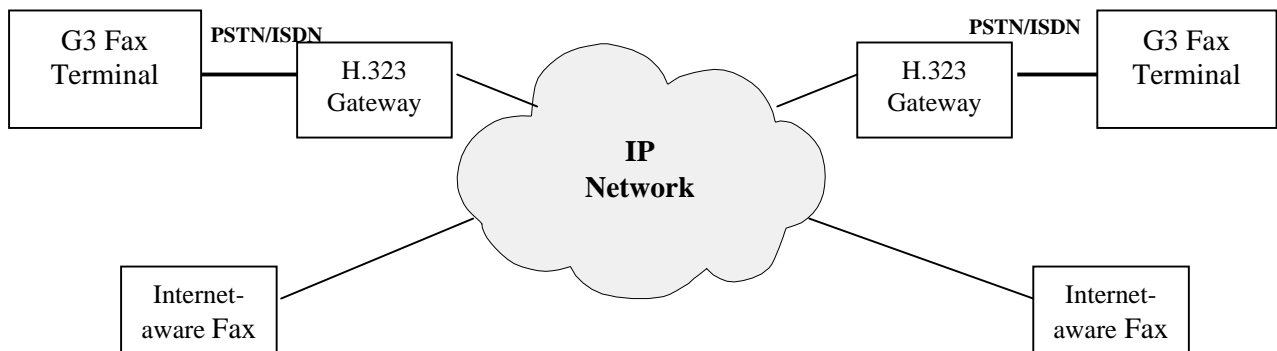


Figure 2: Model for facsimile transmission over IP networks

1. Two traditional Group 3 Facsimile terminals virtually connected through the gateways once the calls are established.
2. A traditional Group 3 Facsimile terminal is connected with an Internet Aware Fax terminal (IAF).
3. Two IAFs are directly connected to the IP network.

There are two approaches for sending Fax over IP networks regardless of the type of Fax terminals connected: "real-time" methods and "store-and-forward" methods. The primary difference in service between these two approaches is in the delivery and method of receipt confirmation.

In the "real-time" session oriented approach, as defined in ITU-T specification T.38 [39], a receipt confirmation is provided to the transmitting terminal prior to disconnection. In the "store-and-forward" approach, as defined in ITU-T specification T.37 [38], the fax transmission/reception is not performed in real-time. Also, T.37 uses Internet email to “store” the entire document at a staging point, prior to transmitting it to the next staging point.

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Recommendations for T.37 (Simple Mode) and T.38 were approved by the ITU-T in June 1998 while T.37 (Full Mode) was approved in September 1998.

T.38 is the fax transmission protocol selected for H.323.

There is also some work in progress in IETF concerning both session-based and messaging-based fax over the Internet.

2.4 Video Conferencing

Video conferencing is the real-time transmission of audio and video between two or more end users, which are physically separated. It requires microphones and cameras for audio and video capturing, as well as screens and headsets/speakers for playing and displaying the content.

Professional “desktop video conferencing solutions” have traditionally been ISDN-based and adhering to the ITU-T H.320 recommendation from 1994. Technically, such a session requires a number (p) of 64 Kbps dedicated lines (B-channels) over the circuit-switched telephony network, with p ranging from 1 to 30.

With the growing popularity of the Internet, an IP-based desktop video conferencing solution has been desirable. The most popular alternative is solutions adhering to the ITU-T H.323 recommendations. The first version was approved in 1996, but most products are now based v2 from 1998. A third version has also been approved, and there's ongoing work with v4. An alternative and complementary solution is coming from the ranks of IETF. The SIP and SDP standards provide lightweight conference session initiation and description compared to the more heavy telecommunication approach from ITU-T.

The main advantage with IP-based solutions is the lower price of bandwidth and the use of an already existing IP infrastructure compared to ISDN solutions. The main technical challenge is to provide stable QoS conference environments when it comes to bandwidth and delay. Resource reservation at IP level is still in its infancy, and firm QoS guarantees can not be given at the present stage. Instead, over-provisioned ATM connections are often the temporary solution until better IP level approaches emerge.

2.5 Properties of Satellite Environments

Communication systems that use satellite links must compensate for the properties of the satellite environment such as propagation delay, Bit Error Rate (BER), free space loss, doppler shifts, etc. The propagation delay is directly dependent on the type of satellite constellation. Three types of satellite constellations are described below.

Most of the satellites used for communications today are orbiting the earth in a geosynchronous orbit (GEO).

A satellite in a geosynchronous orbit is at an altitude of 35,786 kilometers (22,282 miles). GEO satellites have a circular orbit over the equator. Since GEO satellites complete one orbit every 24 hours (same period for 1 rotation of the earth) and move in the same direction as the earth's rotation, the satellite remains in a fixed position relative to a point on the surface of the earth on the equator. Therefore, a GEO satellite provides the possibility of continuous contact with ground stations in its line of sight. The footprint for a GEO satellite is around 1/3 of the earth's surface excluding the earth's poles. A constellation of 3 GEO

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satellites can cover the earth except for the polar regions. GEO systems have high propagation delay and high free space loss.

The round-trip delay when using a GEO satellite is between 479 and 558 milliseconds. This is the delay measured:

- *from the point in time* that a message (e.g. packet A) is sent from a satellite gateway (on the surface of the earth) over a satellite to a terminal (on the surface of the earth)
 - *until the point in time* that a reply message (e.g. packet B sent subsequently by that terminal as a response to packet A) is received by the satellite gateway via the satellite from that same terminal.
- This is just the delay related to radio signal propagation. There is additional delay in satellite based systems due to processing, queuing, etc. that can vary greatly from system to system (e.g. from 50 to 700 ms).

The latest development in satellites is the use of networks of small satellites in low earth orbit (LEO) or medium earth orbit (MEO). LEO satellites have elliptical or circular orbits at altitudes of 2,000 km (1,200 miles) or less. A single orbit takes 1.5 to 2 hours and a satellite is “in-sight” above the horizon for up to 20 minutes. Since the satellite disappear over the horizon, “hand-over” to a satellite in the same or adjacent orbit must occur. The footprint for a LEO is between 3000 and 4000 kilometers. Therefore, a global communication system using LEO satellites usually consist of 40 to 75 satellites in 6 to 8 orbital planes. LEO satellites move at high speeds relative to the earth and systems using them are therefore susceptible to doppler shifts. Globalstar (48 satellites) and Iridium system (66 satellites) are examples of LEO based systems. LEO satellites are slowed gradually by resistance of the atmosphere and their orbits eventually deteriorate.

MEO satellites move in circular or elliptical orbits at altitudes around 10000 km (6,000 miles). A single orbit takes around 6 hours and a satellite is “in-sight” above the horizon for a few hours. Since the MEO satellites also disappear over the horizon, “hand-over” to a satellite in the same or adjacent orbit must occur but less often than with LEO systems. Communications systems using MEO satellites usually consist of around 10 satellites spread in 2 to 3 orbital planes. LEO satellites have propagation delay and free space loss that is higher than LEO systems but lower than GEO systems. MEO systems are much less affected by doppler shifts. The planned ICO (10 satellites) and Odyssey (12 satellites) are examples of MEO based system.

Buffer size of 64K limits maximum throughput as the increased latency of a GEO connection decreases the available bandwidth. (TCP buffers are dimensioned as: bandwidth * delay = buffer size)

With a limited buffer size, a longer end-to-end delay decreases the space available to hold spare copies of unacknowledged data for retransmission. This limits the throughput on a lossless TCP connection.

Wide pipes with large buffer sizes can suffer from the higher bit error rate (BER) with transmission over satellites. Better symbol coding down at the satellite's data-link layer to achieve a lower BER may be more effective than solutions at the application layer[25].

For TCP in a satellite based IP network, implementing Selective Acknowledgements (RFC 2018) [108] and fast recovery (RFC2581) [107] may improve performance if the BER is high compared to terrestrial-based IP networks.

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2.6 ITU H.320: Narrow-band Visual Telephone Systems and Terminal Equipment

ITU-T recommendation H.320 covers the technical requirements for terminal equipment for audiovisual telephone systems over narrow-band ISDN. Details on data framing, control signals, procedures, etc. are defined in the standards H.221 (framing), H.230 (control & indication signals), H.242 (procedures) and H.243 (multipoint conferences). The H.320 terminals may also have capability for data communication services like shared drawings using the T.120 series of standards. Refer to Figure 3.

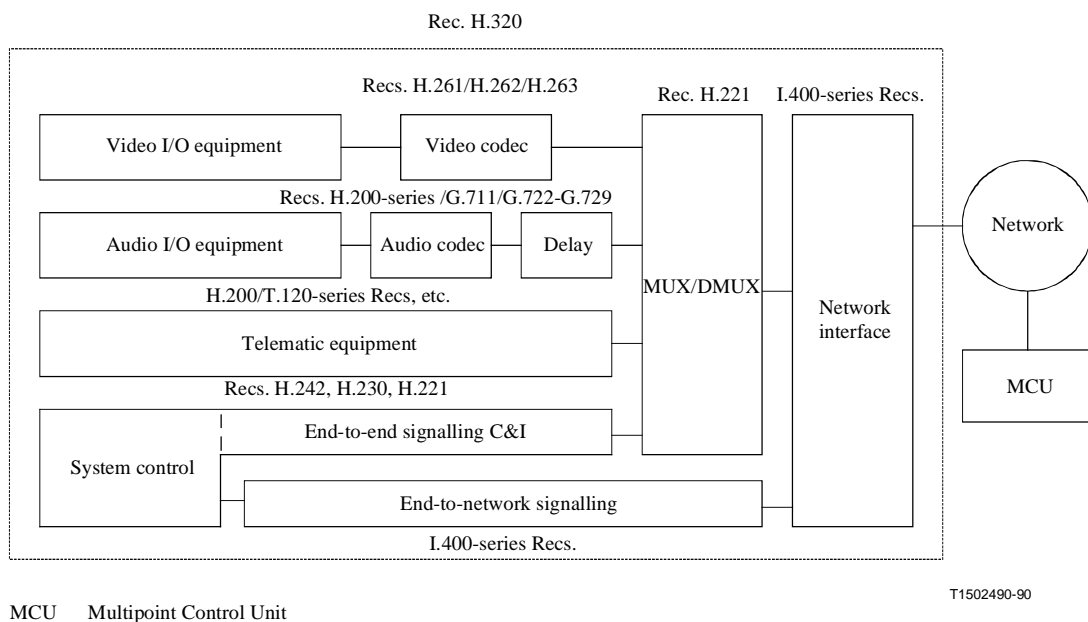


Figure 3: H.320 protocol suite (from [19])

H.320 terminals are mainly stand-alone desktop sets or integrated in special teleconferencing rooms. They operate over ISDN in the bandwidth range 64 kbit/s (single B-channel) to 1920 kbit/s (one H₁₂ channel).

2.7 ITU H.323: Packet-based Multimedia Communications Systems

ITU-T recommendation H.323 [17] was originally titled "Visual telephone systems and equipment for local area networks which provide a non-guaranteed quality of service". The standard describes equipment and procedures for establishing multimedia communication services like point-to-point connections, multipoint conferences and broadcast sessions. The scope, and correspondingly the title, of the standard was expanded in version 2 to include communication across all kinds of packet based networks, including the Internet. Version 3 was determined in September/October 1999, and ITU study group 16 is currently working on version 4. Still, most commercially available H.323 products adhere to version 2.

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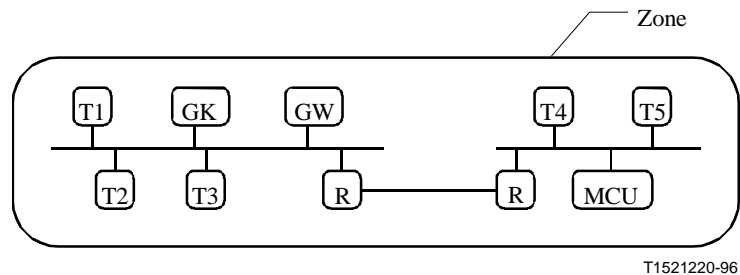


Figure 4: H.323 entities (from [17]).

H.323 defines the following entities (as shown in Figure 4):

- Terminals (Tx):** May be standalone desktop terminals or software inside a computer.
- Gatekeeper (GK):** Entity responsible for name resolution, access control and bandwidth reservation (if possible) for terminals and users within the zone. Optional, but must be used for registration and access requests by the terminals if present. Most installations will probably contain (at least) one.
- Gateway (GW):** Bridge for communication with partners on other kinds of networks and standards, for instance H.320 equipment on narrow band ISDN. Optional.
- Multipoint Control Unit (MCU):** Physical or logical entity inside (one of) the terminals for control of multipoint conferences. Must contain at least one Multipoint Controller (MC) for controlling conference members, etc. May also contain a Multipoint Processor (MP) used for processing the media streams (like mixing all audio tracks to one).
- Zone:** Originally: Local Area Network segment(s) used for H.323 communication and controlled by a single gatekeeper. Now: Logical "area" containing the terminals, Gateways, etc. that is controlled by the same Gatekeeper.

The call, including its setup and termination, proceeds in 5 phases, as described later.

2.8 IETF RFC 2543: Session Initiation Protocol (SIP)

The Session Initiation Protocol draft [21] contains the following introduction:

The Session Initiation Protocol (SIP) is an application-layer control protocol that can establish, modify and terminate multimedia sessions or calls. These multimedia sessions include multimedia conferences, distance learning, Internet telephony and similar applications. SIP can invite both persons and "robots", such as a media storage service. SIP can invite parties to both unicast and multicast sessions; the initiator does not necessarily have to be a member of the session to which it is inviting. Media and participants can be added to an existing session.

[...]

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SIP transparently supports name mapping and redirection services, allowing the implementation of ISDN and Intelligent Network telephony subscriber services. These facilities also enable personal mobility.

[...]

Personal mobility complements terminal mobility, i.e. the ability to maintain communications when moving a single end system from one subnet to another.

SIP supports five facets of establishing and terminating multimedia communications:

***User location:** determination of the end system to be used for communication;*

***User capabilities:** determination of the media and media parameters to be used;*

***User availability:** determination of the willingness of the called party to engage in communications;*

***Call setup:** "ringing", establishment of call parameters at both called and calling party;*

***Call handling:** including transfer and termination of calls.*

[...]

SIP can also be used in conjunction with other call setup and signaling protocols. In that mode, an end system use SIP exchanges to determine the appropriate end system address and protocol from a given address that is protocol-independent. For example, SIP could be used to determine that the party can be reached via H.323, obtain the H.245 gateway and user address and then use H.225.0 to establish the call. In another example, SIP might be used to determine that the callee is reachable via the PSTN and indicate the phone number to be called, possibly suggesting an Internet-to-PSTN gateway to be used. SIP does not offer conference control services such as floor control or voting and does not prescribe how a conference is to be managed, but SIP can be used to introduce conference control protocols. SIP does not allocate multicast addresses. SIP can invite users to sessions with and without resource reservation.

SIP does not reserve resources, but can convey to the invited system the information necessary to do this.

SIP exchanges information coded according to the SDP standard for determination of user capabilities. The SIP and SDP standards are based on exchanging messages built up of structured lines of UTF-encoded text. In SIP these messages can be exchanged via reliable (TCP) or unreliable (UDP) communication.

2.9 IETF RFC 2327: Session Description Protocol (SDP)

SDP (Session Description Protocol) [22] is a simple, text-based protocol for describing the media contents of communication sessions. Pure SDP messages are not exchanged between communicating partners, rather they are used for session descriptions inside certain messages in other IETF standards like SIP, SAP (Service announcement protocol) and RTSP (Real-time streaming protocol).

SDP messages contains both general information about the session, and specific information about each of the media streams that may be exchanged in the session. General session information includes name and originator (creator) of the session, and at least one timestamp indicating when the session is active. Information about each media includes its type, allowed formats, and the destination transport address.

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2.10 ITU G.711 : Pulse Code Modulation (PCM) of Voice Frequencies

ITU-T recommendation G.711 [2] covers analog-to-digital encoding of telephony audio on a 64 kbit/s channel. The standard uses pulse code modulation (PCM) of voice frequencies operating at 8000 samples per second, and with 8 bits per sample.

Two different encoding laws are provided, A-law and μ -law. A-law is the standard for international telephony circuits between different countries. G.711 includes conversion tables between the two variants.

Each of the encoding laws is designed in a roughly logarithmic fashion. Lower signal values are encoded using more bits than higher signal values.

2.11 ITU G.723.1 : Dual Rate Speech Coder for Multimedia Communications

ITU-T recommendation G.723.1 [3] specifies a coded representation that can be used for compressing the speech or other audio signal component of multimedia services at a very low bit rate. The coder has two bit rates associated with it, 5.3 and 6.3 kbit/s. The higher bit rate has greater quality. The lower bit rate gives good quality and provides system designers with additional flexibility. Both rates are a mandatory part of the encoder and decoder. It is possible to switch between the two rates at any frame boundary.

The G.723.1 coder encodes speech or other audio signals in frames using linear predictive analysis-by-synthesis coding. The excitation signal for the high rate coder is Multipulse Maximum Likelihood Quantization (MP-MLQ) and for the low rate coder is Algebraic-Code-Excited Linear-Prediction (ACELP). The frame size is 30 ms and an additional look ahead of 7.5 ms. This results in a total algorithmic delay of 37.5 ms. All additional delays in this coder are due to processing delays of the implementation, transmission delays in the communication link, and buffering delays of the multiplexing protocol.

Input to the encoder (and output from the decoder) is 16-bit linear PCM sampled at 8000 Hz. Other audio characteristics, e.g. 64 kbit/s PCM data specified by G.711, should be converted to 16 bit linear PCM before encoding (and from 16 bit linear PCM after).

2.12 ITU G.729A: Reduced Complexity 8 Kbit/s CS-ACELP Speech Codec

ITU-T recommendation G.729 [45] describes an algorithm for the coding of speech signals at 8 kbit/s using Conjugate-Structure Algebraic-Code-Excited Linear-Prediction (CS-ACELP). Annex A [46] of the recommendation specifies a reduced complexity version of the full G.729 speech codec. The version is known as G.729A, and has been developed for multimedia simultaneous voice and data applications, although the use of the codec is not limited to these applications.

The G.729A coder encodes speech or other audio signals in 10 ms frames. In addition, there is a look-ahead of 5 ms, resulting in a total algorithmic delay of 15 ms. As for G.723.1, all additional delays in this coder are due to processing, transmission and buffering delay.

The CS-ACELP algorithm is based on the Code-Excited Linear-Prediction (CELP) coding model. Input to the encoder (and output from the decoder) is 16-bit linear PCM sampled at 8000 Hz. As for G.723.1, other audio characteristics should be converted to/from that format before use.

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2.13 ITU H.261 : Video Codec for Audiovisual Services at $P \times 64$ Kbit/s

ITU-T recommendation H.261 [1] describes the video coding and decoding methods for the moving picture component of audiovisual services at the rates of $p \times 64$ kbit/s, where p is in the range 1 to 30.

Pictures are coded as a luminance component and two colour difference components for blue and red. A picture in the H.261 model consists of three rectangular matrices with 8 bits per sample.

Two picture scanning formats are specified.

In CIF (Common Intermediate Format) the luminance sampling structure is 352 pixels per line, 288 lines per picture in an orthogonal arrangement. Sampling of each of the two colour difference components is at 176 pixels per line, 144 lines per picture, orthogonal. Colour difference samples are sited such that their block boundaries coincide with luminance block boundaries. The picture area covered by these numbers of pixels and lines has an aspect ratio of 4:3 and corresponds to the active portion of the local standard video input.

QCIF (Quarter-CIF) has half the number of pixels and half the number of lines stated above, i.e. 176 pixels per line, 144 lines per picture for the luminance components, and 88 pixels per line, 72 lines per picture for the two colour difference components. All codecs must be able to operate using QCIF. Some codecs can also operate with CIF.

H.261 does not make use of quantization matrices, thereby enforcing a constant data rate at the output of the coder. Therefore, the quality of the encoded video depends on contents of individual images as well as motion within the respective video scene.

2.14 ITU H.263: Video Coding for Low Bit Rate Communication

The ITU-T Recommendation H.263 [40] specifies a video compression algorithm and protocol which is targeted for the transmission through low bandwidth channel. It builds on the experience of H.261 and was published in 1996. At rates below 64 kbps, H.263 can achieve a picture quality similar to H.261 with 30-50% of the bit rate. Most of this is due to the half pixel prediction and negotiable options in H.263. There is also less overhead and improved VLC (Variable Length Codeword) tables in H.263. H.263 is also better than MPEG-1 and MPEG-2 for low resolutions and low bit rates.

It is expected that the standard will be used for a wide range of bit rates (10kb/s - 2 Mb/s), not just low bit rates, and that H.263 will replace H.261 in many applications.

H.263 supports five resolutions. In addition to QCIF and CIF that were supported by H.261 there is SQCIF, 4CIF, and 16CIF. SQCIF is approximately half the resolution of QCIF. 4CIF and 16CIF are 4 and 16 times the resolution of CIF respectively. The support of 4CIF and 16CIF means the codec could then compete with other higher bitrate video coding standards such as the MPEG standards.

H.263 version 2 ("H.263+") [41], was approved as a standard in 1998. H.263+ is an extension of H.263 providing 16 negotiable modes and additional features. The objective of H.263+ is to broaden the range of applications and to improve compression efficiency. H.263 version 2 is backward compatible with H.263.

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2.15 ITU T.37 : Procedures for Transfer of Fax Data via Store-and-forward on the Internet

The ITU-T Recommendation T.37 [38] describes the technical features necessary for the Store-and-Forward of facsimile document transmission via Internet email. This work was based on the Internet Engineering Task Force (IETF) specifications RFC 2305 [43], which set IP fax protocol specifications, and RFC 2301 [44], defining standard file formats. The result is that both the IETF and ITU-T have endorsed a common definition for extended/full-mode Internet faxing.

When first released, the T.37 "simple mode" offered no negotiation between fax servers, so only the lowest common denominator of image compression and page size formats were supported. In this mode, T.37 IP fax offered no confirmation of delivery.

The T.37 "full mode" standard, released in September 1998, provided capability negotiations with the recipient, positive delivery confirmation and support for six additional Tag Image File formats. These new file formats allow delivery of compressed fax files and colour faxes.

2.16 ITU T.38 : Procedures for Real-time G3 Fax Communication over IP Networks

The ITU-T Recommendation T.38 [39] defines an Internet facsimile protocol consisting of messages and data exchanged between facsimile gateways and/or Internet Aware Facsimile terminals (IAFs) connected via an IP network. The fax signals are sent over the IP network using TCP or UDP depending on the service environment.

T.38 packets are used on the IP network to communicate T.30/T.4 [26][37] facsimile information. Good image quality is provided by error control in the network in addition to the means provided by the recommendation T.30.

Communication between a sending Group 3 facsimile terminal and the incoming gateway is generally effected using dialup procedures over PSTN/ISDN. H.323 procedures are used for opening channels to send T.38 packets in real-time over the IP network [17] (annex D).

3. SIGNALLING PROTOCOLS

The sections below describe the major signalling protocols used for multimedia services in IP based networks and those used on the PSTN/ISDN side of an H.323 Gateway.

The first sections below briefly point out the signalling protocols that can be considered to have a UNI or UUI or NNI.

The sections that follow describe the AAA protocols, H.320, H.323 protocols and SIP based protocols for multimedia.

3.1 Signalling Protocols and UNI and UUI

The terms UNI and UUI are abbreviations for interfaces that are typically associated with ATM or Frame Relay or ISDN. One of the reasons that UNI and UUI are used in this section is to identify the user and network entities in a H.323 based network.

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The UNI is defined as a User to Network interface and is the interface where the user and network entities communicate with protocols so that the end terminal (user) can get access to and make use of the network services.

For H.323 services, two of the protocols that can be associated to a UNI are RAS and a subset of the ISDN Q.931 protocol for H.323, both of which are described in the ITU-T Recommendation H.225.

RAS messages are exchanged between H.323 endpoints and an H.323 Gatekeeper. RAS messages are also exchanged between H.323 Gateways and an H.323 Gatekeeper since a Gateway is considered to be an H.323 endpoint. The H.323 clients and H.323 Gateways are the users and the H.323 Gatekeeper is the network entity on UNIs.

The H.225 subset of Q.931 signalling messages used to set up calls may be routed via the GK or be exchanged directly between H.323 endpoints. When the endpoints exchange RAS messages associated with a call with the H.323 Gatekeeper, the GK tells the H.323 endpoints which call signalling method will be used:

- Gatekeeper Routed Call Signalling (see Figure 5)
- Direct Endpoint Call Signalling (see Figure 6)

In the case of Gatekeeper Routed call signalling, the signalling messages are passed from an H.323 endpoint (GW or client) to the H.323 GK and from the H.323 GK to another H.323 endpoint (GW or client). With the exchange of these Q.931 call signalling messages from user entity to network entity, the interfaces could be considered to be UNI.

In the case of Direct Endpoint call signalling, the call signalling messages are passed directly between the endpoints (users). With the exchange of these Q.931 messages call signalling messages, the interface between the H.323 endpoints could be considered to be a User to User Interface (UUI).

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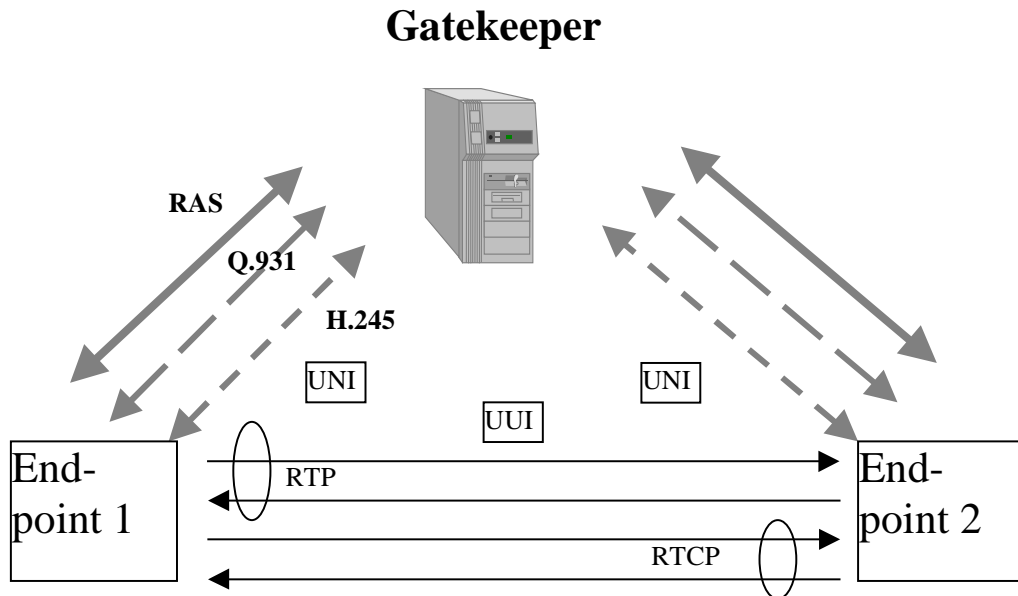


Figure 5: Gatekeeper Routed Call Signalling

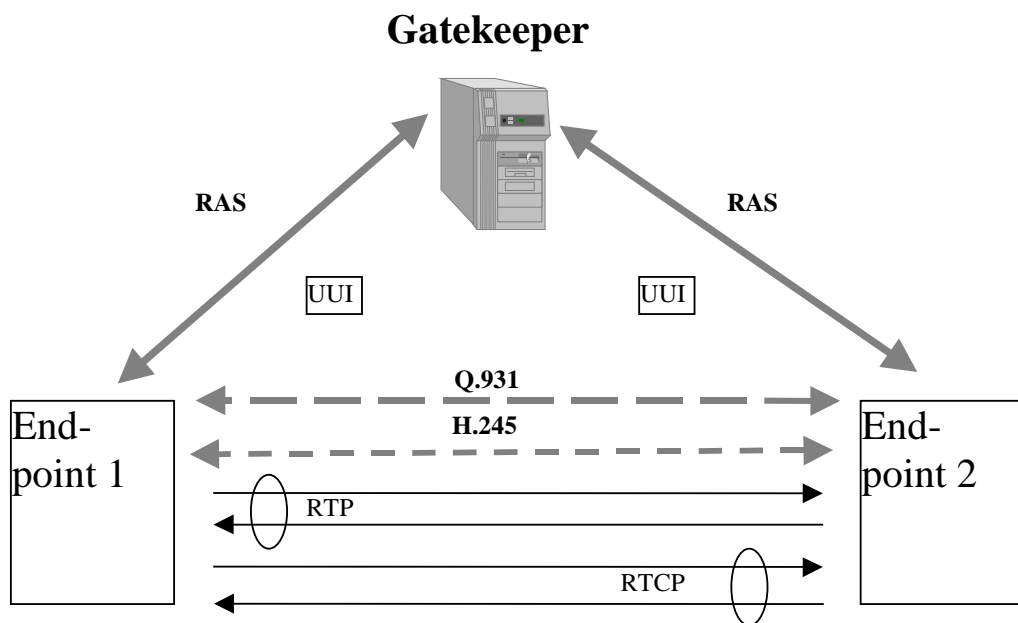


Figure 6: Direct Endpoint Call Signalling

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3.1.1 ISDN

Both the ISDN BRI and PRI can be considered to be UNIs. The user or customer premises equipment at the user side of the network communicates through a UNI to a telephone switch that is a network entity. For both the BRI and PRI, the D channel is used for signalling.

The ISDN signalling interface between the ISDN user terminal and the network is defined on three layers- physical, data link, and network.

The interface for the Physical layer is defined in I.430 for BRI and I.431 for PRI interfaces. The physical layer provides for information to be transmitted and received over a physical media. It provides a means of synchronisation between two end points using a frame pattern and a clocking signal. The frame pattern allows the beginning of a frame to be detected and the clock signal allows the signal to be sampled. In addition, the physical layer uses digital encoding schemes to provide signal balancing and synchronization.

The protocol for the Data Link layer is defined in Q.920 (data link), Q.921 (LAPD), and Q.922 (Frame Relay). The data link layer provides a means for error and flow control for data transferred over the physical link. A method of addressing defined in Q.921 allows for support of multiple logical links. The D channels can therefore be used for signalling, maintenance, and data services. The B channels can also carry signalling in the form of data link layer protocols such as LAPB (X.25) or V.120.

The protocol for the Network layer is defined in Q.930, Q.931, and Q.932. The network layer provides a means for providing routing information. Recommendation Q.931 is an implementation of call signalling and other information that is sent "out of band" using the D channel. Thus, the signalling and data transfer is performed on separate logical channel, the D and B channels respectively. Recommendation Q.932 describes supplementary services that go beyond the basic setting up and tearing down of B channel connections.

3.2 Signalling Protocols and NNI

The term NNI is an abbreviation that is typically associated with ATM or Frame Relay or ISDN rather than with H.323 or SIP. Again here, the user and network entities in a VoIP system can be identified.

The NNI is defined as a Network to Network interface and is the interface where two network entities communicate with protocols in the network segment of the users' end to end connection.

However, a Network to Network Interface does have significance from the SS#7 protocols. The NNI Signalling Protocols are those protocols used for signalling between network entities such as switches. ISDN PRI's are normally not used in the core of the network. The ISDN BRI and PRI interfaces are used toward users and customer premises equipment such as PABXs respectively and are more UNI interfaces.

The signalling protocols used on the NNI in the PSTN/ISDN are those specified in the ITU-T Q series Recommendations on Signalling System No. 7.

Therefore, the ISDN signalling used between a user in the ISDN and an H.323 Gateway may be carried in SS#7 signalling over an NNI consisting of one or more telephone exchanges.

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The interface between H.323 Gatekeepers that cover different zones of H.323 clients or users can also be considered a NNI. The signalling protocols passed on this type of NNI will be RAS for registration of the GKs with each other as well as other control or routing related information. In the situations where Gatekeeper Routed signalling is used, Q.931 messages defined in H.225 will also be passed on the NNI between GKs when the H.323 endpoints involved in a call setup are registered on different H.323 GKs.

3.2.1 Signalling System No. 7

SS7 is the protocol used in the Common Channel Signalling network for today's switched telecommunication network. It is an out-of-band signalling network for exchanging call control information between switches, providing support for both voice and non-voice switched services. Like the ISDN protocols, there are many national variants of the SS#7 protocols.

SS7 provides two basic types of services:

Call Control:

SS7 provides fast and reliable common channel or "out-of-band" signalling for call control. At the heart of the SS7 call control function is a network of ultra-reliable packet switches called Signal Transfer Points.

Intelligent Network:

The SS7 network enables the implementation of IN/Advanced Intelligent Network (AIN) services. SS7 messages traverse STPs and enlist the use of System Control Points (SCP), Service Switching Points (SSP), and Intelligent Peripherals (IP) to deliver these services to the user.

3.2.1.1 The SS7 protocol architecture

SS7 is a layered software architecture that consists of message transfer and user parts.

Message Transfer Part:

MTP 1-3 provide a reliable but connectionless transport service for the higher layers.

The overall purpose of MTP is to transport information from the upper layers (including the user parts and SS7 applications) across the SS7 network.

User Parts:

- ISUP/TUP (call control): Provides standards-based and network-specific call control services for wireless and wireline PSTN networks.
- SCCP (Signal Connection Control Part): Provides address resolution services (i.e., global title) for locating services within the network. SCCP is available in two versions: connectionless only and connection-oriented plus connectionless.
- TCAP (Transaction Capabilities Application Part) (transaction services): Used for transporting transaction-oriented data across the SS7 network. TCAP implements standard Remote Operation Service Element (ROSE) services for applications such as GSM-MAP and IS-41. These "applications" provide IN services such as Home Location Register or Short Message Service.

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- GSM-MAP, IS-41, and INAP: Transaction-based services that allow development of applications such as Short Message Service, access to HLR/VLR (for wireless networks), and other IN applications.

3.3 RADIUS

The Remote Authentication Dial In User Services (RADIUS) protocol is a software-based security authentication protocol developed by the Internet Engineering Task Force (IETF) RADIUS Working Group. The RADIUS protocol is described in RFC 2138[27] and the RADIUS accounting protocol is described in RFC 2139[28].

The two main functions of RADIUS are:

- Authentication and Authorization
- Accounting

Authentication is the process of identifying and verifying a user. Several methods can be used to authenticate a user, but the most common includes a combination of user name and password. Once a user is authenticated, authorization to various network resources and services can be granted. Authorization determines what a user can do, and accounting is the action of recording what a user is doing or has done.

Centralized authentication and authorization of dial-in users provide network security. Accounting then measures the resources that each user consumes.

The RADIUS protocol is based on a client/server model. A Network Access server (NAS) operates as a client towards a RADIUS server. The client is responsible for passing user information to designated RADIUS servers, and then acting on the response that is returned from the NAS.

3.3.1 Packet format

The data between client and server is exchanged in RADIUS packets. Exactly one RADIUS packet is encapsulated in the UDP detailed. Every packet contains the following information:

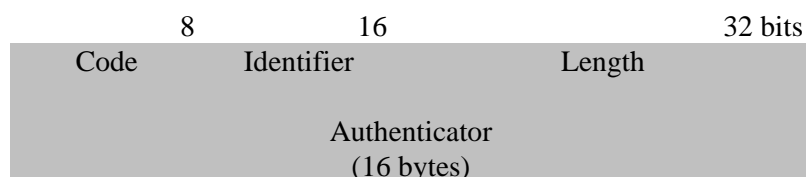


Figure 7: RADIUS packet format (from [16])

The fields in a RADIUS packet are:

Code

An octet containing the RADIUS command/response.

Identifier

An octet used to match the command and response.

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Length

The length of the packet (2 octets).

Authenticator

A 16 octets value used to authenticate the reply from the RADIUS server, and used in the password-hiding algorithm.

Attributes

The data belonging to the command or response.

RADIUS communication uses the request-response paradigm. Requests are issued by the client and sent to the RADIUS server. Responses are issued by the RADIUS server and sent to the client.

Possible request-response pairs are:

- **access-request**, (client->server), request access for a user with certain services. The possible responses for this command are:
 - **access-accept**, (server->client), positive response on an access-request from a client.
 - **access-reject**, (server->client), negative response on an access-request from a client.
 - **access-challenge**, (server->client), response on an access-request, where the server expects a response from the client encapsulated in an access-request.
- **accounting request**, (client->server), request to store accounting data within packet on the server. The response for this command is:
 - **accounting response**, (server->client), response to client when accounting data has successfully been stored on the server.

3.3.2 Sequence diagram

Below in Figure 8 is a drawing of a sequence diagram illustrating the RADIUS messages exchanged when a user accesses and logs into the network through the Network Access Server and when the user disconnects from the network. The sequence is:

1. Network Access Server get username/password pair from remote user, encrypts this information with a shared secret key and sends this with an '**Access-request**' to the RADIUS Server (Authentication phase).
2. After the user and password combination has been validated and found OK, the RADIUS Server sends an '**Access-accept**' with extra information (for example: IP-address, network mask, allowed session time, etc.) to the Network Access Server (Authorization phase).
3. The Network Access Server sends an '**Accounting-request (Start)**' to indicate that the user is logged on to the network (Accounting phase).
4. A call record for this particular call/session is normally created, and the RADIUS Server responds with an '**Accounting-response**' when the accounting information is stored.

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5. When a user logs out, the Network Access Server will send an '**Accounting-request (Stop)**' with the following information:
 - Delay time, the time it is trying to send this message.
 - Input octets, the number of octets received by the user.
 - Output octets, the number of octets send by the user.
 - Session time, the number of second the user is logged on.
 - Input packets, the number of packets received by the user.
 - Output packets, the number of packets send by the user.
 - Reason, reason why the user is disconnected from the network.

6. The RADIUS Server responds with an '**Accounting-response**' when the accounting information is stored.

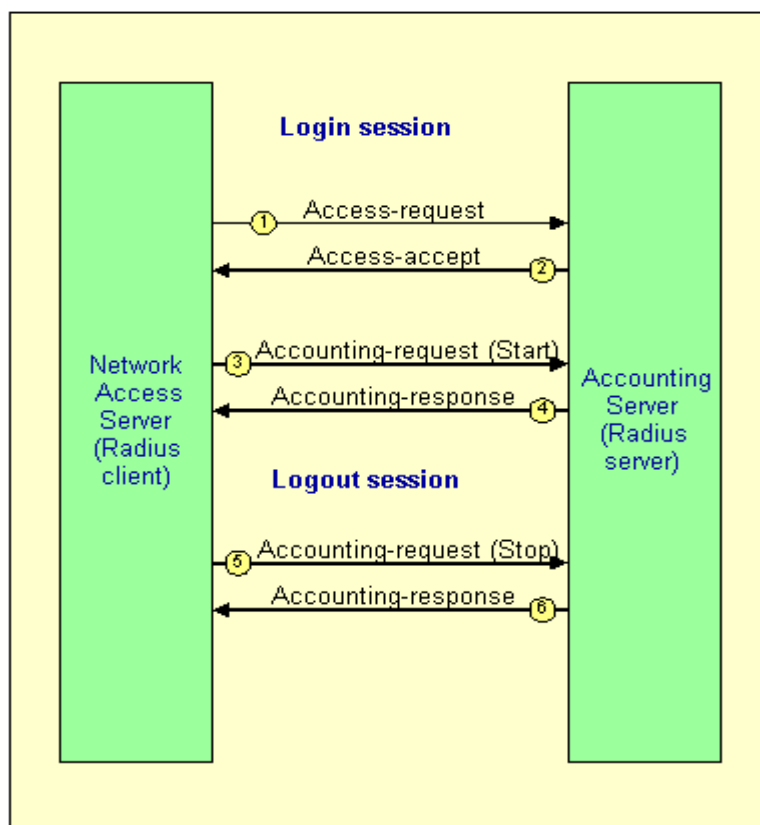


Figure 8: RADIUS Message Flow (fig. From [29])

3.4 H.323 RAS (Registration/Admission/Status)

The RAS signalling function uses H.225.0 [4] messages to perform registration, admissions, bandwidth changes, and status and disengage procedures between endpoints and Gatekeepers. The RAS Signalling Channel is independent from the Call Signalling Channel and the H.245 Control Channel. H.245 open

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logical channel procedures are not used to establish the RAS Signalling Channel. In network environments that do not have a Gatekeeper, the RAS Signalling Channel is *not* used. In network environments which contain a Gatekeeper (a Zone), the RAS Signalling Channel is opened between the endpoint and the Gatekeeper. The RAS Signalling Channel is opened prior to the establishment of any other channels between H.323 endpoints.

RAS messages are transmitted on an unreliable channel (i.e. UDP). H.225.0 recommends timeouts and retry counts for various messages (see below). All RAS messages are given in the table below.

H.225.0 – RAS messages

RAS Message	Endpoint (Tx)	Endpoint (Rx)	Gatekeeper (Tx)	Gatekeeper (Rx)
GRQ	O			M
GCF		O	M	
GRJ		O	M	
RRQ	M			M
RCF		M	M	
RRJ		M	M	
URQ	O	M	O	M
UCF	M	O	M	O
URJ	O	O	M	O
ARQ	M			M
ACF		M	M	
ARJ		M	M	
BRQ	M	M	O	M
BCF	M (Note 1)	M	M	O
BRJ	M	M	M	O
IRQ		M	M	
IRR	M			M
IACK		O	CM	
INAK		O	CM	
DRQ	M	M	O	M
DCF	M	M	M	M
DRJ	M (Note 2)	M	M	M
LRQ	O		O	M
LCF		O	M	O
LRJ		O	M	O
NSM	O	O	O	O

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XRS	M	M	M	M
RIP	CM	M	CM	M
RAI	O			M
RAC		O	M	
SCI	O	O	O	O
SCR	O	O	O	O

M Mandatory, O Optional, F Forbidden, CM Conditionally Mandatory, blank "Not Applicable".

NOTE 1 – If a gatekeeper sends a BRQ requesting a lower rate, the endpoint shall reply with BCF if the lower rate is supported, otherwise with BRJ. If a gatekeeper sends a BRQ requesting a higher rate, the endpoint may reply with either BCF or BRJ.

NOTE 2 – Terminal shall not send DRJ while on a call in response to DRQ from a gatekeeper.

The *mandatory* RAS messages can be divided into 9 functional groups described below. Note that H.323 entities of version 3 or earlier are not able to decode the *optional* SCI message, and reply with the *optional* SCR message.

Gatekeeper discovery (GRQ, GCF, GRJ)

The GRQ message requests that any gatekeeper receiving it respond with a GCF granting it permission to register. The GRJ is a rejection of this request indicating that the requesting endpoint should seek another gatekeeper.

Note that one GRQ is sent per logical endpoint; thus an MCU or a Gateway might send many. As the endpoint-Gatekeeper association is allowed to change over time, the endpoint may *multicast* the GRQ message and receive replies from several Gatekeepers, if it doesn't know which Gatekeeper the endpoint is associated with.

Endpoint registration (RRQ, RCF, RRJ, URQ, UCF, URJ)

The RRQ is a request from a terminal to a gatekeeper to register. If the gatekeeper responds with a RCF, the terminal shall use the responding gatekeeper for future calls. If the gatekeeper responds with a RRJ, the terminal must seek another gatekeeper to register with.

The URQ requests that the association between a terminal and a gatekeeper be broken. Note that "unregister" is bi-directional, i.e. a Gatekeeper can request a terminal to consider itself unregistered, and a terminal can inform a Gatekeeper that it is revoking a previous registration.

Admission (ARQ, ACF, ARJ)

The ARQ message requests that an endpoint be allowed access to the packet-based network by the gatekeeper, which either grants the request with an ACF or denies it with an ARJ.

Bandwidth change (BRQ, BCF, BRJ)

The BRQ message requests that an endpoint be granted a changed packet-based network bandwidth allocation by the gatekeeper, which either grants the request with a BCF or denies it with a BRJ.

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The gatekeeper may request that an endpoint raise or lower the bandwidth in use with a BRQ. If the request is to raise the rate, the endpoint may reply with either BRJ or BCF. If the request is for a lower rate, the endpoint shall reply with a BCF if the lower rate is supported, otherwise with BRJ.

Status (IRQ, IRR, IACK, INAK)

The IRQ is sent from a gatekeeper to a terminal requesting status information in the form of an IRR. The IRR may be also be sent by the terminal at an interval specified in the ACF message without the receipt of an IRQ from the gatekeeper.

When an unsolicited IRR is sent by an endpoint to a Gatekeeper of version 2 or higher, it may indicate that it wishes the Gatekeeper to acknowledge receipt of the IRR. The gatekeeper returns either an IACK (positive acknowledgement) or an INAK (negative acknowledgement) message.

Disengage (DRQ, DCF, DRJ)

If sent from an endpoint to a gatekeeper, the DRQ informs the gatekeeper that an endpoint is being dropped. If sent from a gatekeeper to an endpoint, the DRQ forces a call to be dropped. A DRQ shall not be refused. The DRQ is not sent between endpoints directly.

DRJ is sent by the gatekeeper if the endpoint is unregistered.

Endpoint location (LRQ, LCF, LRJ)

The LRQ requests that a gatekeeper provide address translation. The gatekeeper responds with an LCF containing the transport address of the destination, or rejects the request with LRJ.

Gateway resource availability (RAI, RAC)

The Resource Availability Indication (RAI) is a notification from a gateway to a gatekeeper of its current call capacity for each H-series protocol and data rate for that protocol. The gatekeeper responds with a Resource Availability Confirmation (RAC) upon receiving a RAI to acknowledge its reception.

Others (XRS, RIP)

The XRS message is sent whenever an H.323 endpoint receives a RAS message that it does not understand.

If an entity receives a request from a version 2 (or later) entity to which a response cannot be generated within a typical retry timeout period (see table below), it can send a RIP message specifying the period after which a response should have been generated.

H.225.0 – Recommended default timeout values

RAS message	Timeout value (sec)	Retry count
GRQ	5	2
RRQ (Note 1)	3	2
URQ	3	1
ARQ	5	2
BRQ	3	2

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IRQ	3	1
IRR (Note 2)	5	2
DRQ	3	2
LRQ	5	2
RAI	3	2
SCI	3	2
<p>NOTE 1 - The time-out value should be recalculated based upon both the time-to-live (which may be indicated by the Gatekeeper in the RCF message) and the desired number of retries.</p> <p>NOTE 2 – In cases where the gatekeeper is expected to reply to an unsolicited IRR with IACK or INAK, the timeout may occur if no reply to the IRR is received.</p>		

3.5 Media Gateway Control Protocols

Media Gateway Control Protocol (MGCP) [5] is used for controlling Telephony (Voice over IP) Gateways from external call control elements called media gateway controllers (MGC) or Call Agents. A telephony gateway is a network element that provides conversion between the audio signals carried on telephone circuits and data packets carried over the Internet or over other packet networks.

Logically, a telephony gateway consists of two components: a signalling gateway and a media gateway. The two components can be physically separated or combined in a single unit. MGCP is used for controlling media gateways only, and *not* signalling gateways, which are controlled from Call Agents using other protocols currently under development in IETF sigtran working group [42].

MGCP assumes a call control architecture where the call control "intelligence" is outside the gateways and handled by external call control elements. The MGCP assumes that these call control elements, or Call Agents, will synchronize with each other to send coherent commands to the gateways under their control. MGCP does not define a mechanism for synchronizing Call Agents. MGCP is, in essence, a master/slave protocol, where the gateways are expected to execute commands sent by the Call Agents.

MGCP implements the media gateway control interface as a set of transactions. The transactions are composed of a command and a mandatory response. There are 9 types of command:

- EndpointConfiguration
- CreateConnection
- ModifyConnection
- DeleteConnection
- NotificationRequest
- Notify
- AuditEndpoint
- AuditConnection
- RestartInProgress

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The first five commands are sent by the Call Agent to a gateway. The Notify command is sent by the gateway to the Call Agent. The gateway may also send a DeleteConnection. The Call Agent may send either of the Audit commands to the gateway. The Gateway may send a RestartInProgress command to the Call Agent.

All commands are composed of a Command header, optionally followed by a session description. All responses are composed of a Response header, optionally followed by a session description. Headers and session descriptions are encoded as a set of text lines, separated by a carriage return and line feed character (or, optionally, a single line-feed character). The headers are separated from the session description by an empty line.

MGCP uses a transaction identifier to correlate commands and responses. Transaction identifiers have values between 1 and 999999999. An MGCP entity must not reuse a transaction identifier sooner than 3 minutes after completion of the previous command in which the identifier was used.

The command header is composed of:

- A command line, identifying the requested action or verb, the transaction identifier, the endpoint towards which the action is requested, and the MGCP protocol version
- A set of parameter lines, composed of a parameter name followed by a parameter value

The command line is composed of:

- The name of the requested verb
- The identification of the transaction
- The name of the endpoint that should execute the command (in notifications or restarts, the name of the endpoint that is issuing the command)
- The protocol version

These four items are encoded as strings of printable ASCII characters and separated by white spaces, i.e., the ASCII space (0x20) or tabulation (0x09) characters. It is recommended to use exactly one ASCII space separator.

MGCP is only described in an informational RFC, and not put on the IETF standards track. The protocol will most likely be replaced by the MEGACO (MEdia GAteway COntrol) protocol / H.248 Recommendation [6], which is a joint effort between IETF megaco working group, and ITU-T study group 16.

The MEGACO set of transactions is functionally similar to MGCP, but have different names:

- Add
- Modify
- Subtract
- Move
- Notify
- AuditValue
- AuditCapabilities
- ServiceChange

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3.6 H.320 Signalling

3.6.1 Signalling Protocol Layers

The control and indication (C&I) signalling in H.320 is defined by several standards. The recommendation H.221 [20] defines the framing structure for transmission of encoding media and signals over ISDN circuits (see Figure 9). The standard reserves space inside the frames for control information and defines C&I signals. H.242 and H.230 defines more C&I signals and procedures for exchanging these within the signalling "channel" defined by H.221.

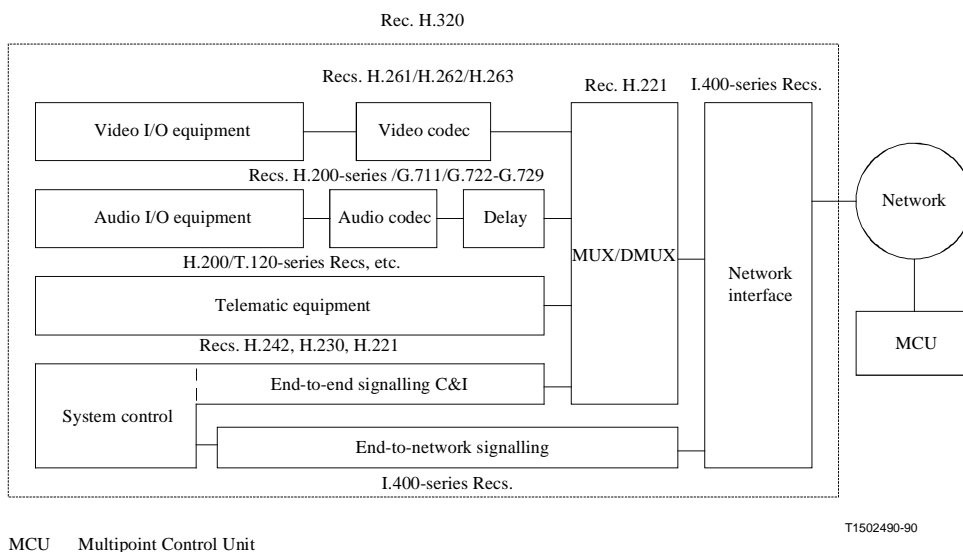


Figure 9: The H.320 signalling suite (fig. from [19])

3.6.2 Framing Structure

H.320 operates over ISDN B- or H_x -channels. A B-channel has a bandwidth of 64000 bit/s (64 kbit/s), and transmits one octet of 8 bits every 1/8000 seconds. H.221 treats H_x -channels as composed of a number of 64 kbit/s timeslots numbered 1 to n. The framing control is applied to each separate B-channel and to the first 64 kbit/s timeslot of each separate H_x -channel used by the terminal.

The bits in the octet are numbered 1 (first/leftmost) to 8 (last). One frame consists of 80 consecutive octets. The numbered bit positions within each octet are regarded as separate sub-channels. Thus, in the duration of a frame, each of the 8 sub-channels send 80 bits, 1 bit at a time, see Figure 10.

Sub-channels 1 to 7 are used for data (i.e. encoded audio). The first bits of subchannel 8 are used as the service channel. The 16 to 24 first bits (2 to 3 octets) in each frame of subchannel 8 are used for frame alignment signals (FAS) (bits 1-8), bandwidth allocation signals (BAS) (bits 9-16) and optionally encryption control signals (ECS) (bits 17-24). The rest of the channel (56 - 64 bits/frame) can be used for data.

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Bit number								
1	2	3	4	5	6	7	8 (SC)	
S	S	S	S	S	S	S	FAS	1 Octet number
u	u	u	u	u	u	u		:
b	b	b	b	b	b	b		8
-	-	-	-	-	-	-	BAS	9
c	c	c	c	c	c	c		:
h	h	h	h	h	h	h		16
a	a	a	a	a	a	a	ECS	17
n	n	n	n	n	n	n		:
n	n	n	n	n	n	n		24
e	e	e	e	e	e	e		25
l	l	l	l	l	l	l		.
#	#	#	#	#	#	#	#	.
1	2	3	4	5	6	7	8	80

Figure 10: Subchannels in one frame (fig. from [20]).

The FAS field is used for framing control, numbering and (optionally) CRC check of frames, sub-multiframes (= 2 consecutive frames) and multiframes (= 16 consecutive frames). The FAS field is also used for attaching a number to each of the framing control channels in the case that multiple B- or H₀-channels are used. It also carries a marking bit that can be used to signal loss of data due to temporary frame misalignment to the partner endpoint.

The C&I signals defined by the various standards in the H.320 signalling suite are in general transmitted in the BAS field(s) of a sub-multiframe. The BAS field of the first frame (even-numbered) in the sub-multiframe contains the signal, and an error correcting code is carried in the following (odd-numbered) frame. The duration of a sub-multiframe is 20msec.

The signals start with a 3-bit classifier. This leaves 32 different possible values for each class of signals. Several classes of signals contain more than 32 different signals, and some signals require numeric arguments like i.e. logical channel IDs. To facilitate this, many of the signals are signalled using the BAS field of several sub-multiframes. This is done in two ways. The so called Single Byte Extensions (SBEs) has a number of "escape codes" in a variable number of consecutive BAS fields before the signal code finally is placed in a single BAS-field (thus the name). Multiple Byte Extensions (MBEs) must be used for signals requiring numerical arguments. They start with a MBE escape-code in the BAS-field of the first sub-multiframe, followed by a byte-count, N, followed by BAS fields of N-1 sub-multiframes containing the signal and its parameters.

3.6.3 Call Establishment and Control

3.6.3.1 Phase A (unnamed in H.320): user location, admission and call setup

The functionality of this phase is combined with Phase B in the description in 3.6.3.2.

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3.6.3.2 Phase B (*sequence A* in H.320): capability exchange

Calls are made to phone numbers and established using the underlying network infrastructure. H.320-enabled telematic equipment will start transmitting properly framed A- or μ -law encoded sound (Mode 0 or 0F for "framed"), while looking for correct multiframe framing control in the bits received from the remote terminal. If no such framing can be found, the remote telephone is obviously an ordinary phone. In such a case, call establishment is finished by switching to unframed sound-only mode. If framing is found, *sequence A* proceeds with exchange of capabilities that apply to audiovisual terminals. The terminal initiating capability exchange will set a timer (min. 10 sec.) and transmit codes for its own capabilities. A terminal starting to receive capability codes shall immediately (switch to framed mode and) start transmitting its capabilities. If the initiating terminal has received a capability set before the timer times out, it can proceed to *sequence B*. An error conditions has occurred if correct framing was found but no capabilities received. In this case *sequence A* is restarted.

3.6.3.3 Phase C (*sequence B: mode switching* in H.320): establishment of A/V-comm.

Mode switching is performed using BAS command codes, each being effective from the beginning of the even frame following the sub-multiframe in which the code is first transmitted [23].

3.6.3.4 Phase D: call services

Call services include opening and closing of logical channels, mode switching and forced switching (of the other terminal) to mode 0F (G.711 sound only, framed transmission). Depending on the underlying network infrastructure, the call may be transferred to other terminals. This may force switching to mode 0F and restarting sequences A and B. ITU-T Recommendation H.243 [24] defines signals and procedures for establishing contact between more than two terminals (conferences).

3.6.3.5 Phase E: call termination

Not mentioned by the standards. Probably handled by the underlying network breaking the connection.

3.6.4 Comparison of H.320 Versions

ITU-T Recommendation H.320 from 5/99 supersedes the version from 7/97 and is used for this report.

3.7 H.323 Signalling

3.7.1 Signalling Protocol Layers

The signalling protocol layers of ITU-T Recommendation H.323 is shown in Figure 11 and Figure 12. Recommendation H.225 defines the interpretation, packetization and use of Q.931 messages, RAS messages and RTP/RTCP. It also describes how to make use of reliable and unreliable communication channels on packet networks in general and on TCP/IP networks in particular. Recommendation H.245 defines messages (in ASN.1) for establishment and co-ordination of multimedia conferences over network connections provided by H.225 and some procedures for exchanging these messages. H.323 defines entities that exchange H.225 and H.245 messages in order to provide multimedia conferencing, and also procedures for establishing contact between these entities.

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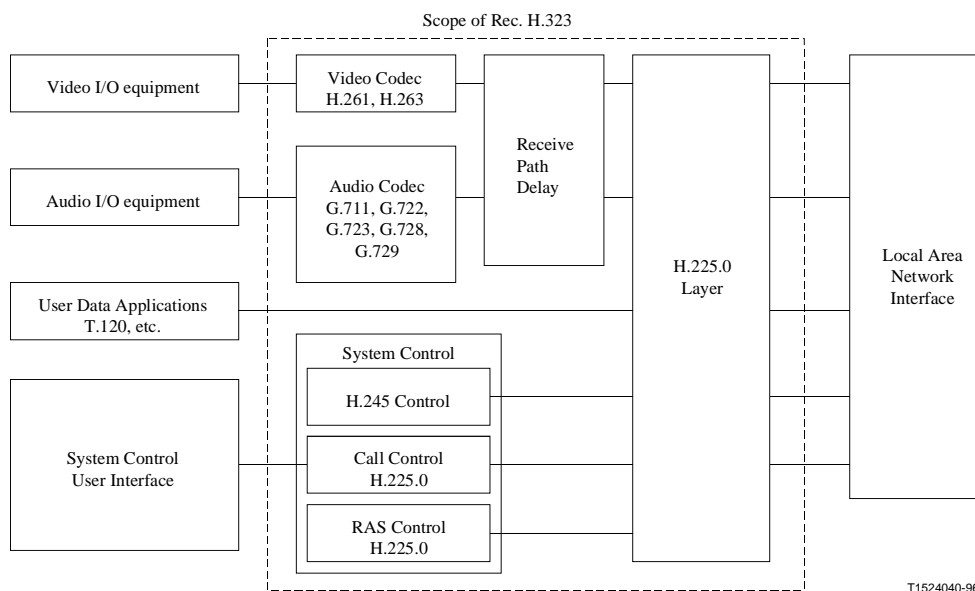


Figure 11: The focus of recommendation H.323 is mainly the box marked "System Control" (fig. from [17])

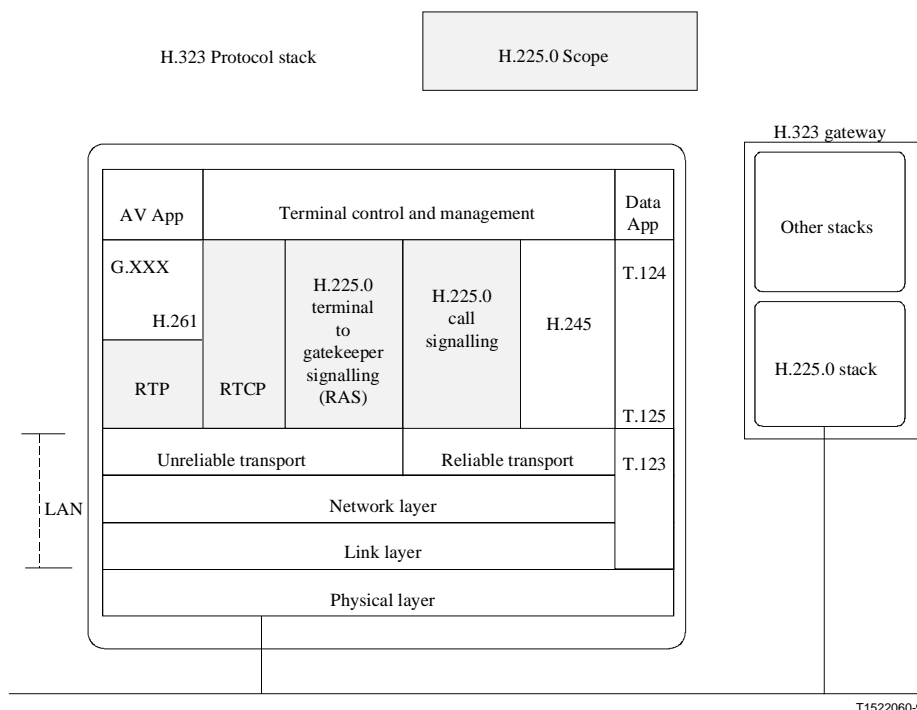


Figure 12: Detailed view of the H.323 protocol stack. Note that IP/UDP/TCP can be used as network and transport layers, not only LANs. (fig. from [4])

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The following signalling is used in H.323:

Signalling standard/ kind:	Signalling between terminal and:	Used for	Reliable channel ¹ ?	Dynamic transport address?
RAS	Gatekeeper	Registration, Admission control, Status, Location.	Unreliable	Well-known or dynamic
Call signalling (Q.931)	Gatekeeper ² or other endpoint (terminal, gateway or MCU)	Set up and tear down connections, indicate user alert etc.	Reliable	Dynamic or well-known
H.245	Gatekeeper ³ or other endpoint	Negotiate session parameters, media etc.	Reliable	Dynamic
RTP	Other endpoint	Wraps media packets.	Unreliable ⁴	Dynamic
RTCP	Other endpoint	Media transport quality information.	Unreliable ⁴	Dynamic
Data (T.120)	Other endpoint	Shared whiteboard and similar applications.	Reliable	Well-known or dynamic
1) Reliable channels use TCP on IP networks. Unreliable use UDP. 2) If gatekeeper routed call control is used (probably most cases) 3) Signalling is sent through the gatekeeper only if GK routed call control is used. 4) RTP/RTCP doesn't necessarily use IP/UDP if the IP stack is built on top of an end-to-end packet service like i.e. ATM.				

3.7.2 Call Establishment and Control

Calls are established by terminals and possibly assisted or controlled by Gatekeepers. A call may also have its origin outside the H.323 network through a gateway.

A H.323 session (call or conference) proceeds in 5 phases, labelled A to E:

- Phase A: Call setup.
- Phase B: Initial communication and capability exchange.
- Phase C: Establishment of audiovisual communication.
- Phase D: Call services.
- Phase E: Call termination.

3.7.2.1 Phase A: user location, admission and call setup

If a Gatekeeper (GK) is used, which normally is the case, every terminal has to register itself with the gatekeeper at power-up or user login. The registration message contains the transport level address of the terminal and a string identifying the user (e.g. his email address). Terminals can ask the GK for addresses if they don't know the transport address of the person (string-ID) the user is calling. A GK may ask other GKs for the address if it doesn't recognise the address itself, i.e. if the calling terminal is requesting the address of a user in a different zone (other network). The Gatekeepers are supposed to know the addresses of the other Gatekeepers. Otherwise a GK can broadcast the location request on the Gatekeepers' well-known multicast address. H.225.0[4] contains an informative appendix describing how DNS and URLs can be used to locate a remote gatekeeper if the terminal-id (callee-ID) is an email-like identifier.

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A calling terminal registered with a GK must always ask the GK for permission to contact another endpoint. This is performed by sending an Access Request (ARQ) on the GK's RAS channel. The GK will indicate in the reply (Access Confirm /Reject - ACF / ARJ) whether the call signalling should go directly from endpoint to endpoint or if the GK routed call control model is to be used. If the terminal was allowed to contact the named endpoint, a reliable call control channel is opened to the endpoint, eventually through the local gatekeeper. The calling terminal then sends a Q.931 *Setup*-message on the call control channel. If the called endpoint (callee) is registered with a GK it must ask its GK for permission to receive the call, using a RAS ARQ-message. If the callee's GK indicates use of GK routed call control, the call control channel has to be closed, and new channels opened via the callee's GK.

Figure 13 illustrates the case when the fact "that both GKs require GK routed call control" is discovered as late as possible. This method is quite time consuming as the opening and closing of reliable TCP channels (before *setup* and after *release complete*) takes 1.5 roundtrips. If the address of the called endpoint is located using location requests to the gatekeeper, the called endpoints gatekeeper (GK2) can force routing of calls through itself by inserting it's own address in the location reply. This saves several roundtrips.

Gatekeepers can also offer resource management services to the terminals, for instance bandwidth reservation. The terminals indicate the amount of bandwidth they need in the ARQ-messages. These parameters can be changed during Phase D *call services*.

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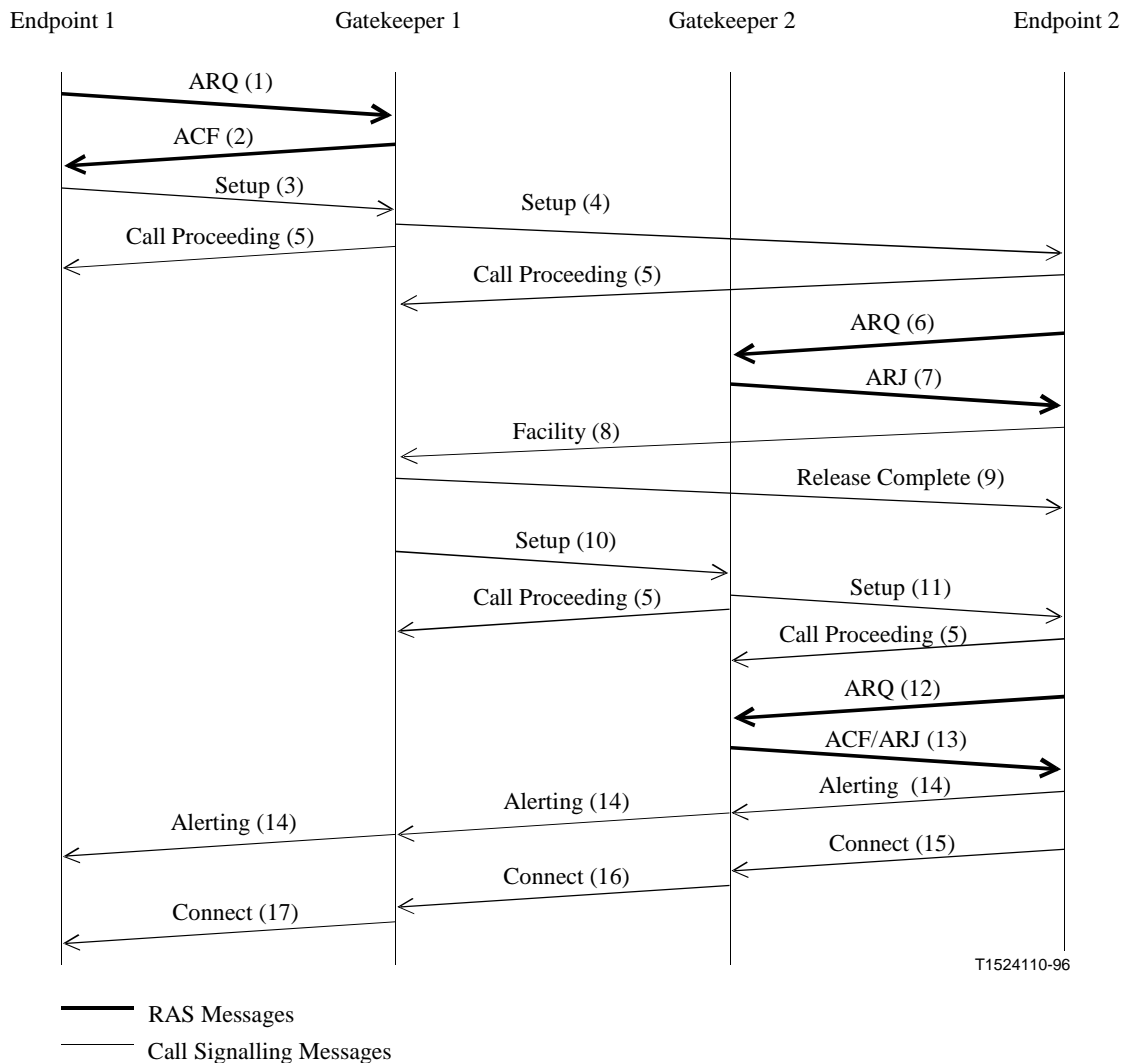


Figure 13: Call setup when both GKs require GK routed setup (fig from [17]).

3.7.2.2 Phase B: capability exchange

After a connection has been established, the terminals exchange capabilities for media formats and conferencing. After the exchange, it is clear to both endpoints what media formats the other endpoint can receive and transmit and who will be the Active MC (coordinator) of the conference/call. The default action is to set up a separate H.245 control channel for the exchange of these messages, but they can also be tunneled over the H.225 call control channel or exchanged in the *setup* message using the *fastStart* procedure. It is legal to set up a separate H.245 channel at any time after Phase A is finished, even if tunneled H.245 or *fastStart* has been used earlier in the call.

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3.7.2.3 Phase C: establishment of A/V-communications

After finishing Phase B, audio and video communication can be established using the *openLogicalChannel*-message (and its related responses) as defined in H.245. The media data is transferred using RTP over an unreliable channel. Media mode changes during the call are possible, as defined in H.245.

3.7.2.4 Phase D: call services

A number of services are available to H.323 endpoints and MCs. These include bandwidth (re-)negotiation with GKs, turning calls into conferences and making changes to the number of conference participants. The procedures for making changes to conferences vary depending on the conference mode (centralised, using MCU...), who's initiating the changes (coordinator or not), whether the initiator is a member of the conference or not, etc.

The recommendations H.450.x define procedures for supplementary services like call transfer, call hold and so on.

Changing media mode can be regarded as a call service.

3.7.2.5 Phase E: call termination

Calls are terminated by one of the endpoints sending the H.245 EndSessionCommand in the H.245 channel, the other endpoint replying the same command and the first endpoint sending Q.931 Release Complete on the Call control channel if that is still open. If the endpoints are registered with a gatekeeper, they have to send Disengage Request (DRQ) on the GKs RAS channel in the end.

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3.7.3 Comparison of H.323 Versions

The following table lists some of the more important changes between the H.323 versions.

Version	Major changes since previous	Lesser changes
2	<ul style="list-style-type: none"> Targeted at TCP/IP (the Internet) instead of LANs. New optional SIP-style "fast setup procedure" shortens startup delay considerably. Low bitrate operation. GK registrations can have finite life and may include alternative addresses. 	<ul style="list-style-type: none"> New Annexes B "<i>Procedures for Layered Codecs</i>" and C "<i>H.323 on ATM</i>" incorporated. Optional call setup over UDP. Lots and lots of clarifications and adjustments.
3	<ul style="list-style-type: none"> Alternative procedures for terminal registration with GK. Multiple calls over the same Call control channel allowed. Allows all audio and video formats that can be expressed using H.245 messages. Call signalling multiplexing over one logical channel. 	<ul style="list-style-type: none"> (Those listed as "major" aren't really that big). Remote device control possible using H.282. An endpoint may have multiple alias addresses.
4	<ul style="list-style-type: none"> Explicit support for separate physical entities for media gateways (MGs) and media gateway controllers (MGS) that are external to signalling and network (trunking) gateways. Gateways may thus be logically and/or physically decomposed, as described in H.GCP / H.248. Multiplexed media streams on one logical channel. New <i>additive registrations</i> allow adding information to an existing terminal registration with a GK. 	<ul style="list-style-type: none"> Corrections to errors assuming (not existent) identity between called digits and global E.164 numbers. Alternate Gatekeeper procedures. Usage information reporting

3.7.4 Known Problems with H.323 Version 2

Most of the changes made since version 2 addressed problems. More problems are known to exist, but cannot be easily solved without breaking backward compatibility. For instance, "Fast Setup" can potentially cause race conditions.

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3.8 Session Initiation Protocol (SIP)

3.8.1 Entities

The following entities are defined in SIP:

- User Agent Client:** The piece of software that initiates establishment of a session via SIP, for instance a web browser plug-in to handle clicks on sip:user@domain.dom -URLs. UACs negotiate session parameters and starts media tools on success.
- User Agent Server:** Piece of software running on the user's terminal looking/waiting for incoming SIP-invitations on this terminal. UASs notify the users of requests, negotiates session parameters and starts media tools.
- SIP-server:** Server software handling SIP-requests for (typically) an entire domain. Locates the current terminal of the user and redirects requests to the UAS on this/these terminal(s). Can operate in proxy- or redirect mode.
- Location Service:** Used by the SIP-server to locate a specific user on incoming requests.
- Registrar:** Receives REGISTER-messages from the user agent server on user login/logout and updates the location service's database.

3.8.2 Signalling Protocol Layers

The Session Initiation Protocol (SIP)[21] is text based and can be used over both TCP and UDP. It is mandatory for servers to support both, but clients can support only one of these. SIP is not dependent on any specific features of UDP or TCP, and could be used on top of any network offering a message based or reliable stream service (and global addressing of course). The message MTU should be sufficiently large to hold an entire SIP message, perhaps more than 1000 characters long.

SIP uses the protocol SDP [22], which also is text based, to negotiate session parameters for media formats and addresses. SIP messages use the same header/body structure as e-mail, HTTP, and other text based Internet standards. The SDP content is carried as the body of certain SIP messages.

SIP is a classic request-response protocol modelled after HTTP. Unlike HTTP (and like RTSP) the media content is transferred in separate connections, *external* to the requests/replies. The servers and clients also have a *state* connected to the state of the session.

SIP defines 6 request methods and 6 different classes of reply status codes. A number of header fields are defined. A request starts with a "request-line" containing the SIP method, the targeted URL and the SIP version. It is followed by several header fields and eventually a message body. A reply starts with a reply-line indicating the status of the request followed by header fields.

Method:	Meaning / use:	Body content:
INVITE	"Ring" the other endpoint/partner.	Media capabilities.
ACK	Acknowledge session establishment (positive response on Invite) for 3-way handshake session initiation.	Media capabilities if Invite didn't contain one
BYE	Terminate session ("hang up")	Empty
OPTIONS	Query about the other endpoints capabilities.	Suggested media

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		capabilities (optional).
CANCEL	Cancel ongoing request before final reply is sent, for instance to hang up if the user doesn't want to wait for reply any longer.	Empty
REGISTER	Send registration message to registrar to update the user location database.	Script for handling requests (optional).

Header type:	Used for:	Examples:
General header	Fields that apply to both requests and replies and describes general attributes of the session.	To, From, Call-ID, CSeq
Entity header	Meta-information about the body of the message.	Content-length, Content-type
Request header	Information that only makes sense in requests.	Accept-Language, Subject, Record-Route
Response header	Information that only makes sense in responses.	Warning, Accept-Encoding, Authorization

Status code class:	Used for:	Examples:
100 – informational	Preliminary info, request processing proceeds.	100 Trying, 180 Ringing
200 – successful	Request succeeded.	200 OK
300 – redirection	Further action by the client required to process request.	300 Multiple choices, 305 Use Proxy
400 – request failure	Unacceptable request.	400 Bad Request, 402 Payment Required, 403 Forbidden
500 – server failure	Apparently valid request could not be fulfilled by the server.	500 Internal Server Error, 504 Gateway Time-out, 505 SIP version not supported
600 – global failure	The request cannot be fulfilled by any server.	600 Busy Everywhere, 603 Decline

3.8.3 Session Description Protocol (SDP)

SDP session descriptions consist of several strictly ordered lines of text. Each line has the form <fieldname>=<value> where the fieldname is a single character and the format of the value depends upon the kind of field. The message consists of a set of lines describing the session in general followed by several lines describing each of the media included.

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Mandatory session fields are:

- v= <protocol version>
- o= <owner / creator of session, including network type & address >
- s= <session name (string) >
- c= <network type & address unless specified for each media >
- t= <at least one time specification of when the session is valid / active >

Optional session fields include:

- i= <informational text>
- u= <URI for additional information >
- e= / p= <email / phone number to originator >
- b= <bandwidth information>
- k= <encryption method> : <encryption key>

Mandatory fields for each medium are:

- m= <media> <port> <transport type> <format list>
- c= <network type & address. Optional if specified for session>

Optional fields include:

- a= <attribute> : <value> One media stream can have several attributes.
- i= <title>
- b= <bandwidth info>

IANA maintains a list of numbers identifying specific formats. Codec developers can obtain a registered number for their format. A set of numbers is reserved for experimental use. Developers using these numbers can utilise the "rtpmap" attribute to specify format and encoding details (e.g. sample rate). Other uses of the attribute line include the keywords "sendonly", "recvonly", "lang", "framerate" and "quality".

3.8.4 Call Establishment and Control

3.8.4.1 Phase A: user location, admission and call setup

Calls are established by SIP User Agent Clients (UAC) implemented, for instance, as web browser plugins registered to handle clicks on sip://user@domain-URIs. The UAC sends a SIP INVITE-request to the SIP-server of the domain identified by the URI. The address of this server is looked in DNS, preferably using the new DNS SRV service.

The SIP-server for the specified domain looks up the current location(s) of the specified user using the location service, with which the user has already registered. If the SIP server operates in proxy mode, it will set up a new session from itself to the user's terminal (similar to GK routed call control). In Redirect mode, it will return status 302 and the address of the terminal(s) the user is registered with, see Figure 14.

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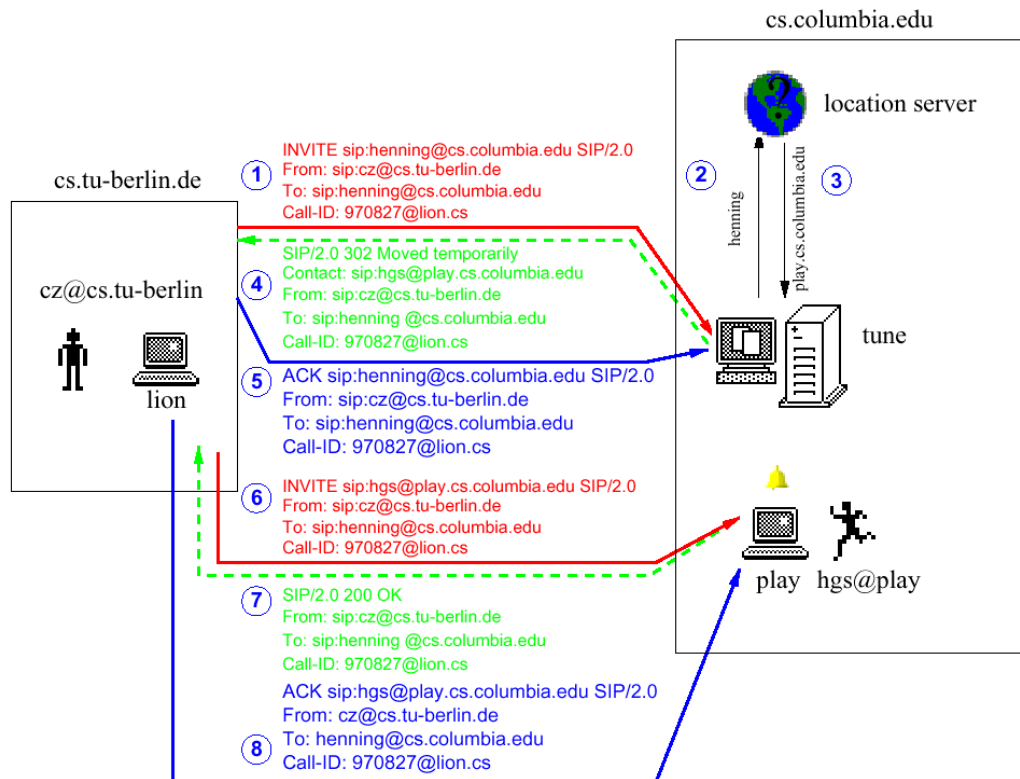


Figure 14: SIP call using redirecting server (fig from [21]).

3.8.4.2 Phase B: capability exchange

Capability exchange is done using SDP in the INVITE request and the 200 OK response. The initiator can, subsidiary, put his capabilities in the ACK message following the 200 OK. In any case, the SDP message will contain entries for each of the perception media the sender can receive simultaneously along with a (prioritised) list of acceptable compression formats for the medium and the transport address it should be sent to.

3.8.4.3 Phase C: establishment of A/V-communications

The media addresses expressed in the SDP messages are expected to be valid upon reception. If the initiator includes capabilities in the INVITE message (which is default behaviour), the receiver can send media data (e.g. "ringing") to the target address immediately even if the session isn't formally established until 200 OK has been sent. Note that controlling the media interchange is partly outside the scope of the SIP protocol. The purpose of SIP is to locate partners, exchange media capabilities and open/close logical sessions. The actual control of media tools is regarded as an implementations issue.

3.8.4.4 Phase D: call services

Call services aren't explicitly defined in SIP. Most of the functionality provided by H.323 is, however, supported. Group communication is supported for loosely connected groups, using Internet multicast for

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signalling and/or media exchange. New partners can be added by sending an INVITE containing the multicast address in a Via: header.

Making changes to the set of media exchanged is done by send a new INVITE with the same Call-ID as the existing call and the new set of media in the body of the message.

3.8.4.5 Phase E: call termination

Call termination is done by sending BYE to the partner and by stopping the media tools.

3.8.5 Comparison of SIP Versions

RFC 2543bis is the draft for SIP's transition from *proposed standard* to *draft standard* (IETF has 3 levels of standard maturity). The changes made since the draft standard are mainly clarifications. A few headers that are used in HTTP and Internet mail have been added, including *Content-Language*, *MIME-Version* and *Content-Disposition*.

3.9 H.323 versus SIP

The differences between H.323 and SIP are large, but not as large as they at first might seem. The tasks and set of services supported by the standards are essentially the same. The client functionality is essentially the same, but H.323 assumes a tighter integration between call signalling software and media exchange software. The Server and Registrar in SIP together have essentially the same functionality as the gatekeeper in H.323. In both protocols they handle name-to-terminal translation and user location registration. One obvious difference is that H.323 clients must "ask for permission" to send and receive calls. In SIP blocking of calls will have to be performed using traditional Internet firewalls to stop unwanted SIP requests. The lack of user name format standardisation in H.323 (any string is allowed) makes name resolution impossible in principle. We expect, however, that e-mail addresses will used everywhere, making DNS based username-to-remote-gatekeeper resolution possible via DNS. The fact that SIP is text based makes it easier to make and debug programs for handling the protocol messages than H.323's ASN.1 bit-patterns. It also makes the protocol easier to extend. The messages get bigger than with ASN.1, but the amount of signalling data is supposed to be insignificant compared to the audio and video data transmitted in the sessions anyway.

One major difference is the service creation process. In SIP, one user may be registered with several actual and virtual "locations" in terms of services, protocols and URLs. These registrations may have different priorities, making it possible for users to create their own intelligent network services on the server. There is also some work going on to create a standardized Call Processing Language for clients to further control server actions. In H.323 additional services are developed in separate standards, making the service creation process slow and cumbersome.

In general, SIP is more aligned with Internet standards and procedures, whereas H.323 is adapted to the Internet after the basic design was determined. Many of H323's design decisions do not align with Internet practices at all. For instance, using the gatekeeper to perform RSVP resource reservation on behalf of the clients, will of course only work if the client is on the same local area network as its gatekeeper. But then, associating a name with a location out of the office becomes impossible to combine with guaranteed quality calls. The standard also allows caller addresses that are impossible to solve without a (non-existent) global directory or global broadcast.

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The implementation complexity of both standards is significant, making it difficult to implement using low-level languages and/or special-purpose hardware. SIP was designed to bring telephony session establishment to computers and is designed for high-level languages from the beginning, whereas H.323's bit-based signalling protocols are more suited for low-level languages and embedded devices and less targeted at high-level programming. Since both implementation complexity and hardware demands suggest implementation on full-fledged computers with feature rich operating and runtime systems anyway, SIP does seem to be easier to implement than H.323. SIP does however have more loosely defined signalling semantics require some interoperability testing to make the software actually interoperable, not only "standards compliant".

3.10 Trends

The signaling protocols described in this chapter have been developed by different *standards organisations*.

ITU-T (International Telecommunication Union - Telecommunication standardization sector) is the organization behind the further development of standards such as H.323, H.225.0, H.245 which cover protocols such as RAS, and Q.931.

IETF (Internet Engineering Task Force) is the organization behind the further development of standards such as SIP, SDP, and MGCP. The latter is being replaced by a joint IETF/ITU-T effort under the name of Megaco/H.248.

The TIPHON (Telecommunications and Internet Protocol Harmonization Over Networks) project in ETSI (European Telecommunications Standards Institute) is working towards harmonization of the standards for Voice-over-IP and related multimedia services from the two organizations.

The *maturity* of the protocols is shown by what is implemented by vendors so far. For VoIP, products based on the H.323 protocols have been most prevalent as that standard is the oldest, already in its fourth version. However, more and more products supporting the SIP protocol are appearing. Especially at the SIP interoperability tests ("bake-offs"), when many big players show up with their yet-to-be-launched products.

When it comes Video-over-IP, the H.323 and proprietary solutions are prevalent. Some gateway vendors don't have a video solution yet, so this area is somewhat open so far.

Predicting the *future* is always difficult. There appears to be a market for both H.323 protocols and the SIP protocol, so both will most likely share the market for VoIP. Gateways for interoperability will then be a major concern. In media gateway control the Megaco/H.248 protocol is expected to be dominant.

A leading VoIP analyst, Jeff Pulver, who publishes the Pulver Report [105], has labeled year 2000 as the "year of transition" with the whole telecommunication industry really adopting an IP strategy. He expects this trend to continue in 2001, and predicts next year to be the year when VoIP becomes the establishment. Year 2000 was also the year "SIP happened" on a commercial basis.

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3.11 Applicability to a GEO Satellite Environment

This study will consider the general applicability of the terrestrial based signalling protocols described in this chapter to a GEO satellite environment taking into consideration the characteristics of the satellite environment.

It appears that the main characteristics that influence the applicability of the VoIP terrestrial based signalling protocols are satellite delay and the resulting latency, jitter, and high BER. These will be addressed in more detail in chapter 6 below.

4. TRANSMISSION PROTOCOLS

This chapter is concerned with the major transmission protocols used for multimedia services in IP based networks and those used on the PSTN/ISDN side of an H.323 Gateway.

In the first section, transmission of user media in ISDN is described, followed by a section on the most popular LAN technologies, including the various Ethernet specifications.

The third section is a comprehensive introduction to fundamental packet network protocols in the TCP/IP protocol suite, including RTP/RTCP for real-time data transmission.

The next section describes the most important, ITU-T standardized audio and video codecs used for IP based multimedia services. For each media protocol it is indicated how the encoded data is transported in RTP payload.

Finally, there is a section on different Fax over IP protocols.

4.1 ISDN

As mentioned in section 2.1, the B or H channel is used for transmission of user media in ISDN. The services provided to the users over ISDN are defined in three types of services grouped as bearer, teleservice, and supplementary services. Bearer services don't require intervention by the elements of the network after the connection is established. Teleservices may require some intervention. Supplementary services enhance the capabilities of both bearer and teleservices[32].

The bearer service, that defines what type of information is carried on the B channels, is passed on the D channel at call setup in the bearer service information element. The network layer protocol for ISDN defines bearer capabilities in the information element in ITU-T Q.931. The bearer capability information element is specified for packet-based multimedia communication systems in ITU-T H.225.

The procedures of ITU-T Q.931 for circuit mode connection setup are followed also for H.323 and H.225. Although "bearer" is being signalled, no actual "B-channels" of the ISDN type exist on the packet-based network side. Successful completion of the "call" results in an end-to-end reliable channel supporting H.245 messaging. Actually "bearer" setup is done using H.245[18].

Analog traffic, such as voice, Group 3 fax, and modem data must be converted to a digital format before it can be carried over an ISDN line. This conversion is done by an adapter know as an ISDN Terminal Adapter

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or TA. By using a TA, you can connect your present analog telephone, fax machine or modem to an ISDN line.

When using ISDN equipment, which interfaces directly to your computer (e.g., from your PC's COM port to the data port of an ISDN Terminal Adapter), special software or firmware is needed in order to connect end-to-end with an analog modem or fax machine. The software/firmware "emulates" the analog modulated waveforms of modems and fax machines [30].

4.2 LAN

A local area network (LAN) is usually confined to a single geographic area, such as a single building or a campus. LANs can be small, just linking two or three computers, or large linking hundreds of computers.

Various standards have been developed for networking computers. Ethernet is the most popular of these standards. Other LAN types include LocalTalk, Token Ring, Fast Ethernet, Fiber Distributed Data Interface (FDDI) and Asynchronous Transfer Mode (ATM). There are five major types of wiring used in Ethernet networks: thickwire for 10Base5 networks, thin coax for 10Base2 networks, fiber optic, unshielded twisted pair (UTP) for 10BaseT and 100BaseT, and wireless.

Multimedia, computer graphics and video conferencing all need fast networks. The two most important types of high-speed LANs are Fast Ethernet (including Gigabit Ethernet) and ATM LAN.

4.2.1 Ethernet

Classical 10 Mbit/s Ethernet, standardized as IEEE 802.3, uses carrier sensed multiple access with collision detection (CSMA/CD) as its method of medium access control. Before attempting to transmit a message, an end station determines whether or not another end station is transmitting a message on the media. If the media is available, the end station transmits the message. If not, the end station delays its transmission until the other end station has finished sending.

All end stations on the network, including the transmitting end stations, detect the collision and ignore the message. Each end station that wants to transmit waits a random amount of time and then attempts to transmit again. The random transmission delays reduce the probability that the end stations will transmit simultaneously again.

When a node on an Ethernet LAN (end station) transmits data, every end station on the LAN receives the data. Each end station checks each data unit to see whether the destination address matches its own address. If the addresses match, the end station accepts and processes the packet. If they do not match, it disregards the packet.

4.2.2 Fast Ethernet

Fast Ethernet refers to a set of specifications developed by the IEEE 802.3 committee to provide a low-cost, Ethernet-compatible LAN operating at 100 Mbit/s. Two approaches have been worked out, the 100BASE-T and the 100VG-AnyLAN. These two differ in the medium access control mechanisms.

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The 100BASE-T keeps the original CSMA/CD medium access control mechanism and was standardized as IEEE 802.3u. The 100VG-AnyLAN approach created a completely new medium access control mechanism, the "demand priority" mechanism. This approach was standardized as IEEE 802.12.

Fast Ethernet transceivers support switching and in many cases full duplex services.

Gigabit Ethernet is an even faster Ethernet technology developed by the IEEE 802.3 committee. The technology operates at data rates of 1000 Mbit/s. Two approaches have been worked out depending on wiring used. 1000BASE-T twisted pair is standardized as IEEE 802.3ab, whereas 1000BASE-X fiber optic is standardized as IEEE 802.3x.

Gigabit Ethernet has not been fully embraced and employed in LANs, as the effective data rate (throughput) has proven to be lower than expected.

4.2.3 ATM LAN

ATM protocols are designed to handle isochronous (time critical) data such as telephony (audio) and video, in addition to more conventional inter-computer data communications. Using ATM, information to be sent is segmented into fixed length cells, transported to and reassembled at the destination.

The ATM cell has a fixed length of 53 bytes. Being fixed length allows the information to be transported in a predictable manner. This predictability accommodates different traffic types on the same network. The cell is broken into two main sections, the header and the payload. The payload (48 bytes) is the portion that carries the actual information – voice, data or video. The header (5 bytes) is the addressing mechanism.

ATM operates at standardized data rates of 155 and 622 Mbit/s. For each connection a virtual circuit is set up between the sender and destination.

Since LAN protocols have been developed in a shared media environment, however, they do not work well with circuits. Supporting LAN protocols with ATM is expensive and complicated.

4.3 Packet Network Protocols

Traditionally, telecommunication packet network protocols has operated in connection-oriented mode. This is based on setting up virtual circuits across the network with enough resources allocated to each connection. Routing functionality in such a mode is mainly relaying packets along the already determined routed path. The approach is inefficient when it comes to utilizing network resources. This is how e.g. ATM works.

A different approach is packet networks operating in connectionless-oriented mode. In this mode, no resources are set aside or circuits set up, but each packet is treated independently and a routing decision is made in every router from the sender to destination. Thus, packets belonging together are not necessarily transported along the same way. The approach is highly flexible when it comes to utilizing network resources, and robust as long as efficient queuing mechanisms are employed in the routers. However, it makes network QoS very difficult to achieve as guarantees are impossible to make. This is how IP-based packet networks work.

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In an IP network environment the underlying network protocol is of course IP (Internet Protocol). Depending on the reliability of the communication, the transport protocol is either UDP (User Datagram Protocol), or TCP (Transmission Control Protocol).

In the H.323 protocol suite, H.225.0 call signalling and H.245 signalling use TCP as transport protocol. However, H.225.0 RAS signalling use UDP for unreliable transport, as do the SIP and MGCP protocols.

All media transmission protocols in this document use UDP as the underlying transport protocol. However, to provide end-to-end transport functions suitable for real-time data (coded audio, video), two intermediate transport protocols are necessary: RTP (Real-time Transport Protocol) and its control protocol, RTCP.

Below, all these fundamental packet network protocols are described (from [16]).

4.3.1 IP (Internet Protocol)

The Internet Protocol [7] is the routing layer datagram service of the TCP/IP suite. All other protocols within the TCP/IP suite use IP to route packets from host to host. The IP packet header contains routing information and control information associated with datagram delivery.

The IP header structure is as follows:

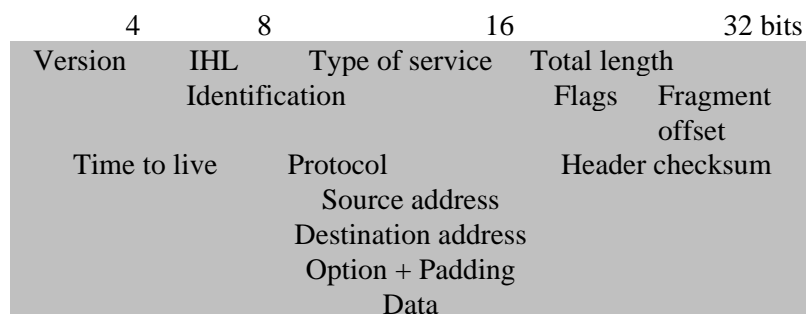


Figure 15: IP header structure

Version

Version field indicates the format of the Internet header.

IHL

Internet header length is the length of the Internet header in 32-bit words. Points to the beginning of the data. The minimum value for a correct header is 5.

Type of service (ToS)

Indicates the quality of service desired. Networks may offer service precedence, meaning that they accept traffic only above a certain precedence at times of high load. There is a three-way trade-off between low delay, high reliability and high throughput.

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Bits 0-2: Precedence

- 111 Network control.
- 110 Internetwork control.
- 101 CRITIC/ECP.
- 100 Flash override.
- 011 Flash.
- 010 Immediate.
- 001 Priority.
- 000 Routine.

Bit 3: Delay

- 0 Normal delay.
- 1 Low delay.

Bit 4: Throughput

- 0 Normal throughput.
- 1 High throughput.

Bit 5: Reliability

- 0 Normal reliability.
- 1 High reliability.

Bits 6-7: Reserved for future use.

Historically, the ToS field has never really been used in commercial routers for setting QoS priorities as was originally intended. The interpretation of the field has therefore recently changed with the development of the Differentiated Service architecture. The name of the field is now DS field. The exact usage of all the bits is still under standardization within the IETF diffserv working group.

Total length

Length of the datagram measured in bytes, including the Internet header and data. This field allows the length of a datagram to be up to 65,535 bytes, although such long datagrams are impractical for most hosts and networks. All hosts must be prepared to accept datagrams of up to 576 bytes, regardless of whether they arrive whole or in fragments. It is recommended that hosts send datagrams larger than 576 bytes only if the destination is prepared to accept the larger datagrams.

Identification

Identifying value assigned by the sender to aid in assembling the fragments of a datagram.

Flags

3 bits. Control flags:

Bit 0 is reserved and must be zero.

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Bit 1: Don't fragment bit:

- 0 May fragment.
- 1 Don't fragment.

Bit 2: More fragments bit:

- 0 Last fragment.
- 1 More fragments.

Fragment offset

13 bits. Indicates where this fragment belongs in the datagram. The fragment offset is measured in units of 8 bytes (64 bits). The first fragment has offset zero.

Time to live

Indicates the maximum time the datagram is allowed to remain in the Internet system. If this field contains the value zero, the datagram must be destroyed. This field is modified in Internet header processing. The time is measured in units of seconds. However, since every module that processes a datagram must decrease the TTL by at least one (even if it processes the datagram in less than 1 second), the TTL must be thought of only as an upper limit on the time a datagram may exist. The intention is to cause undeliverable datagrams to be discarded and to bound the maximum datagram lifetime.

Protocol

Indicates the next level protocol used in the data portion of the Internet datagram.

Header checksum

A checksum on the header only. Since some header fields change, e.g., Time To Live, this is recomputed and verified at each point that the Internet header is processed.

Source address / destination address

32 bits each. A distinction is made between names, addresses and routes. A *name* indicates an object to be sought. An *address* indicates the location of the object. A *route* indicates how to arrive at the object. The Internet protocol deals primarily with addresses. It is the task of higher level protocols (such as host-to-host or application) to make the mapping from names to addresses. The IP modules map Internet addresses to local net addresses. It is the task of lower level procedures (such as local net or gateways) to make the mapping from local net addresses to routes.

Options

Options may or may not appear in datagrams. They must be implemented by all IP modules (host and gateways). What is optional is their transmission in any particular datagram, not their implementation. In some environments, the security option may be required in all datagrams.

The option field is variable in length. There may be zero or more options. There are two possible formats for an option:

- A single octet of option type.
- An option type octet, an option length octet and the actual option data octets.

The length octet includes the option type octet and the actual option data octets.

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The option type octet has 3 fields:

1 bit: Copied flag. Indicates that this option is copied into all fragments during fragmentation:

- 0 Copied.
- 1 Not copied.

2 bits: Option class

- 0 Control.
- 1 Reserved for future use.
- 2 Debugging and measurement.
- 3 Reserved for future use.

5 bits: Option number.

Data

IP data or higher layer protocol header.

4.3.1.1 IPv6

IP version 6 [8] is a new version of the Internet Protocol based on the current IPv4. IPv4 and IPv6 are demultiplexed at the link layer. For example, IPv6 packets are carried over Ethernet with the content type 86DD (hexadecimal) instead of IPv4's 0800.

IPv6 increases the IP address size from 32 bits to 128 bits, to support more levels of addressing hierarchy, a much greater number of addressable nodes and simpler auto-configuration of addresses. Scalability of multicast addresses is introduced. A new type of address called an *anycast address* is also defined, to send a packet to any one of a group of nodes.

Improved support for extensions and options - IPv6 options are placed in separate headers that are located between the IPv6 header and the transport layer header. Changes in the way IP header options are encoded allow more efficient forwarding, less stringent limits on the length of options, and greater flexibility for introducing new options in the future. The extension headers are Hop-by-Hop Option, Routing (Type 0), Fragment, Destination Option, Authentication, and Encapsulation Payload.

Flow labelling capability - A new capability has been added to enable the labelling of packets belonging to particular traffic flows for which the sender requests special handling, such as non-default Quality of Service or real-time service. The exact usage of such labelling has yet to be standardized by IETF. Thus, contrary to widespread belief, IPv6 offers little improvement in terms of QoS specification.

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The IPv6 header structure is as follows:

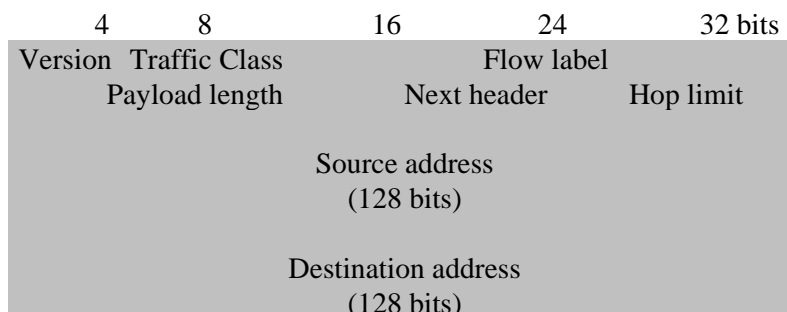


Figure 16: IPv6 header structure

Version

Internet Protocol Version number (IPv6 is 6).

Traffic Class

Enables a source to identify the desired delivery traffic class (priority) of the packets. Equivalent to the ToS field in IPv4 with the same interpretation.

Flow label

Used by a source to label those packets for which it requests special handling by the IPv6 router. The flow is uniquely identified by the combination of a source address and a non-zero flow label.

Payload length

Length of payload (in octets).

Next header

Identifies the type of header immediately following the IPv6 header.

Hop limit

8-bit integer that is decremented (by one) by each node that forwards the packet. The packet is discarded if the Hop Limit is decremented to zero.

Source address

128-bit address of the originator of the packet.

Destination address

128-bit address of the intended recipient of the packet.

4.3.2 UDP (User Datagram Protocol)

User Datagram Protocol [9] provides a simple, but unreliable message service for transaction-oriented services. Each UDP header carries both a source port identifier and destination port identifier, allowing high-level protocols to target specific applications and services among hosts.

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The UDP header structure is shown as follows:

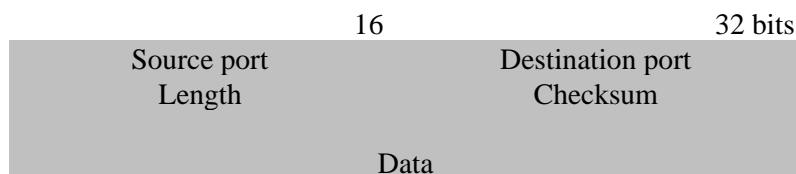


Figure 17: UDP header structure

Source port

Source port is an optional field. When used, it indicates the port of the sending process and may be assumed to be the port to which a reply should be addressed in the absence of any other information. If not used, a value of zero is inserted.

Destination port

Destination port has a meaning within the context of a particular Internet destination address.

Length

The length in octets of this user datagram, including this header and the data. The minimum value of the length is eight.

Checksum

The 16-bit one's complement of the one's complement sum of a pseudo header of information from the IP header, the UDP header and the data, padded with zero octets at the end (if necessary) to make a multiple of two octets.

Data

UDP data field.

4.3.3 TCP (Transmission Control Protocol)

Transmission Control Protocol [10] provides a reliable stream delivery and virtual connection service to applications through the use of sequenced acknowledgement with retransmission of packets when necessary.

TCP operates with a windowing mechanism to dynamically adjust its transmission rate. The window is defined as the number of outstanding packets the sender can transmit before an acknowledgement is required. At connection start-up the window size is one, and a slow-start algorithm is used which exponentially increases the size by one for each acknowledgment received until a threshold is reached. Then, a congestion avoidance phase takes over, which increases the window size more slowly (by 1/size) until a packet is lost. This is taken as an indication of network congestion. The window size is reduced (halved), the lost packet retransmitted and the congestion avoidance process repeated. This fast retransmit/fast recover algorithm allows TCP to continue transmission at a relatively high rate, which improves the overall protocol performance.

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The TCP header structure is as follows:

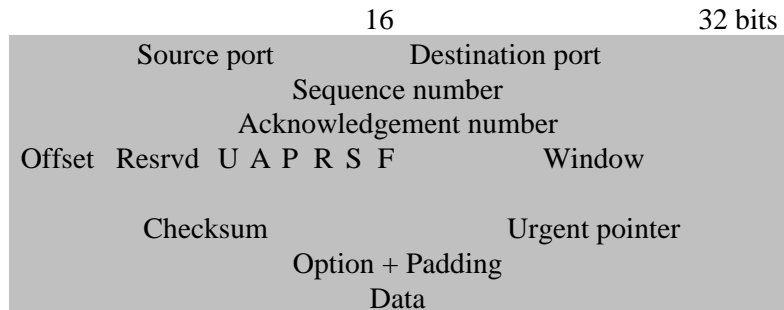


Figure 18: TCP header structure

Source port

Source port number.

Destination port

Destination port number.

Sequence number

The sequence number of the first data octet in this segment (except when SYN is present). If SYN is present, the sequence number is the initial sequence number (ISN) and the first data octet is ISN+1.

Acknowledgment number

If the ACK control bit is set, this field contains the value of the next sequence number, which the sender of the segment is expecting to receive. Once a connection is established, this value is always sent.

Data offset

4 bits. The number of 32-bit words in the TCP header, which indicates where the data begins. The TCP header (even one including options) has a length which is an integral number of 32 bits.

Reserved

6 bits. Reserved for future use. Must be zero.

Control bits

6 bits. The control bits may be (from right to left):

- U (URG) Urgent pointer field significant.
- A (ACK) Acknowledgement field significant.
- P (PSH) Push function.
- R (RST) Reset the connection.
- S (SYN) Synchronize sequence numbers.
- F (FIN) No more data from sender.

Window

16 bits. The number of data octets which the sender of this segment is willing to accept, beginning with the octet indicated in the acknowledgement field.

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Checksum

16 bits. The checksum field is the 16 bit one's complement of the one's complement sum of all 16-bit words in the header and text. If a segment contains an odd number of header and text octets to be checksummed, the last octet is padded on the right with zeros to form a 16-bit word for checksum purposes. The pad is not transmitted as part of the segment. While computing the checksum, the checksum field itself is replaced with zeros.

Urgent Pointer

16 bits. This field communicates the current value of the urgent pointer as a positive offset from the sequence number in this segment. The urgent pointer points to the sequence number of the octet following the urgent data. This field can only be interpreted in segments for which the URG control bit has been set.

Options

Options may be transmitted at the end of the TCP header and always have a length which is a multiple of 8 bits. All options are included in the checksum. An option may begin on any octet boundary.

There are two possible formats for an option:

A single octet of option type.

An octet of option type, an octet of option length, and the actual option data octets.

The option length includes the option type and option length, as well as the option data octets.

The list of options may be shorter than that designated by the data offset field because the contents of the header beyond the End-of-Option option must be header padding, i.e. zero.

A TCP must implement all options.

Data

TCP data or higher layer protocol.

4.3.4 RTP (Real-time Transport Protocol)

Real-time Transport Protocol [11] provides end-to-end network transport functions suitable for applications transmitting real-time data such as audio, video or simulation data, over multicast or unicast network services. RTP does not address resource reservation and does not guarantee quality-of-service for real-time services. The data transport is augmented by a control protocol (RTCP) to allow monitoring of the data delivery in a manner scalable to large multicast networks, and to provide minimal control and identification functionality. RTP and RTCP are designed to be independent of the underlying transport and network layers. The protocol supports the use of RTP-level translators and mixers.

The intention of RTP is to provide a set of simple delivery services such as payload type identification, sequence numbering, timestamping and delivery monitoring for data with real-time characteristics. 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. Note that RTP itself does not provide any

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mechanism to ensure timely delivery, 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 is reliable and delivers packets in sequence.

The format of the RTP Fixed Header Fields is as follows:

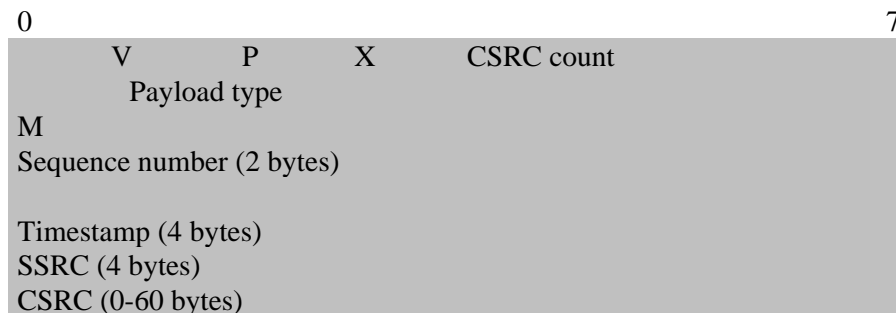


Figure 19: RTP fixed header fields

V

Version. Identifies the RTP version.

P

Padding. When set, the packet contains one or more additional padding octets at the end which are not part of the payload.

X

Extension bit. When set, the fixed header is followed by exactly one header extension, with a defined format.

CSRC count

Contains the number of CSRC identifiers that follow the fixed header.

M

Marker. The interpretation of the marker is defined by a profile. It is intended to allow significant events such as frame boundaries to be marked in the packet stream.

Payload type

Identifies the format of the RTP payload and determines its interpretation by the application. A profile specifies a default static mapping of payload type codes to payload formats. Additional payload type codes may be defined dynamically through non-RTP means.

Sequence number

Increments by one for each RTP data packet sent, and may be used by the receiver to detect packet loss and to restore packet sequence.

Timestamp

Reflects the sampling instant of the first octet in the RTP data packet. The sampling instant must be derived

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from a clock that increments monotonically and linearly in time to allow synchronization and jitter calculations. The resolution of the clock must be sufficient for the desired synchronization accuracy and for measuring packet arrival jitter (one tick per video frame is typically not sufficient).

SSRC

Identifies the synchronization source. This identifier is chosen randomly, with the intent that no two synchronization sources within the same RTP session will have the same SSRC identifier.

CSRC

Contributing source identifiers list. Identifies the contributing sources for the payload contained in this packet.

4.3.5 RTCP (Real-time Transport Control Protocol)

Real-time Transport Control Protocol [11] is based on the periodic transmission of control packets to all participants in the session, using the same distribution mechanism as the data packets. The underlying protocol must provide multiplexing of the data and control packets, for example using separate port numbers with UDP.

RTCP is used for the monitoring of QoS and for conveying information about participants in an ongoing session. Two forms of quality feedback are provided depending upon whether the receiver is also a sender. The sender report (SR) is issued if a site has sent any data packets during the interval since issuing the last report or the previous one, otherwise the receiver report (RR) is issued.

The RTCP header structure is as follows:

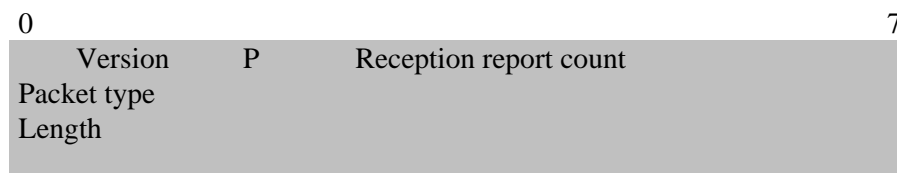


Figure 20: RTCP header structure

Version

Identifies the RTP version which is the same in RTCP packets as in RTP data packets.

P

Padding. When set, this RTCP packet contains some additional padding octets at the end which are not part of the control information. The last octet of the padding is a count of how many padding octets should be ignored. Some encryption algorithms with fixed block sizes may need padding. In a compound RTCP packet, padding should only be required on the last individual packet because the compound packet is encrypted as a whole.

Reception report count

The number of reception report blocks contained in this packet. A value of zero is valid.

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Packet type

Identifies the form of the RTCP packet. The constant 200 indicates a sender report, whereas 201 indicates a receiver report. Other values are also defined.

Length

The length of this RTCP packet in 32-bit words minus one, including the header and any padding. (The offset of one makes zero a valid length and avoids a possible infinite loop in scanning a compound RTCP packet, while counting 32-bit words avoids a validity check for a multiple of 4.)

4.4 Media protocols

4.4.1 G.711 Audio Codec

The ITU-T standard for encoding telephony audio on a 64 kbit/s channel is G.711 [2]. The standard uses a pulse code modulation (PCM) scheme operating at an 8 kHz sample rate, with 8 bits per sample. According to the Nyquist theorem, which states that a signal must be sampled at twice its highest frequency component, G.711 can encode frequencies between 0 and 4 kHz. Two different variants of G.711 exist, A-law and μ -law. A-law is the standard for international circuits between different countries. It is additionally used within European countries, whereas μ -law is predominantly used within USA.

Each of these encoding schemes is designed in a roughly logarithmic fashion. Lower signal values are encoded using more bits; higher signal values require fewer bits. This ensures that low amplitude signals will be well represented, while maintaining enough range to encode high amplitudes.

However, the actual encoding doesn't use logarithmic functions. The input range is broken into segments, each segment using a different interval between decision values. Most segments contain 16 intervals, and the interval size doubles from segment to segment. Figure 21 shows three segments with four intervals in each.

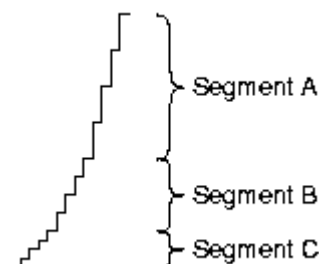


Figure 21: G.711 example encoding scheme

Both encodings are symmetrical around zero. μ -law uses 8 segments of 16 intervals each in each of the positive and negative directions, starting with a interval size of 2 in segment 1, and increasing to an interval size of 256 in segment 8. A-law uses 7 segments. The smallest segment, using an interval size of 2, is twice the size of the others (32 intervals). The remaining six segments have 16 intervals each, with the interval size increasing to 128 in segment 7. Thus, A-law is skewed towards representing smaller signals with greater fidelity.

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An illustration of the A-law encoding of positive input values is given below.

TABLE 1a/G.711
A-law, positive input values

1 Segment number	2 Number of intervals \times interval size	3 Value at segment end points	4 Decision value number n	5 Decision value x_n (see Note 1)	6 Character signal before inversion of the even bits								7 Quantized value (value at decoder output) y_n	8 Decoder output value number
					Bit number 1 2 3 4 5 6 7 8									
7	16×128	4096	(128)	(4096)	-----								4032	128
		127	3968	1 1 1 1 1 1 1 1										
6	16×64	2048	113	2176	1 1 1 1 0 0 0 0								2112	113
		112	2048	(see Note 2)										
5	16×32	1024	97	1088	1 1 1 0 0 0 0 0								1056	97
		96	1024	(see Note 2)										
4	16×16	512	81	544	1 1 0 1 0 0 0 0								528	81
		80	512	(see Note 2)										
3	16×8	256	65	272	1 1 0 0 0 0 0 0								264	65
		64	256	(see Note 2)										
2	16×4	128	49	136	1 0 1 1 0 0 0 0								132	49
		48	128	(see Note 2)										
1	32×2	64	33	68	1 0 1 0 0 0 0 0								66	33
		32	64	(see Note 2)										
			1	2	1 0 0 0 0 0 0 0								1	1
			0	0										

Note 1 - 4096 normalized value units correspond to $T_{max} = 3.14$ dBm0.
 Note 2 - The character signals are obtained by inverting the even bits of the signals of column 6. Before this inversion, the character signal corresponding to positive input values between two successive decision values numbered n and $n + 1$ (see column 4) is $(128 + n)$ expressed as a binary number.
 Note 3 - The value at the decoder output is $y_n = \frac{x_n - 1 + x_{n+1}}{2}$ for $n = 1, \dots, 127, 128$.
 Note 4 - x_{128} is a virtual decision value.
 Note 5 - In Tables 1/G.711 and 2/G.711 the values of the uniform code are given in columns 3, 5 and 7.

4.4.1.1 RTP payload for G.711

With a sampling rate of 8 kHz, each 8-bit sample represents 0.125 ms audio.

How to transport G.711 encoded samples in RTP payload is described in [12]. The payload type for A-law is 8, and corresponding encoding name PCMA. For μ -law, the payload type is 0, and corresponding encoding name PCMU. It is not specified how many (more than one) samples are normally transported in each RTP packet.

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4.4.2 G.723.1 Audio Codec

G.723.1 [3] is the ITU-T standard for compressing the speech or other audio signal component of multimedia services at a very low bit rate. The coder has two bit rates associated with it, 5.3 and 6.3 kbit/s. The higher bit rate has greater quality. The lower bit rate gives good quality and provides system designers with additional flexibility. Both rates are a mandatory part of the encoder and decoder. It is possible to switch between the two rates at any main frame boundary (every 240 samples).

The G.723.1 coder is designed to operate with a digital signal obtained by first performing telephone bandwidth filtering (G.712) of the analogue input, then sampling at 8000 Hz and then converting to 16-bit linear PCM for the input to the encoder. The output of the decoder should be converted back to analogue by similar means. Other input/output characteristics, such as those specified by G.711 for 64 kbit/s PCM data, should be converted to 16-bit linear PCM before encoding or from 16-bit linear PCM to the appropriate format after decoding.

The coder is based on the principles of linear prediction analysis-by-synthesis coding and attempts to minimize a perceptually weighted error signal. The encoder operates on blocks (*frames*) of 240 samples each. This is equal to 30 ms at an 8 kHz sampling rate. Each block is first high pass filtered to remove the DC component and then divided into four subframes of 60 samples each. For every subframe, a 10th order Linear Prediction Coder (LPC) filter is computed using the unprocessed input signal. The LPC filter for the last subframe is quantized using a Predictive Split Vector Quantizer (PSVQ). The unquantized LPC coefficients are used to construct the short-term perceptual weighting filter, which is used to filter the entire frame and to obtain the perceptually weighted speech signal.

For every two subframes (120 samples), the open loop pitch period, L_{OL} , is computed using the weighted speech signal. This pitch estimation is performed on blocks of 120 samples. The pitch period is searched in the range from 18 to 142 samples.

From this point the speech is processed on a 60 samples per subframe basis.

Using the estimated pitch period computed previously, a harmonic noise-shaping filter is constructed. The combination of the LPC synthesis filter, the formant perceptual weighting filter, and the harmonic noise-shaping filter is used to create an impulse response. The impulse response is then used for further computations.

Using the pitch period estimation, L_{OL} , and the impulse response, a closed loop pitch predictor is computed. A fifth order pitch predictor is used. The pitch period is computed as a small differential value around the open loop pitch estimate. The contribution of the pitch predictor is then subtracted from the initial target vector. Both the pitch period and the differential value are transmitted to the decoder.

Finally the non-periodic component of the excitation is approximated. For the high bit rate, Multi-pulse Maximum Likelihood Quantization (MP-MLQ) excitation is used, and for the low bit rate, an algebraic-code-excitation (ACELP) is used.

The decoder operation is also performed on a frame-by-frame basis. First the quantized LPC indices are decoded, and then the decoder constructs the LPC synthesis filter. For every subframe, both the adaptive codebook excitation and fixed codebook excitation are decoded and input to the synthesis filter. The adaptive postfilter consists of a formant and a forward-backward pitch postfilter. The excitation signal is

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input to the pitch postfilter, which in turn is input to the synthesis filter whose output is input to the formant postfilter. A gain scaling unit maintains the energy at the input level of the formant postfilter.

4.4.2.1 RTP payload for G.723.1

With a sampling rate of 8 kHz, each frame (240 samples) represents 30 ms audio. For the low bit rate codec (5.3 kbit/s), an encoded frame requires 158 bits. For the high bit rate codec (6.3 kbit/s), an encoded frame requires 189 bits. Additionally, 2 control bits are needed for each frame, to indicate whether the high (0) or low (1) rate is used, and to indicate whether the current frame is active speech (0) or non-speech (1). Thus, 20 octets are required for the low bit rate codec, whereas 24 octets (including 1 unused bit) are required for the high bit rate codec.

Illustrations of the bit allocation and octet bit packing for the 6.3 kbit/s coding algorithm are given below.

One frame (30 ms) bit allocation of the 6.3 kbit/s coding algorithm (2 control bits not included)

Parameters coded	Subframe 0	Subframe 1	Subframe 2	Subframe 3	Total
LPC indices					24
Adaptive codebook lags	7	2	7	2	18
All the gains combined	12	12	12	12	48
Pulse positions	20	18	20	18	73 (Note)
Pulse signs	6	5	6	5	22
Grid index	1	1	1	1	4
Total:					189

NOTE – By using the fact that the number of codewords in the fixed codebook is not a power of 2, 3 additional bits are saved by combining the 4 MSB of each pulse position index into a single 13-bit word.

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Octet bit packing for the high bit rate codec

Transmitted octets	PARx By, ...
1	LPC_B5...LPC_B0, VADFLAG_B0, RATEFLAG_B0
2	LPC_B13...LPC_B6
3	LPC_B21...LPC_B14
4	ACL0_B5...ACL0_B0, LPC_B23, LPC_B22
5	ACL2_B4...ACL2_B0, ACL1_B1, ACL1_B0, ACL0_B6
6	GAIN0_B3...GAIN0_B0, ACL3_B1, ACL3_B0, ACL2_B6, ACL2_B5
7	GAIN0_B11...GAIN0_B4
8	GAIN1_B7...GAIN1_B0
9	GAIN2_B3...GAIN2_B0, GAIN1_B11...GAIN1_B8
10	GAIN2_B11...GAIN2_B4
11	GAIN3_B7...GAIN3_B0
12	GRID3_B0, GRID2_B0, GRID1_B0, GRID0_B0, GAIN3_B11...GAIN3_B8
13	MSBPOS_B6...MSBPOS_B0, UB
14	POS0_B1, POS0_B0, MSBPOS_B12...MSBPOS_B7
15	POS0_B9...POS0_B2
16	POS1_B2, POS1_B0, POS0_B15...POS0_B10
17	POS1_B10...POS1_B3
18	POS2_B3...POS2_B0, POS1_B13...POS1_B11
19	POS2_B11...POS2_B4
20	POS3_B3...POS3_B0, POS2_B15...POS2_B12
21	POS3_B11...POS3_B4
22	PSIG0_B5...PSIG0_B0, POS3_B13, POS3_B12
23	PSIG2_B2...PSIG2_B0, PSIG1_B4...PSIG1_B0
24	PSIG3_B4...PSIG3_B0, PSIG2_B5...PSIG2_B3

How to transport G.723.1 encoded frames in RTP payload is *not* described in any IETF standard document. However, an authoritative list of RTP parameters is maintained at IANA [14] (now ICANN [15]). The payload type for G.723.1 is 4, and corresponding encoding name G723. It is not specified how many (probably only one) frames are normally transported in each RTP packet.

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4.4.3 G.729A Audio Codec

G.729A [46] is the ITU-T standard for coding of speech signals at 8 kbit/s using Conjugate-Structure Algebraic-Code-Excited Linear-Prediction (CS-ACELP). The coder is a reduced complexity version of the full G.729 speech codec [45]. The two versions are bit stream interoperable, but the performance of the reduced complexity codec may not be as good as the full version. G.729A has been developed for multimedia simultaneous voice and data applications, although the use of the codec is not limited to these applications.

The G729A coder is designed to operate with a digital signal obtained by first performing telephone bandwidth filtering (G.712) of the analogue input signal, then sampling it at 8000 Hz, followed by conversion to 16-bit linear PCM for the input to the encoder. The output of the decoder should be converted back to an analogue signal by similar means. Other input/output characteristics, such as those specified by G.711 for 64 kbit/s PCM data, should be converted to 16-bit linear PCM before encoding, or from 16-bit linear PCM to the appropriate format after decoding.

Generally, the CS-ACELP coder is based on the Code-Excited Linear-Prediction (CELP) coding model. The coder operates on speech frames of 10 ms corresponding to 80 samples at a sampling rate of 8000 samples per second. It has a look-ahead of 5 ms. For every 10 ms frame, the speech signal is analysed to extract the parameters of the CELP model (linear-prediction filter coefficients, adaptive and fixed-codebook indices and gains). These parameters are encoded and transmitted. At the decoder, these parameters are used to retrieve the excitation and synthesis filter parameters. The speech is reconstructed by filtering this excitation through the short-term synthesis filter. The short-term synthesis filter is based on a 10th order Linear Prediction (LP) filter. The long-term, or pitch synthesis filter is implemented using the so-called adaptive-codebook approach. After computing the reconstructed speech, it is further enhanced by a postfilter.

The general description of the G.729A coding/decoding algorithm above is similar to that of the full version. The major algorithmic changes are concerned with simplification of several filtering steps in the reduced version.

4.4.3.1 RTP payload for G.723A

With a sampling rate of 8 kHz, each frame (80 samples) represents 10 ms audio. An encoded frame requires 80 bits. The bit allocation is illustrated below.

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One frame (10 ms) bit allocation of the 8 kbit/s CS-ACELP algorithm

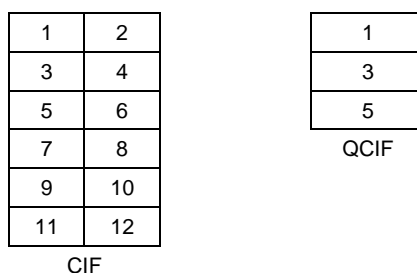
Parameter	Codeword	Subframe 1	Subframe 2	Total per frame
Line spectrum pairs	<i>L0, L1, L2, L3</i>			18
Adaptive-codebook delay	<i>P1, P2</i>	8	5	13
Pitch-delay parity	<i>P0</i>	1		1
Fixed-codebook index	<i>C1, C2</i>	13	13	26
Fixed-codebook sign	<i>S1, S2</i>	4	4	8
Codebook gains (stage 1)	<i>GA1, GA2</i>	3	3	6
Codebook gains (stage 2)	<i>GB1, GB2</i>	4	4	8
Total				80

How to transport G.729A encoded frames in RTP payload is not described in any IETF standard document. However, in the authoritative list of RTP parameters maintained at ICANN, the payload type for G.729A is 18, and corresponding encoding name G729. It is not specified how many (probably only one) frames are normally transported in each RTP packet.

4.4.4 H.261 Video Codec

H.261 [1] is the ITU-T standard for video coding and decoding methods for the moving picture component of audiovisual services at the rates of $p \times 64$ kbit/s, where p is in the range 1 to 30. Two picture scanning formats are specified, CIF (352 x 288 pixels) and QCIF (176 x 144 pixels). Only QCIF is mandatory.

The H.261 coding is organized as a hierarchy of groupings. The video stream is composed of a sequence of pictures or frames, which are organized as a set of Groups of Blocks (GOB). The arrangement of GOBs for the two picture formats is illustrated below. CIF consists of 12 GOBs, whereas QCIF consists of only 3.



Each GOB holds a set of 3 lines of 11 macro blocks (MB). Each MB carries information on a group of 16x16 pixels: luminance information is specified for 4 blocks of 8x8 pixels, while chrominance information is given by two "red" and "blue" colour difference components at a resolution of only 8x8 pixels.

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This grouping is used to specify information at each level of the hierarchy:

- At the frame level, one specifies information such as the delay from the previous frame, the image format, and various indicators.
- At the GOB level, one specifies the GOB number and the default quantifier that will be used for the MBs.
- At the MB level, one specifies which blocks are present and which did not change, and optionally a quantifier and motion vectors.

Blocks, which have changed, are encoded by computing the discrete cosine transform (DCT) of their coefficients, which are then quantized and Huffman encoded (Variable Length Codes).

The H.261 Huffman encoding includes a special "GOB start" pattern, composed of 15 zeroes followed by a single 1, that cannot be imitated by any other code words. This pattern is included at the beginning of each GOB header (and also at the beginning of each frame header) to mark the separation between two GOBs, and is in fact used as an indicator that the current GOB is terminated. The encoding also includes a stuffing pattern composed of seven zeroes followed by four ones. That stuffing pattern can only be entered between the encoding of MBs, or just before the GOB separator.

H.261 knows two types of coding macro blocks, intraframe and interframe. In the case of intraframe coding, no advantage is taken of the redundancy between frames. Beyond this, H.261 tries to make use of temporal redundancies by means of motion-compensated prediction. If the motion estimation is successful and the prediction error is below a certain threshold, interframe coding is applied for the macro blocks.

4.4.4.1 RTP payload for H.261

How to transport an H.261 encoded bitstream in RTP payload is described in [13]. The payload type for H.261 is 31, and corresponding encoding name H261. The clock rate is 90 kHz, which is a multiple of the natural H.261 frame rate. Finally, the marker bit in the RTP header is set to one in the last packet of a video frame, and otherwise, must be zero.

All the bits resulting from the Huffman encoding are included in the payload, but directly transmitting the result over an unreliable stream of UDP datagrams would have poor error resistance characteristics. Also, because of the interdependent hierarchical structure of H.261 bit stream, and the sometimes too large size of a picture or even one GOB by itself to fit in a single packet, the MB is taken as the unit of fragmentation. Packets must start and end on a MB boundary; i.e. a MB cannot be split across multiple packets. Multiple MBs may be carried in a single packet when they will fit within the maximal packet size allowed. To allow each packet to be processed independently for efficient resynchronization in the presence of packet losses, some state information from the frame header and GOB header is carried with each packet to allow the MBs in that packet to be decoded. Thus, the H.261 data consists of a H.261 header, in addition to the bitstream itself.

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The H.261 header structure is as follows:

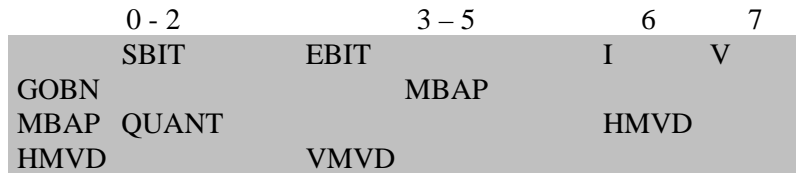


Figure 22: H.261 header structure

SBIT

Start bit. (3 bits) Number of most significant bits that are to be ignored in the first data octet.

EBIT

End bit. (3 bits) Number of least significant bits that are to be ignored in the last data octet.

I

INTRA-frame encoded data field. (1 bit) Set to 1 if this stream contains only INTRA-frame coded blocks and to 0 if this stream may or may not contain INTRA-frame coded blocks. The sense of this bit may not change during the course of the RTP session.

V

Motion Vector flag. (1 bit) Set to 0 if motion vectors are not used in this stream and to 1 if motion vectors may or may not be used in this stream. The sense of this bit may not change during the course of the session.

GOBN

GOB number. (4 bits) Encodes the GOB number in effect at the start of the packet. Set to 0 if the packet begins with a GOB header.

MBAP

Macro block address predictor. (5 bits) Encodes the macro block address predictor (i.e., the last MBA encoded in the previous packet). This predictor ranges from 0-32 (to predict the valid MBAs 1-33). Because the bit stream cannot be fragmented between a GOB header and MB 1, the predictor at the start of the packet can never be 0. Therefore, the range is 1-32, which is biased by -1 to fit in 5 bits. Set to 0 if the packet begins with a GOB header.

QUANT

Quantizer field. (5 bits) Shows the Quantizer value (MQANT or GQUANT) in effect prior to the start of this packet. Set to 0 if the packet begins with a GOB header.

HMVD

Horizontal motion vector data field. (5 bits) Represents the reference horizontal motion vector data (MVD). Set to 0 if V flag is 0, if the packet begins with a GOB header, or when the MTYPE of the last MB encoded in the previous packet was not MC. HMVD is encoded as a 2's complement number and '10000' corresponding to the value -16 is forbidden (motion vector fields range from +/-15).

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VMVD

Vertical motion vector data (VMVD). (5 bits) Reference vertical motion vector data (MVD). Set to 0 if V flag is 0, if the packet begins with a GOB header, or when the MTYPE of the last MB encoded in the previous packet was not MC. VMVD is encoded as a 2's complement number and '10000', corresponding to the value -16, is forbidden (motion vector fields range from +/-15).

4.4.5 H.263 Video Codec

H.263 is the ITU-T standard for video coding and decoding methods for the moving components of audio-visual services at low bit rates. A first version [40] was approved in 1996, whereas version 2 [41], also named H.263+, followed two years later. H.263+ is backward compatible with H.263 version 1.

Image Resolution

H.263 supports five resolutions. In addition to QCIF and CIF that were supported by H.261 there is sub-QCIF, 4CIF, and 16CIF. Details about the 5 standardized picture formats are summarized in the table below.

Number of pixels per line and number of lines for each of the standardized H.263 picture formats

Picture format	Number of pixels for luminance (dx)	Number of lines for luminance (dy)	Number of pixels for chrominance (dx/2)	Number of lines for chrominance (dy/2)
Sub-QCIF	128	96	64	48
QCIF	176	144	88	72
CIF	352	288	176	144
4CIF	704	576	352	288
16CIF	1408	1152	704	576

Source coding algorithms

As for H.261, a hybrid of inter-picture prediction to utilize temporal redundancy and transform coding of the remaining signal to reduce spatial redundancy is adopted. The decoder has motion compensation capability, allowing optional incorporation of this technique in the encoder. Half pixel precision is used for the motion compensation, as opposed to H.261 where full pixel precision and a loopfilter are used. Variable length coding (such as Huffman coding) is used for the symbols to be transmitted.

In addition to the core H.263 coding algorithm, 16 negotiable coding options can be used. All these options can be used together, or separately. Coding performance is significantly improved when these coding options are utilized. Of particular importance is the scalability option, which allows for the decoding of a video sequence at more than one quality level. Additional supplemental information may also be included in the bitstream for enhanced display capability and for external usage. A forward error correction method for application to the resulting video bitstream is provided for use when necessary.

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Resynchronization

In H.263 a GOB is defined as one or more rows of macroblocks (MBs). At the start of a new GOB, information called a GOB header is placed within the bitstream. This header information contains a GOB start code, which is different from a picture start code, and allows the decoder to locate this GOB. Furthermore, the GOB header contains information which allows the decoding process to be restarted (i.e., resynchronize the decoder to the bitstream and reset all predictively coded data).

Bit rate

The transmission clock is provided externally. The video bitrate may be variable. In recommendation H.263 no constraints on the video bitrate are given. Constraints will be given by the terminal or the network.

Signalling

H.323 entities that want to transmit H.263 video streams have to send an H.245 **OpenLogicalChannel** message specifying the payload format.

4.4.5.1 RTP payload for H.263

How to transport an H.263 (version 1) encoded bitstream in RTP is described in [35]. The payload type for H.263 is 34, the corresponding encoding name is H263 and the clock rate is 90 kHz. As for H.261, the marker bit in the RTP header is set to one when the current packet carries the end of current frame, and is zero otherwise.

For H.263+ the encapsulation in RTP is described in [36]. Contrary to H.263, the payload type for H.263+ is a dynamically defined number, e.g. through a conference control protocol, in the range 96-127, with a corresponding encoding name of H263-1998. The clock rate is still 90 kHz though. The use of the marker bit is also the same.

The output of the H.263 encoder is packetized directly. The H.263 bitstream is carried in the RTP payload without modification, including the picture start code, the picture header, and any fixed length codes and variable length codes.

Each RTP packet carries only one H.263 video packet (fragment). However, the fragment size is variable depending on the modes used. The RTP fixed header is followed by the H.263 payload header, which is followed by the H.263 payload.

Three formats (mode A, mode B and mode C) are defined for the H.263 (version 1) payload header.

Mode A allows fragmentation at GOB boundaries. A payload header of four octets is present before the actual compressed H.263 bitstream.

In mode B an H.263 bitstream is fragmented at MB (macroblock) boundaries. Mode B packets are intended for a GOB whose size is larger than the maximum octet size allowed in the underlying protocol, thus making it impossible to fit one or more complete GOBs in a packet. This mode can only be used without the PB-frames option. In mode B an eight octets payload header is used.

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In mode C an H.263 bitstream is fragmented at MB boundaries of P-frames with the PB-frames option. Mode C is intended for those GOBs whose sizes are larger than the maximum packet size allowed in the underlying protocol when PB-frames option is used. A payload header of twelve octets is used.

Of the three modes, use of mode A is strongly recommended whenever possible. Both because of its simplicity and because it simplifies error recovery in the presence of packet loss due to fragmentation at GOB and not MB boundaries.

For H.263+ a different header format is used. The format can also be used with H.263 version 1, as it is only a subset of the H.263+ syntax, and new implementations are recommended to do so. The H.263+ payload header consists of a 16 bit basic header, which may be followed by two optional fields. An 8 bit field for Video Redundancy Coding (VRC) information, and/or a variable length extra picture header.

The H.263+ basic header is structured as follows:



Figure 23: H.263+ basic header structure

RR

Reserved bits (5 bits). Shall be zero.

P

1 bit. Indicates the picture start or a picture segment (GOB/Slice) start or a video sequence end (EOS or EOSBS).

V

1 bit. Indicates the presence of an 8 bit field containing information for Video Redundancy Coding (VRC), which follows immediately after the initial 16 bits of the payload header if present.

PLEN

Length in bytes of the extra picture header (6 bits). If no extra picture header is attached, PLEN is 0. If PLEN > 0, the extra picture header is attached immediately following the rest of the payload header.

PEBIT

3 bits. Indicates the number of bits that shall be ignored in the last byte of the picture header. If PLEN is not zero, the ignored bits shall be the least significant bits of the byte. If PLEN is zero, then PEBIT shall also be zero.

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4.5 Fax Over IP Protocols

4.5.1 T.37 Store-and-forward Fax

T.37 [38] is the ITU-T standard for enabling fax to be transferred using Internet e-mail as a store and forward system. The protocol uses Simple Mail Transfer Protocol (SMTP) addressing to send fax messages. Store-and-Forward fax is thus more of an integration of fax and e-mail. In most cases the fax transmission will go step-by-step from one server to another. The fax eventually ends up on a mail server, and then as a unified message in an inbox of an e-mail client, in an IP-enabled fax terminal, or, after conversion, in a Group 3 (G3) fax terminal.

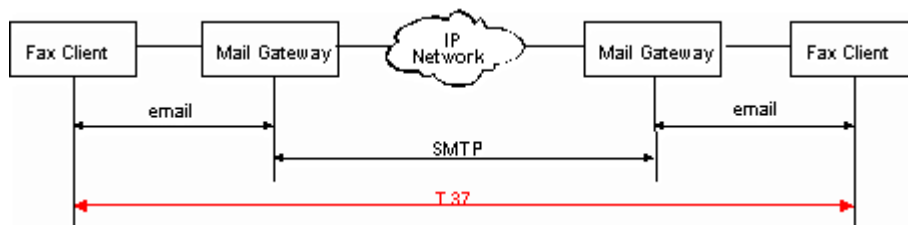


Figure 24: Store-and Forward Fax

The addresses of sending and receiving terminals are specified by the use of Internet mail addresses. Addressing information for the PSTN and ISDN conforms to ITU-T Recommendation E.164.

Store-and-forward facsimile may operate in one of two modes:

- Simple Mode – Supports the transfer of image data. Capability exchange and confirmation of receipt **are not** required for Simple Mode.
- Extended Mode (Full Mode) – Supports the transfer of image data. Capability exchange and confirmation **are** required for Full Mode. This mode is for further study.

In order to connect to traditional fax terminals, the terminals must have either two interfaces (T.37 and traditional G3), or they need access to a gateway or “on-ramp/off-ramp” function.

An off-ramp is a system capable of receiving e-mail with MIME attachments (Multipurpose Internet Mail Extension for fax) from a T.37 fax machine or a PC, that is destined for a G3 fax machine on the PSTN. The off-ramp detaches the MIME attachment, extracts the phone number of the destination fax from the address information of the e-mail, and sends the fax over the PSTN to the G3 device. The MIME attachment is defined in the T.37 specification to be a TIFF (Tagged Image File Format) document.

On-ramps accept calls from traditional G3 fax machines and sends the resulting fax images to the appropriate Internet recipient as a MIME attachment to an e-mail.

4.5.2 T.38 Real-time Fax

T.38 [39] is the ITU-T standard for enabling fax to be transferred in real-time between two Group 3 and/or IP-enabled fax terminals over the Internet. As an example, Figure 25 shows the protocols in use when a Group 3 Fax (F2) is connected with an Internet Aware Fax terminal (IAF1). The emitting gateway (G2)

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demodulates the T.30 transmission received from the calling terminal. The T.30 facsimile control and image data is transferred as an octet stream structure (T.38) using the IFP (Internet Facsimile Protocol) packets over a transport protocol (TCP or UDP).

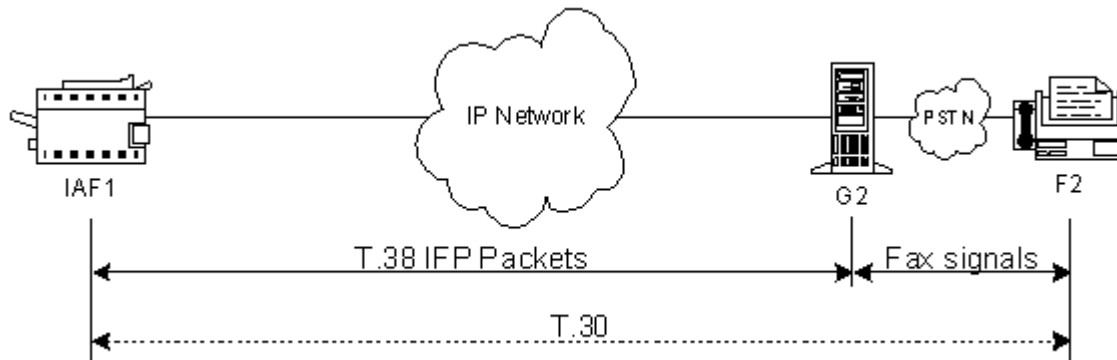


Figure 25: Real-time Fax

The following signals are generated or handled locally between the gateway and the G3FE: CNG (calling tone), CED (answer tone) and, in one mode, TCF (T.30 Training Check). The gateways may indicate the detection of the tonal signals CNG and CED so that the other gateway or IAF terminal can generate them.

The receiving gateway or IAF terminal decodes the transferred information and establishes communication with the called facsimile terminal using normal T.30 procedures. The receiving gateway forwards all relevant responses from the called terminal to the gateway.

The emitting gateway may optionally ignore NSF (Non-Standard Facilities), NCS and NSS, take appropriate action or pass the information to the receiving gateway. The receiving gateway may optionally ignore NSF and NSS or take appropriate action including passing the information to the receiving G3FE.

4.5.2.1 IFP packets

IFP (Internet Facsimile Protocol) operates over TCP or UDP using a port determined during call setup. All communication between IFP peers is done using packets, identified as IFP packets.

An IFP packet element consists of two parts: TYPE and DATA.

The TYPE element describes the function of, and optionally, the data of the packet. There are two different TYPES:

- T30_INDICATOR
Used by the gateways to indicate the detection of signalling information such as CED, HDLC preamble flags, and modem modulation training. It is sent by the receiving gateway to the emitting gateway, and by the emitting gateway to the receiving gateway. The use of this message is optional for TCP implementations and mandatory for UDP implementations. A peer may send this message in order to notify its peer about upcoming messages.

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- **T30_DATA TYPE**
Used to indicate that the packet contains actual data in the DATA element and what modulation was used to carry the data. The T30_DATA TYPE is used to indicate both HDLC control data as well as any Phase C data.

The DATA-Field element contains the T.30 HDLC control data and the Phase C image data. It contains the data from the PSTN connections and some indication of the data format. The DATA element is a structure containing one or more Fields. Each Field has two parts: the first part indicates the Field-Type, the second part contains the Field-Data. The structure follows the international standard X.680 Abstract Syntax Notation One (ASN.1). This ensures gateway interoperability and allows for backward compatibility.

The IFP packets are combined with the appropriate headers for TCP and UDP as shown in Figure 26 and 27. In Figure 27, the UDPTL header represents the additional header information required for error control over UDP.

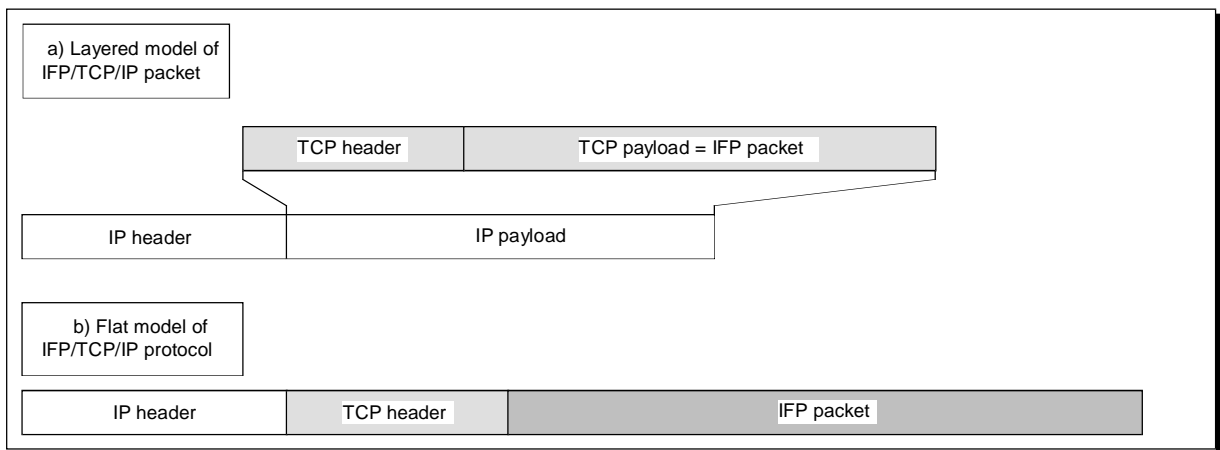
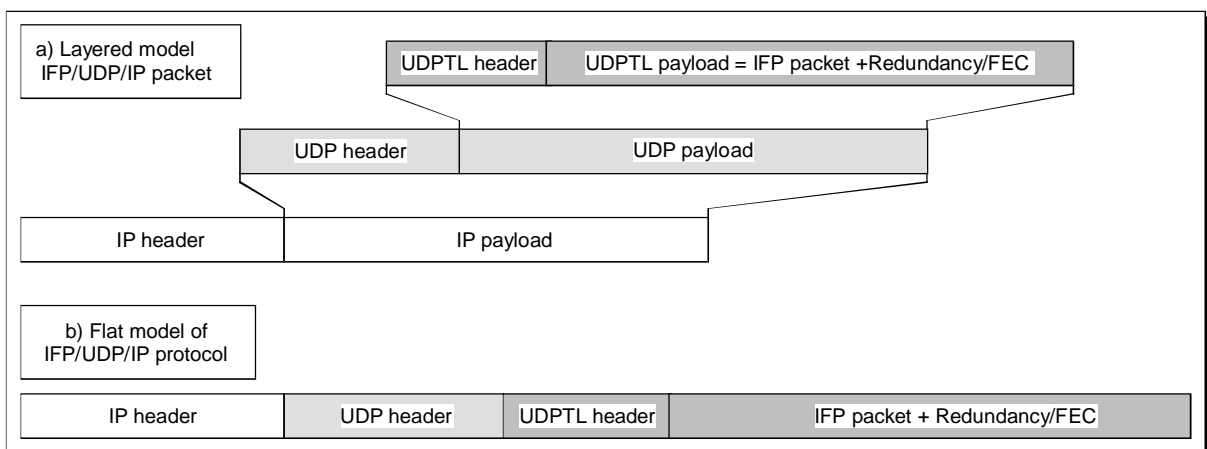


Figure 26: High-level IFP/TCP packet structure



T0827900-98/d05

Figure 27: High-level IFP/UDP packet structure

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4.5.2.2 IFP over UDP transport

UDPTL (Facsimile UDP Transport Layer protocol) packets comprise a Sequence Number and a variable length, octet aligned, payload.

UDPTL packets are based upon the principle of framing. Each packet may contain one or more IFP packets in its payload section. The IFP packet in a UDPTL payload is referred to as the "primary". Additional fields may be included in a payload after the primary. These fields are referred to as "secondaries".

Each packet, and therefore primary field, has its own corresponding unique sequence number which specifies an ordering at the receiving gateway should packets arrive out of sequence. To enable gateways to be synchronized upon receipt of any packet, the first primary field transmitted shall have sequence number zero. Successive primaries shall have linearly increasing (integer adjacent) sequence numbers.

Each primary contains an IFP Packet. As packets, and therefore primaries, are assigned unique and linearly increasing sequence numbers, receiving gateways can detect packet loss and re-sequencing requirements. By imposing a simple structure it is possible to provide error recovery by means of transmitting redundant information in the form of prior primary packets within each payload.

4.5.2.3 IFP message flow

As T.38 gateways must support both TCP and UDP, there are two methods of handling the TCF signal for determining the high speed data rate. When IFP packets are sent directly in TCP stream, data rate management (TCF) method 1 is used. When UDPTL packets are sent in UDP datagrams, data rate management (TCF) method 2 is used.

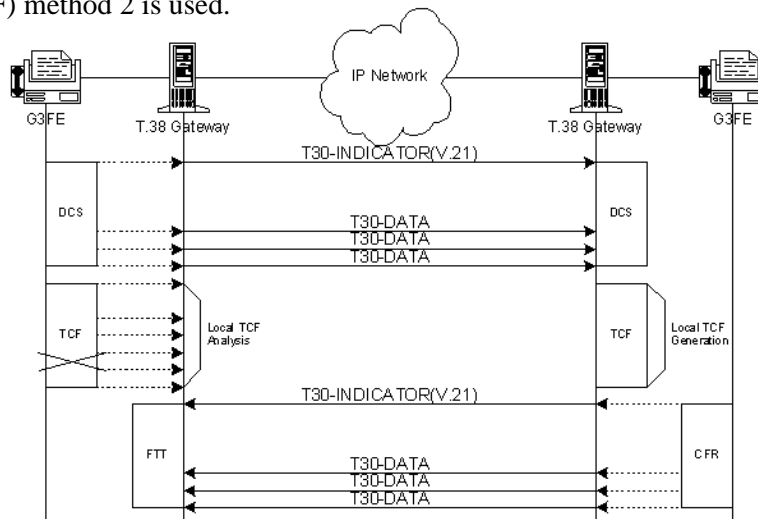


Figure 28: TCF Method 1

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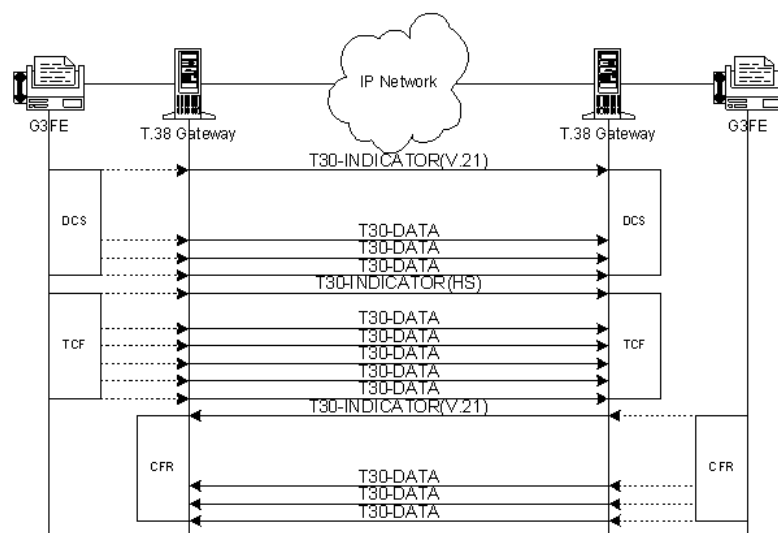


Figure 29: TCF Method 2

CFR = Confirmation to Receive
DCS = Digital Command Signal
TCF = Training Check Field

4.5.2.4 T.38 packets over H.323

Annex B of the T.38 recommendation describes how H.323 can be used for call establishment procedures of T.38 fax transmissions and gives more guidelines for real-time facsimile over H.323. Note that Annex B is outside the scope of the T.38 recommendation itself. It is therefore possible to be T.38 compliant without following all the guidelines from a particular annex.

Procedures for opening channels to send T.38 packets over H.323 is therefore gateway vendor specific. In the case of Dialogic IP telephony and fax gateways, the call setup is established using the H.323 procedures for voice calls first. As the default, a voice coder is selected. As soon as the gateway detects fax signals, the gateways switch on the fly to fax operation using the T.38 coder. If the fax signal is interrupted, the gateways switch back to voice mode without breaking the connection.

Both Fast Connect and traditional procedure in H.323 can be used for opening channels for the transportation of T.38 packets.

During H.323 capabilities exchange when using UDPTL, a gateway shall indicate its support of the available error protection schemes, parity FEC, or redundancy. Based on these capabilities, a choice may be made on which scheme is used for error protection.

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4.6 Trends

Like the signaling protocols, the *standards organisations* responsible for developing the transmission protocols described in this chapter are either IETF or ITU-T.

All the fundamental IP-based packet network protocols presented above are further developed by IETF. This concerns IP, UDP, TCP, RTP, and RTCP. The first three are very mature as they constitute the core of any IP-based network solution. Although, some more work is expected to optimize the use of TCP over satellite. RTP and RTCP are not equally mature, but still in wide use because of their lower-layer independence.

For the media protocols described in section 4.4 ITU-T is responsible for all further development of those audio and video codecs. This is also the case for the fax-over-IP protocols T.37 and T.38.

As for the *future*, IP, UDP, TCP, RTP, and RTCP are expected to be dominant protocols in any kind of networks and for any kind of services, simply because IP-solutions will become the norm rather than the exception.

Of the media protocols, G.723.1 (audio), and H.263+ (video) are the most deployed codecs in products today. However, many of the older protocols continue to be supported for backward compatibility reasons.

A couple of protocols not described above are also worth mentioning as they might prevail in the long run. The various MPEG audio and video codecs from ISO might prove to be successful, and follow in the footsteps of MP3 (MPEG-1 audio layer 3) for music download. Proprietary codecs used by popular multimedia players from Real and Microsoft might also take over if released and standardised, as they already have a large installed base on the Internet.

What all these codecs have in common is that they operate at a higher bit rate than those described in this chapter, but with broadband networks that might be less critical in the time to come.

It is difficult to see a future for fax over IP protocols as anything else than for backward compatibility with existing and deployed equipment.

4.7 Applicability to a GEO Satellite Environment

This study will consider the general applicability of the terrestrial based transmission protocols described in this chapter to a GEO satellite environment taking into consideration the characteristics of the satellite environment.

It appears that the main characteristics that influence the applicability of the VoIP terrestrial based transmission protocols are satellite delay and the resulting latency, jitter, and high BER. This will be addressed in more detail in chapter 6 below.

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5. QUALITY OF SERVICE

5.1 Aspects of Service Quality

The term “*Quality of Service*” refers to the various non-functional requirements or aspects of a system (or service). A number of such aspects may be of interest. According to Blair and Stefani [33], when considering distributed multimedia services, it is useful to group the different aspects into the following three discrete categories:

Category	Aspects in this Category
Timeliness	Aspects like end-to-end-delay, delay jitter, setup latency (if applicable)
Volume	Throughput dimensions like frame rate, packet rate or bit rate.
Reliability	Various aspects of data and service loss like Bit Error Rate (BER), Packet error rate, Mean time between failure/repair (MTBF, MTBR)

Other aspects and categories may also be considered as important to the overall usability or user satisfaction with a certain service. Blair and Stefani mention *criticality*, *perception quality* and *cost*, but consider these to be of secondary importance compared to those above.

5.2 Service Provision and Use of Resources

Application programs, like for instance telephony applications, utilize *resources* in order to provide certain services (or functionality) to the user. A *resource* may in this regard be any piece of hardware (processing power, memory) or service provided by underlying software (networking library, window system). What is a service provided by one entity, may be used as a resource by another. The most obvious example is network transport capacity, provided by network libraries, routers and links, being used as a single resource by the applications. In the end, these underlying services are provided utilizing underlying hardware. Certain quality aspects of the delivered service, like, for instance, correctness and speed, are limited by the quality of resources utilized and by physical aspects of the underlying hardware resources. Many resources, like router processing power, are shared among several users (or application instances). The quality experienced by a user (or, rather, application) will then be limited by the amount of resources devoted to (or spent servicing) each application.

It is an axiom of network research that quality lost can never be regained. The quality experienced by the end user and by all components involved in creating the service will thus (to a certain extent) construct a quality dependability graph.

Even if lost quality never can be regained, it is (to a certain extent) possible to trade off some quality aspects (like delay) and resources (like buffer memory) in order to compensate for insufficient level of another quality aspect (like delay jitter). Another example is sending redundant data, and thus using more bandwidth and computing power, to reconstruct lost/corrupted packets.

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5.3 Descriptions of GEO Satellite Environments and Systems

5.3.1 General Information

The general properties of GEO satellite environments and systems are described in section 2.5.

5.3.2 Use Case Topologies for GEO Satellite Network Environments

When considering QoS, it is important to take into consideration the used case topologies for GEO satellite networks, since there exist several distinct alternatives that influence the overall quality and performance. These alternatives are illustrated in Figure 30.

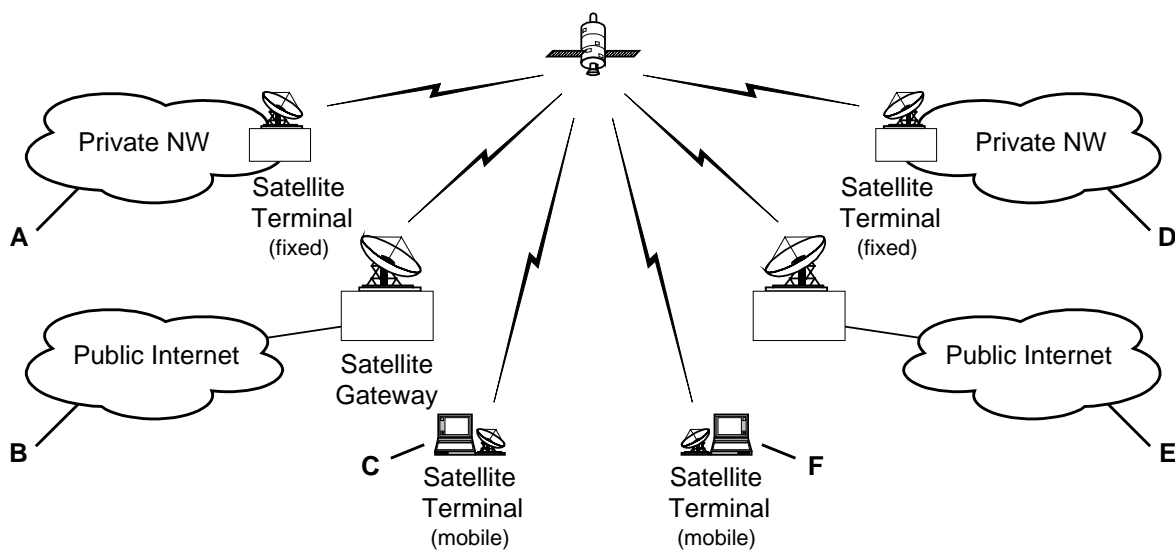


Figure 30: Alternative topologies for GEO satellite network environments

The figure illustrates — on each side of the satellite link — three qualitatively different “paths” by which to access the satellite. The purpose in distinguishing connections to the satellite via private (data and/or telecommunication) networks from connections via public internet is to demarcate the fact that it *may* be possible to obtain guarantees about the QoS characteristics of data traffic (e.g., delay) when data is transmitted over private networks.

The satellite gateway is comprised by a collection of advanced components. In addition to complex satellite technologies, routers and switches are also located in the satellite gateway. In an H.323-based satellite gateway, an H.323 Gatekeeper and H.323 Gateway are also included. Satellite terminals can be both large and small in size, depending upon bandwidth capacity, transmission power, etc. Satellite terminals can be connected to networks and/or connected directly to individual PCs / workstations. They may also be fixed in position or portable.

Figure 30 depicts nine different combinations of connections over the satellite link. As an illustration of the capacity to control one aspect of QoS, let us assume that one has control of the delay within the satellite terminals and the satellite gateways. Let us also assume that a certain degree of control over traffic delay can be obtained through business agreements with the private network operators. Using these assumptions,

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the different combinations of connections over the satellite link can be divided into four distinct groups. Here, these groups are distinguished from one another based upon *the ability and manner by which to control the total delay from end-to-end*. These groups are:

- *group1*: connection types which employ public internet on one or both sides of the satellite link, without employing a private network on either side (e.g., B-E, B-F, C-E);
- *group2*: connection types which employ public internet on one side of the satellite link, and a private network on the other side (e.g., A-E, B-D)
- *group3*: connection types which employ a private network on one or both sides of the satellite link, without employing public internet on either side (e.g., A-D, A-F, C-D); and
- *group4*: connection types which do not employ private network nor public internet on either side of the satellite link (e.g., C-F).

Since one has a certain degree of control over delay in this example, one may easily presume that delays associated with *group3* and *group4* are always less than that of *group1* and *group2*. Knowledge about control may be misleading. **It is very important to understand the difference between the *capacity to control* a QoS parameter and the actual magnitude of that parameter during system operation.**

Consider the example in which a VoIP connection between C and F requires one type audio format at one terminal and a second type audio format at the other terminal. Since the GEO satellites that exist today cannot do switching/routing between terminals, all IP to IP calls must use the satellite gateway to route the IP packets between terminals C and F.

The audio media from each terminal must be transferred *twice* over the satellite link. The audio media from C is transmitted over the satellite to a satellite gateway and then to the H.323 GW where it undergoes transcoding. The encoded audio is then transmitted from the satellite gateway via the satellite to terminal F. The same is true for the audio media being transmitted from F to C. In this case, delays, as well as other quality aspects of the signal (e.g., noise) will be double in magnitude.

5.4 QoS Properties of GEO Satellite Environments

5.4.1 Delay and Loss in GEO Sat Systems

The long wireless jumps of satellite data links have serious impact on the quality properties on data streams utilizing such links. GEO satellites orbit the earth at approx. 36000 km distance. Often, the actual satellite used will not be located “right above” the user, adding further distance to the hop. As radio waves travel at about 3×10^8 m/s, the total propagation delay over a GEO satellite link may expected to be in the approximate range of 260 – 600 msec. including satellite hop, signal processing and encoding/decoding at sender and receiver.

The total delay experienced for each transmitted packet will be greater than the pure signal propagation time due to various queuing, transcoding et cetera. The delays will not be the same for all packets transmitted during a session. The expected delay jitter between the packets will be between plus and minus 5-30 msec.

Wireless links are also exposed to noise and interruptions causing loss of data. Even if some data may be recovered using various methods in link level hardware and software, the average Bit Error Rate (BER) is in the range of 10^{-12} - 10^{-6} . Losses often occur in bursts. The average packet error rate will therefore be considerably smaller than the product of average packet size and BER:

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$PER \ll (BER \times \text{avg. packet size}).$

The packet error rate is expected to be between 10^{-9} and 10^{-5}

5.4.2 Comparing to the Internet

Comparing these figures with similar figures for the Internet is difficult, as there is no “average hop”, distance or network to measure. *Matrix Internet Quality*[34] performs some ratings based on Echo requests (“ping”) from a number of “beacons” to a number of hosts in many networks, measuring latency (Roundtrip time to host and back again), packet loss and reachability. The value for each network is, of course, highly dependent on its distance from a number of “beacons”. The scale used is therefore of more value for this report. The networks are labeled with the colors red (worst), yellow and green (best) depending on their measured performance. Yellow color is given for two-way latency between 100 and 175 msec, packet loss between 1.5% and 3.5% and reachability between 97% and 100 %. With this coloring scheme, there are usually most yellow, and more green than red networks. The delay jitter is not measured for networks or the Internet as a whole, but a few traceroute tests indicate that the jitter, measured as delay standard deviation, is at about the same size as expected in GEO satellites. Jitter also tends to increase as the distance of the satellite hop increases.

5.4.3 Real-time Media and Bandwidth Guarantees

For delivery of real-time multimedia communication, bandwidth is obviously an issue. In the internet there is no widely deployed method for prioritizing or reserving bandwidth for real-time traffic, even if some standards for this have evolved over a few years. Many websites now contain multimedia and, in general, the bandwidth available is sufficient for streaming of compressed sound and highly-compressed mediocre-quality video. Over shorter distances, even average- to high-quality video may be transferred in real time successfully.

It is not likely that IP links over GEO satellites will be open to the public. Rather, we can expect that clients wanting to use such links will have to connect through a gateway where they need to identify themselves. In a closed environment like this, guaranteeing a certain minimum bandwidth to clients — **over the satellite hop** — is a lot easier. The total bandwidth on a satellite link is sufficient to serve several clients with higher capacity than they will get through the rest of the Internet connection. Several techniques can be used to guarantee each user a minimum bandwidth. The easiest is to only allow new users when sufficient capacity is left. Differentiated services like “premium membership” can be used to serve some clients with higher capacity than others. In any case, a border gateway should be used to make sure no single user is allowed to consume more than his allocated amount of resources. Dynamic resource distribution services like RSVP [51] or DiffSrv [52] are more difficult to offer, as these in general require specific network stacks and/or software in the end systems. RSVP is intended for end-to-end reservation, but the service is generally not available except within more than a few “islands” in the Internet. Implementing a DiffSrv or RSVP “island” across the GEO sat link should be possible, since different radio frequencies can be assigned to the different streams or priority queues as needed.

5.5 QoS Properties of Distributed Telephony and Multimedia Systems

In packet based communication systems, delay is introduced by all elements that need an entire packet to continue its work. Such entities have to wait for the end of the packet to arrive before it can start processing

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the beginning of the packet. This will delay the start of the packet by the processing time and by the “duration” of the packet. The most important entities of this kind are:

- **Routers:** Certain routers do not have features for routing packets as soon as the header bytes have been received, but have to wait for the entire packet. The duration of a packet in this case is:
[duration] = [packet size] / [link speed] + [net interface and protocol processing overhead]
- **Network protocols:** Network protocols especially in end systems, often operate on entire packets.
- **Algorithms for audio compression:** In general, such algorithms work on a set of samples representing 20-120 msec of sound input. Since the data arrive in real time, the duration of a frame is the actual number of milliseconds of music used as computational unit. The decompression algorithm will of course operate on a corresponding set of data. A similar delay will thereby be generated if the decompressed sound samples are to be delivered in real time, as the case is when the data is delivered to the user. The actual processing may also consume measurable time, but this is expected to be unnoticeable compared to the other sources of delay.

A media gateway performing media transcoding does not double the algorithmic delay, since neither input nor output of data has to be performed as slowly as real time. The processing may, however, take some time, as will receiving and sending the packet over the packet network interface(s).

5.6 Threats to VoIP Stability and Quality Arising from QoS Characteristics of GEO Satellite Environments

This topic is treated in sections 6.3.4.1 and 6.3.4.2.

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6. CRITERIA, APPLICABILITY AND RECOMMENDATIONS FOR A GEO SATELLITE ENVIRONMENT

6.1 Trends

In section 3.10 and 4.6, general trends for IP-based multimedia signaling and transmission protocols were described, without taking into account the type of network infrastructure the protocols should run over. However, as all the protocols are foremost developed for terrestrial (wired) systems, the trends are mostly applicable to such systems.

This section attempts to compare terrestrial and GEO sat systems, and indicate whether there are any obvious differences between the systems when it comes to *technology*, *services* and *market* trends. In the remaining sections of this chapter the focus will be specifically on IP-based services in GEO sat environments.

The main general trend in computing today is digital convergence with IP technology introduced everywhere, on all platforms, and systems in the market. There is also more expanded use of wireless and mobile services and networks, as well as an increasing number of different "last-mile" broadband technologies with a potentially high degree of multimedia content.

The consequence of this trend can especially be seen in GEO sat systems, since all three important characteristics are inherent in such systems: IP is introduced, communication is wireless (and terminals often mobile), and the pipeline is broad.

Both terrestrial and GEO sat systems offer real-time, interactive and/or high bandwidth services. And the trend is to offer the *same* services in both systems. However, the technological differences (inherently large delay and BER in satellite link) makes GEO sat systems better suited for *low interactive, broadcast* services. Still, both networks share the technological challenge of reserving resources and providing QoS guarantees across the network in order to offer these kind of multimedia services.

6.2 Using IP as Basis for Services in GEO Satellite Environments

This section concerns the use of IP as a basis for the provision of *general services* in GEO satellite environments. The subsections below which cover this topic are organized as follows:

- Commercial Aspects in General
- Technical Aspects in General
- Standards, Products and Maturity
- Special Issues.

6.2.1 Commercial Aspects in General

6.2.1.1 Ideas from emerging satellite systems and suppliers

There is a large and well-developed commercial market for products, solutions and services that involve — or are related to — media transmission via satellite. Traditionally, this market has had its focus in commercial broadcasting. Given the technological convergence which has become more and more manifest throughout the last decade, this commercial market has become both wider and more specialized; products, solutions and services can be found at many levels and niches in the value-chain. These include:

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- protocol products
- equipment suppliers (satellite terminals and modems, gateways, accelerators, etc.)
- global carriers, with products such as provision of broadband, satellite-based connectivity at several levels (e.g., link level, network level, etc.)
- developers / providers of end-user services, for both business and consumer segments
- turn-key solutions
- consultancy.

To provide an overview of the market, a number of companies delivering products / services which involve satellite transmission are profiled below. For each, information about their respective business focus and available or planned services/products is provided. Most of these companies offer regional, transcontinental or global connectivity as broadband "carriers". Most of these same companies also offer end-user services developed through either their own organization, organizational subsidiaries, business partners or other service providers. The exceptions here are: Fourelle Systems, which offers web acceleration solutions; and Mentat, which offers protocol and gateway products.

It is important to recognize that not all carriers below use exclusively GEO satellite constellations. Furthermore, not all products, solutions and services available from these carriers employ IP over the satellite link: instead, a number of them are employing other standard or proprietary protocols for packet transmission between gateways (or client-server proxy pairs) on each side of the satellite link(s). Nevertheless, these companies are included here in order to provide a more comprehensive overview of *commercial developments and opportunities for support of services which utilize (or can utilize) IP as the source and/or destination packet format.*

The companies profiled here are:

ASTRA	http://www.astra.lu/
Astrolink	http://www.astrolink.com/welcome.html
Broadlogic	http://www.broadlogic.com/
Cyberstar, Loral	http://www.cyberstar.com/
EUTELSAT	http://www.eutelsat.org/home.html
Fourelle Systems	http://www.fourelle.com/
Gilat Satellite Networks	http://www.gilat.com/gilat/
Gilat-To-Home	http://www.gilat2home.com/
Inmarsat	http://www.inmarsat.org/index3.html
INTELSAT	http://www.intelsat.com/
Loral Skynet	http://www.loralskynet.com/
Mentat	http://www.mentat.com/skyx/skyx.html
Skystream Networks	http://www.skystream.com/
Spaceway (Hughes Network Systems)	http://www.hns.com/spaceway/spaceway.htm
Sterling Satellite Communications (S2COM)	http://www.s2com.net/
Teledesic	http://www.teledesic.com/
Thaicom (Shin Sat)	http://www.thaicom.net/
WildBlue (formerly iSKY, KaSTAR)	http://www.wildblue.net/

6.2.1.1.1 ASTRA [85]

Business Plan / Focus: Direct-to-home (DTH) transmission of TV, radio and multimedia in Europe.

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Services: Provision of bulk transmission capacity to multimedia service providers. Through ASTRA-NET, the multimedia service platform operated by the 100% SES-owned SES Multimedia S.A., ASTRA provides bandwidth and value-added services for broadcasting and multicasting customers. ASTRA is also creating a two-way broadband communications system via the ASTRA Satellite System by enhancing ASTRA-NET with satellite return channel technology.

6.2.1.1.2 Astrolink [86]

Business Plan / Focus: To provide worldwide, wireless connectivity for businesses.

Services (plan): To provide data, video and voice services that support business applications; interactive or two-way high-speed connections point-to-point service; bandwidth-on-demand; multicasting services.

6.2.1.1.3 Broadlogic [87]

Business Plan / Focus: The company is focused on providing innovative products for data broadcasting and 2-way satellite applications that bring media-rich content to large corporations, Internet infrastructure providers and DTV broadcasters and their customers.

Products: broadband networking equipment for sending and receiving high-speed video and data

6.2.1.1.4 Cyberstar, Loral [88]

Business Plan / Focus: To be the leading provider of innovative telecommunications services differentiated by reliability and responsiveness to Businesses, Content Producers and Providers, Broadcasters and Carriers.

Services / Solutions:

- Internet Services
 - IP Multicast Services
 - NewsFree
- Infomedia Services
 - Business Television
 - ClearStream Forum
 - Interactive Distance Learning
 - e-Package Delivery
 - Video Kiosk
- Managed Network Services
 - *Digital LinkSM* is a satellite network service that uses very small aperture terminals (VSATs) to send and receive digital signals for voice, data, video, e-mail and fax transmissions.
 - *Digital Channelized LinkSM* is a multiplexed version of Digital Link that integrates the full range of digital services on a single satellite link.
 - *DynaLinkSM* offers two distinct types of services, sold separately. The first service, DynaLink Network Management Station Service (DNMS), allows customers to dynamically allocate or share satellite bandwidth between network nodes or sites between countries or across continents. The second service is a DynaLink File Broadcast Service (FBS). DynaLink Applications include:
 - Publishing
 - Manufacturing
 - Video Conferencing

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6.2.1.1.5 EUTELSAT [90]

Business Plan / Focus: EUTELSAT is an intergovernmental organization with 48 member States whose main purpose is the design, development, construction, establishment, operation and maintenance of the space segment of a European communications satellite system. This space segment is used for all types of telecommunications and audiovisual services including telephony, telecommunications services for business use (VSATs, video-conferencing) and multimedia services, as well as mobile services.

By early-2001, EUTELSAT's structure will be streamlined into two tiers: a limited company (EUTELSAT S.A.) based in France, to which all assets and activities will be transferred, and an intergovernmental organization which will ensure that basic principles of pan-European coverage, universal service, non-discrimination and fair competition are observed by the company.

Services: EUTELSAT was the first satellite operator to support digital interactive services and the first to introduce DVB [89] signal multiplexing on-board a satellite. It shall continue to develop and deploy services for both the business and consumer segments. These include: TV, radio and interactive services; Internet-via-satellite; corporate television, corporate communications and mobile satellite-based communications (e.g., Euteltracs and EMSAT systems).

6.2.1.1.6 Fourelle Systems [91]

Business Plan / Focus: Fourelle Systems, Inc. is the global market leader in the deployment of web acceleration solutions for wide area networks including satellite, frame relay, wireless, cable and dial-up.

Products: Venturi, Fourelle's flagship product platform, is the best selling solution for speeding web content delivery, increasing network capacity and reducing network operating costs. Utilizing a dual-proxy client/server architecture and patent-pending compression technology, Venturi minimizes the TCP "slow-start" mechanism, and significantly reduces the inherent latency and backhaul bandwidth requirements caused by satellite transmissions.

Venturi consists of three elements: the Venturi Server, Venturi Workgroup Client or Venturi Personal Client, and the Venturi Transport Protocol (VTP), an optimized protocol, which handles data communications between the two. The Venturi Server resides on one end of a bandwidth-constrained link and the client resides on the other end. Data is compressed and decompressed on the fly as Venturi's unique technology greatly reduces the amount of data transmitted across wide-area IP links. See also section 6.2.2.4.4.

6.2.1.1.7 Gilat Satellite Networks [92]

Business Plan / Focus: To remain a leading provider of telecommunications solutions based on VSAT satellite network technology. The Company delivers satellite-based, end-to-end enterprise networking and rural telephony solutions to customers across six continents, and markets interactive broadband data services. In conjunction with Microsoft Corporation, EchoStar Communications Corporation and ING Furman Selz Investments, Gilat has formed a new company, *Gilat-to-Home*, the first consumer service to offer two-way Internet connectivity powered by broadband satellite technology.

Products:

- *SkyBlaster VSAT:* delivers broadband video and data directly to the LAN or desktop, using a PC-based satellite transmitter as a return channel.
- *SkySurfer:* a PC-based DVB satellite receiver suited for private IP networks. The SkySurfer hub, typically located at a company's headquarters or at the facilities of a service provider, delivers a scalable 2-35 Mbps pipe with a terrestrial or satellite-based (SkySurfer Pro) return channel. SkySurfer represents

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a complete DVB solution supported by enhanced Network Management capabilities. It provides individual IP access for Internet/Intranet browsing (unicast) and IP multicast capabilities

- *WebSat*: an enhancement of ISAT used for Internet and Intranet connections and service to locations where alternative connections are not easily available. It enhances the standard advantages of ISAT, along with high outbound rates and low inbound rates to match typical traffic patterns, for efficient use of bandwidth. WebSat integrates an IP accelerator that overcomes the typical satellite speed limitations for IP data, and manages data flow from multiple connections, using Quality of Service (QoS) bandwidth controls.

Services / Solutions (Corporate-level):

- *Corporate Training*: Based on its TrainNet application, Gilat provides turnkey, one-shop solutions for corporate training requirements at the desktop. Featuring full broadband software video decoding with multicast capabilities, TrainNet provides on-line training directly and simultaneously to hundreds of employee LANs and PCs. In addition, TrainNet incorporates a comprehensive set of tools for instructors to deliver live, on-line and fully interactive training sessions.
- *Interactive Business TV*: Gilat offers IP-based multicast and Intranet technologies that provide interactive business television with MPEG 2 decoding quality. Gilat provides a turnkey solution, beginning with the customer's video source, continuing with the video encoding server (including the IP multicast data layer) and ending with the video decoder card or software.
- *Data Push*: SkySurfer/Blaster offers an integrated end-to-end Push client/server solution, optimized for satellite delivery and IP multicast. High-end Push technologies – with a range of notification techniques – allow companies to proactively deliver corporate information from a variety of sources and to notify employees and management of its availability.
- *Reliable IP Multicast*: SkySurfer/Blaster provides a turnkey solution for reliable IP multicast. It offers a solution that turns IP multicast – a non-acknowledged protocol – into reliable data multicasting.
- *Broadband Intranet/Internet access*: Using satellite technology – enhanced by Gilat's IP spoofing technology – corporate users can enjoy data rates of more than 1 Mbps while accessing their corporate Web servers or Internet sites.

6.2.1.1.8 Gilat-To-Home [93]

Business Plan / Focus: Gilat-To-Home (GTH) is a new consumer-focused company formed by Gilat Satellite Networks Ltd. to deliver two-way broadband satellite Internet access to consumers.

Services / Solutions:

GTH is the first two-way, always-on satellite-delivered Internet service for the consumer market. GTH uses a specially designed 24x36-inch satellite dish equipped with both a satellite transmitter and receiver for two-way satellite connectivity. Inside the home, the GTH system currently consists of a pre-configured desktop PC that includes a satellite receiver card and a satellite transmitter card plugged into two PCI slots. In the near future, GTH will also offer an external "satellite modem" option that packages the PCI cards in a stand-alone box and connects to a PC through the USB port. In both cases, no telephone connection, no dial-up account and no terrestrial Internet service provider are needed.

6.2.1.1.9 Inmarsat [94]

Business Plan / Focus: Inmarsat is actively following a business strategy that will place it at the heart of the convergence between computers, communications and mobility. This strategy will enable it to continue to serve its traditional markets in the maritime, aeronautical and land mobile communities, along with new markets for personal and multi-media mobile satellite communications.

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Services / Solutions: Inmarsat offers satellite links as a basis for communication services. Service providers / customers associated within the "Inmarsat Partnership" have developed *Inmarsat-based* communications services including: direct-dial telephone, data, facsimile, telex, electronic mail, high quality audio, compressed video and still video pictures, telephoto, slow-scan television, videoconferencing and telemedicine.

6.2.1.1.10 INTELSAT [95]

Business Plan / Focus: To provide satellite capacity to ISPs and work with global telecommunications operators, teleports and satellite service providers to package complete solutions, including: business-to-business satellite services for telecommunications applications, such as voice, data, Internet, corporate networking and video services.

Products and Services:

- *Broadband VSAT:* INTELSAT's newest product offering a cost-effective and rapidly deployable networking solution for data, voice, multimedia and high speed Internet using IP, Frame Relay, ATM, ISDN or SS7 services.
- *Internet Services*
- *Video Services*
- *Voice and Data Services:* INTELSAT's telephone, fax, videoconferencing and multimedia traffic reaches virtually every country in the world. IBS, IDR, TDMA, DAMA, Intelnet, VSATs.

6.2.1.1.11 Loral Skynet [96]

Business Plan / Focus: Continued development and provision of advanced satellite-based communications networks. Focus upon development of strategic partnerships (Loral's "Global Alliance") for creation and provision of leading-edge services.

Services / Solutions:

Loral Skynet's basic data services include Internet backbone extension, private data circuits and data circuits for telephony applications. Allied service providers have created end-user services such as:

- Television Broadcasting
- Cable & Direct-to-Home
- Sports & Events
- Data Transmission
- Internet Applications
- Satellite News Gathering
- Business Television
- Distance Learning
- Video Conferencing
- Telephony

6.2.1.1.12 Mentat [97]

Business Plan / Focus: To supply high performance networking solutions to the computer and satellite markets.

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Products:

- Protocol products:
 - *Mentat TCP*
 - *Mentat Portable Streams*
 - *MPS FOR WINDOWS NT*
 - *Mentat XTP* (see section 6.2.3.2.2).
- Satellite products:
 - *SkyX Gateway*: The SkyX Gateway system replaces TCP over the satellite link with a protocol optimized for the long latency, high loss, asymmetric bandwidth conditions typical of satellite communications. See section 6.2.3.2.2.

6.2.1.1.13 Skystream Networks[99]

Business Plan / Focus: SkyStream Networks is a worldwide networking infrastructure company that empowers service providers and broadcasters to create new revenue streams by delivering branded digital media services like TV-quality Internet video to PCs or TVs over linked broadband and broadcast networks. SkyStream makes high-performance media routers and software to help service providers deliver a consistently high-quality Internet experience to millions of people. SkyStream's routers are the intelligent links of converged broadband and broadcast networks at the core of the Internet, joining networks of satellite providers, cable providers and broadcasters with those of traditional Internet bandwidth owners of fiber and ATM networks.

Products / Solutions:

SkyStream offers a full-scale networking solution for service providers that include networking routers and content distribution software to meet their individual needs.

- *SkyStream Source Media Routers (SMRs)*: enable broadcasters, satellite and digital cable providers to add/inject Internet content into a secure, broadcast stream, using their existing networks and reliably deliver the content to stadium-size audiences.
- *SkyStream Edge Media Routers (EMRs)*: The partner to SkyStream SMRs, these EMRs reliably receive data in MPEG-2 transport stream format and cost-effectively deliver streaming IP-multicast content to remote servers and networks - and ultimately to massive numbers of consumers.
- *zBand content distribution system software*: features server and client components that enable service providers and broadcasters to create and deliver Internet services such as regional news channels, streaming content and corporate training videos to millions of consumer and businesses. zBand enables service providers to deliver this content over broadcast and broadband networks, resulting in a consistently high-quality Internet viewing experience. zBand can be used in conjunction with SkyStream networking hardware or other standards-based broadband equipment.

6.2.1.1.14 Spaceway (Hughes Network Systems) [100]

Business Plan / Focus: Hughes Network Systems (HNS) is a world leader in telecommunications technology — satellite, digital cellular, frame relay, ATM, and IP switching. HNS designs, manufactures and installs advanced networking solutions for businesses and governments worldwide. All HNS products

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are turnkey systems integrated by HNS engineers into customers' environments. HNS supplies hardware, software, installation, training, documentation, and maintenance. HNS can also supply satellite transponders and shared hub services for customers.

Products:

- *Enterprise Networks:* The Hughes Network Systems (HNS), VSAT product line is designed to provide cost-effective telecommunications to a wide array of industries around the world. VSATs are effective tools for LAN internetworking, multimedia image transfer, batch and interactive data transmission, interactive voice, and broadcast data and video communications.
 - Voice Products
 - Interactive Data Products
 - High Speed Data Products
 - Interactive Videoconferencing
 - VSAT Software Packages
- *Broadband Carrier Networks:* The AIReach® family of wireless network products is designed to meet a broad spectrum of demanding applications for wireless carriers, including digital mobility, wireless local loop, wireless data, and broadband business access.
- *Consumer Satellite Products:* HNS offers a series of consumer satellite products designed to provide world-class, direct-to-home satellite TV and Internet access.
 - *DIRECTV Systems:* direct-to-home reception of satellite-delivered DIRECTV programming
 - *DirecPC®:* satellite receive system and service that provides up to 400 Kbps Internet access.
 - *DirecDuo:* hybrid satellite receive system that provides satellite-delivered TV reception and DirecPC Internet access using the same antenna.

6.2.1.1.15 Sterling Satellite Communications (S2COM) [101]

Business Plan / Focus: S2COM is a 4th generation (4G) integrated global carrier that offers voice and data services, both fixed and mobile. S2COM as an emerging IT services provider has an emphasis on direct personal financial, commerce and business knowledge / advise / transaction services to its financially advantaged consumer base. S2COM is designing a patent NextGen global IP LEO satellite network to bring radically lower cost data, Email, Fax, Voicemail, voice and video to mobile users worldwide.

Services (planned):

- Children's Services
- Unified Messaging
- E-Business Services
- E-Medicine Services
- Security Services
- Rural Services
- Location
- Education
- Banking
- Multiplayer Games

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- S2COM Music
- S2COM Movie Theater
- Emergency Net

6.2.1.1.16 Teledesic [102]

Business Plan / Focus: Teledesic is building a global, broadband Internet-in-the-Sky™ network using a constellation of LEO satellites. Teledesic will develop alliances with service partners in countries worldwide, enabling service providers to extend their networks, both in terms of geographic scope and in the kinds of services they can offer. Teledesic will also market services directly to select customers. Service is targeted to begin in 2005.

Services (planned):

Teledesic and its international partners are creating a worldwide, fiber-like access to support telecommunications services such as computer networking, broadband Internet access, high-quality voice and other digital data needs.

6.2.1.1.17 Thaicom (Shin Sat) [103]

Business Plan / Focus: The first privately-owned satellite company in Asia (1991). Shin Satellite is a subsidiary of Shin Corporations., Thailand’s largest and most successful telecommunications group. It has one Thai subsidiary, C.S. Communications Co. Ltd., which operates a teleport and an Internet Service Provider, CS Internet.

Shin Satellite now has three satellites in geostationary orbit with corresponding Thai-based customer service facilities.

Services / Solutions:

- Transponder Leasing
- Internet-via-Satellite
- Teleport and Digital DTH Services
- Global Digital TV Service
- Business and Education TV
- Rural Telephony
- VSAT

6.2.1.1.18 WildBlue (formerly iSKY, KaSTAR) [104]

Business Plan / Focus: Plans to deliver affordable high-speed Internet access services via satellite to homes and small offices by early 2002. Currently creating the operational infrastructure of sales, marketing, installation, customer service and billing. WildBlue was originally founded as KaSTAR Satellite Communications in Colorado in April 1995.

Services (Future): Interactive TV, Home Networking & Net Appliances. Service is targeted to begin in 2002.

6.2.1.2 Market projections

Recent studies (Pioneer Consulting, 1999; Booz-Allen & Hamilton, 1998) predict that the broadband-satellite market will be worth between US\$20 billion and US\$30 billion 2005 [86].

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6.2.1.3 Expected costs

It has been difficult to obtain general information about actual and expected prices for equipment and services via the WWW. Obtaining such information often requires an initial and somewhat time-consuming *indirect contact* with the sales organization. Such indirect contact is often initiated by the potential buyer through WWW-based forms, wherefore the selling party promises to return contact (usually via email).

6.2.2 Technical Aspects in General

6.2.2.1 Network topologies

When considering the provision of general, IP-based services from an *end-to-end perspective*, it is important to keep in mind the kind of network topology over which they are being delivered (see section 5.3.2). Allman et al. [50] provide a short summary of several possible network topologies where satellite links may be integrated into the communication path. These topological considerations, which are included below, provide both an extended and orthogonal view upon those depicted in Figure 30.

Asymmetric Satellite Networks

Some satellite networks exhibit a bandwidth asymmetry, a larger data rate in one direction than the reverse direction, because of limits on the transmission power and the antenna size at one end of the link. Meanwhile, some other satellite systems are unidirectional and use a non-satellite return path (such as a dialup modem link). The nature of most TCP traffic is asymmetric with data flowing in one direction and acknowledgments in opposite direction. However, the term asymmetric in this document refers to different physical capacities in the forward and return links. Asymmetry has been shown to be a problem for TCP [BPK97,BP].

Satellite Link as Last Hop

Satellite links that provide service directly to end users, as opposed to satellite links located in the middle of a network, may allow for specialized design of protocols used over the last hop. Some satellite providers use the satellite link as a shared high speed downlink to users with a lower speed, non-shared terrestrial link that is used as a return link for requests and acknowledgments. Many times this creates an asymmetric network, as discussed above.

Hybrid Satellite Networks

In the more general case, satellite links may be located at any point in the network topology. In this case, the satellite link acts as just another link between two gateways. In this environment, a given connection may be sent over terrestrial links (including terrestrial wireless), as well as satellite links. On the other hand, a connection could also travel over only the terrestrial network or only over the satellite portion of the network.

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Point-to-Point Satellite Networks

In point-to-point satellite networks, the only hop in the network is over the satellite link. This pure satellite environment exhibits only the problems associated with the satellite links, as outlined in [AGS99]. Since this is a private network, some mitigations that are not appropriate for shared networks can be considered.

Multiple Satellite Hops

In some situations, network traffic may traverse multiple satellite hops between the source and the destination. Such an environment aggravates the satellite characteristics described in [AGS99].

(from: Allman, et al., [50], p. 3-4)

6.2.2.2 The root of the problem

Before proceeding further with this section, three introductory remarks are in order:

- This section concerns the use of IP as a basis for the provision of *general services*; it is not limited to VoIP, Video over IP and Fax over IP alone; use of IP for these *specific* services is the focus of section 6.3.
- Regarding such use of IP, it is important to remember that IP defines both a *packet format* and a *transmission protocol*. The discussion in the remainder of this section derives its primary focus from aspects of IP as a transmission protocol. Considerations of IP as a packet format have their greatest focus in subsection 6.2.2.5.
- In addition, the focus of the discussion primarily addresses problems and issues for GEO satellite environments, though certain of these issues are relevant for satellite environments as a whole. Other complex issues and problems associated with MEO / LEO environments — problems such as on-board processing, on-board switching, intersatellite links, etc. [53][57] — are not within the scope of this study.

With this focus in mind, there is perhaps one, most central hurdle to overcome when considering the use of IP as a basis for the provision of general services in GEO sat environments: **the hurdle is TCP**. Quite simply, services which use IP as a network-level protocol *almost always* employ TCP and/or UDP as transport-level protocol(s) — these transport protocols are at the heart of many services which are implemented in a “standardized” manner. Furthermore, since TCP is designed and implemented as a *reliable* protocol, many IP services utilize TCP for reliable exchange of control signals.

6.2.2.2.1 Satellite characteristics impacting TCP

TCP is designed as an end-to-end protocol, with features and mechanisms which help guard against network congestion (i.e., slow-start, congestion avoidance, fast retransmit and fast recovery). These mechanisms have not been designed in a manner which naturally copes with the inherent characteristics of satellite links. A concise summary of the characteristics of satellite channels which may degrade TCP performance is found in [49] and included below:

Long feedback loop

Due to the propagation delay of some satellite channels (e.g., approximately 250 ms over a geosynchronous satellite) it may take a long time for a TCP sender to determine

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whether or not a packet has been successfully received at the final destination. This delay hurts interactive applications such as telnet, as well as some of the TCP congestion control algorithms.

Large delay*bandwidth product

*The delay*bandwidth product (DBP) defines the amount of data a protocol should have "in flight" (data that has been transmitted, but not yet acknowledged) at any one time to fully utilize the available channel capacity. The delay used in this equation is the round-trip time (RTT), and the bandwidth is the capacity of the bottleneck link in the network path. Because the delay in some satellite environments is large, TCP will need to keep a large number of packets "in flight" (that is, sent but not yet acknowledged) .*

Transmission errors

Satellite channels exhibit a higher bit-error rate (BER) than typical terrestrial networks. TCP uses all packet drops as signals of network congestion and reduces its window size in an attempt to alleviate the congestion. In the absence of knowledge about why a packet was dropped (congestion or corruption), TCP must assume the drop was due to network congestion to avoid congestion collapse [Jac88] [FF98]. Therefore, packets dropped due to corruption cause TCP to reduce the size of its sliding window, even though these packet drops do not signal congestion in the network.

Asymmetric use

Due to the expense of the equipment used to send data to satellites, asymmetric satellite networks are often constructed. For example, a host connected to a satellite network will send all outgoing traffic over a slow terrestrial link (such as a dialup modem channel) and receive incoming traffic via the satellite channel. Another common situation arises when both the incoming and outgoing traffic are sent using a satellite link, but the uplink has less available capacity than the downlink due to the expense of the transmitter required to provide a high bandwidth back channel. This asymmetry may have an impact on TCP performance.

Variable Round Trip Times

In some satellite environments, such as low-Earth orbit (LEO) constellations, the propagation delay to and from the satellite varies over time. Whether or not this will have an impact on TCP performance is currently an open question.

Intermittent connectivity

In non-GEO satellite orbit configurations, TCP connections must be transferred from one satellite to another or from one ground station to another from time to time. This handoff may cause packet loss if not properly performed.

Most satellite channels only exhibit a subset of the above characteristics. Furthermore, satellite networks are not the only environments where the above characteristics are

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found. However, satellite networks do tend to exhibit more of the above problems or the above problems are aggravated in the satellite environment.

(from: Allman, Glover, Sanchez , [49], p. 4-5)

6.2.2.2.2 The impact upon TCP

Delays are problematic for TCP, since TCP's "standard" slow-start mechanism uses feedback from the receiver in order to increase the rate of injection of data into the network. Furthermore the slow-start algorithm is "conservative"; that is, the injection rate is increased quite slowly. Clearly, long round-trip times exacerbate TCP's slow-start behavior.

Since TCP is a loss-sensitive protocol, it cannot discern between losses due to congestion and those due to corruption (i.e., losses due to higher bit error rate (BER)). TCP's performance can greatly degrade when large BER values prematurely trigger window reduction mechanisms. This condition is further exacerbated by loss of ACKs in the reverse direction, as well as "bursty" errors (which are more common on satellite links) [55].

The large delay*bandwidth product (DBP) inherent in satellite links represents the maximum window size a TCP sender can achieve. This window size limits the amount of data a TCP sender is allowed to inject into the network before receiving an ACK. Metz [57] writes: "A larger [DBP] value translates to more buffers in the sender and receiver, as well as more time spent in the sub-optimal slow-start phase."

TCP also suffers greatly when asymmetric channels are employed for packet transmission. This is the result of ACK loss and "compression" [54], since ACK packets can get dropped or queued behind larger data packets [55].

Delay variance (jitter) are problematic for TCP, since its responsiveness to packet drops is reduced. Jitter can have a specific, negative effect upon TCP timer estimates: it renders these estimates more inaccurate. As a result, window sizes are improperly adjusted thereby decreasing overall bandwidth utilization [55].

Concerning the inclusion of satellite links along the data path, Ghani and Dixit [55] summarize as follows:

"Many studies have shown that satellites have a largely adverse impact upon TCP performance, with the main problems arising from increased bit error rates, and large propagation delays. Such conditions usually cause excessive TCP timeouts and retransmissions, resulting in large bandwidth inefficiency."([55], p. 64)

When the control signals for a service must be exchanged under such conditions, there is a certain risk that the service is at times slow to start and slow to respond. The degree to which this kind of service behavior/performance is experienced as *unsatisfactory* is dependent upon the nature of the service itself and the expectations of the user. It is *not* expected that the service may suffer complete breakdown during operation, though the possibility may still exist.

6.2.2.2.3 Other considerations

In a network containing satellite links, the performance of the transport protocol is not the only consideration. Data link protocol, application protocol, router buffer size, queuing discipline and proxy location are some of the other considerations that must be taken into account [49].

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6.2.2.2.4 Alternatives

To provide real-time, interactive and/or high bandwidth services via GEO sat environments, *knowing the inefficiency of TCP over satellite links*, there are basically three alternatives:

1. to stick with TCP/IP and look for techniques by which to improve TCP's performance
2. to use an alternative to TCP
3. to use an alternative to IP.

It is possible to combine certain of these approaches (e.g., by using gateways between links). The approach to choose is perhaps most dependent upon the *range* and *characteristics* of the services one wishes to provide.

6.2.2.3 Improving TCP performance

6.2.2.3.1 IETF work

There has been very significant work carried out within the IETF's TCP Over Satellite Working Group (TCPSAT WG)[71]. The Working Group is closed now, but its efforts led to the publication of two IETF RFCs. These RFCs concern approaches by which to modify TCP implementations so as to mitigate the effects that satellite links have upon the performance of TCP connections. The first RFC (Allman, Glover, Sanchez , [49]) describes modifications to TCP of which *all* are IETF standards-track mechanisms or otherwise compliant with IETF standards. The second RFC (Allman, et al., [50]) was written to educate researchers as to the current work and progress being done in TCP research related to satellite networks. The work is theoretical or experimental, and at the time of publishing, the algorithms and mechanisms outlined therein were not judged to be mature enough to be recommended by the IETF.

Several approaches for mitigating the effects of satellite upon TCP are found in [50]; these include:

- TCP For Transactions
- Slow Start
 - Larger Initial Window
 - Byte Counting
 - Delayed ACKs After Slow Start
 - Terminating Slow Start
- Loss Recovery
 - Non-SACK Based Mechanisms
 - SACK Based Mechanisms
 - Explicit Congestion Notification
 - Detecting Corruption Loss
- Congestion Avoidance
- Multiple Data Connections
- Pacing TCP Segments
- TCP Header Compression
- Sharing TCP State Among Similar Connections
- ACK Congestion Control
- ACK Filtering

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In the RFC, each of these approaches is described succinctly, yet clearly. In addition, information is also provided therein concerning:

- the sources of research in the specific area
- implementation issues
- network-topological considerations, and
- possible interaction and relationships with other research.

Readers wishing deeper understanding of these approaches are referred to [49] and [50].

6.2.2.3.2 Silicon TCP™

The only other work concerning *direct* improvement of TCP, which has been identified during this study, is work carried out by InterProphet [68]. All other approaches encountered thus far involve the *substitution* of TCP with other protocols (see section 6.2.2.3.3).

InterProphet has patented a technology called *Silicon TCP™*, which can be conceived as a sort of "TCP-in-hardware". InterProphet explains [69]:

Silicon TCP™ technology is an enhancement to existing Internet software technology. It removes the performance-limiting bottlenecks inside the software. Instead of forcing the processor to "move bits", it allows the bits to be moved by hardware under the advice of the software. Silicon TCP™ implements industry-standard protocols via dataflow techniques that generate and receive datagrams entirely without processor instructions of any kind, with the absolute minimum delay. Since the packets are constructed or decoded in lock step with physical layer signaling, one benefit of the technology is that the transport layer performance improves as the physical and link layers improve.

Instead of constructing packets in a store/forward memory buffer, Silicon TCP™ constructs packets "on the wire" as they are transmitted without any memory buffer (retransmission is accomplished by simply reconstructing the packet from scratch). As a result, the hardware generates in parallel physical, link, network, and transport layer functions to accomplish the send and receive of Internet packets.

InterProphet currently has three products. These target increased content storage and delivery rates. From the information available through InterProphet's WWW site, it is not immediately clear as to what degree these products could be used with great advantage across satellite links. It is certainly conceivable that overall network throughput could be dramatically increased; however, in order to make such determinations, further investigation is required.

6.2.2.3.3 Use of PEPs

Metz writes: "Specialized gateways (also called performance-enhancing proxies, or PEPs) can be positioned between terrestrial and satellite segments in the IP-forwarding path to optimize performance and reliability for end-to-end TCP application sessions." [57]

Two techniques commonly employed by PEPs include *TCP spoofing* and *TCP split connections*. Spoofing is carried out by the gateway on the sender side of the TCP connection. It involves the generation of "nicely" spaced ACKs to maintain a stable, open sending window (see Figure 31a). [57]

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In the case for network connections containing (one) satellite link, the split connections approach involves breaking an end-to-end TCP connection into three TCP connections, while doing so in a manner which is transparent to the end terminals (see Figure 31b). Metz explains:

"...the connection originating at the sender terminates at the local PEP, and the data is then mapped onto a separate inter-PEP connection using TCP or another link-optimized transport-layer protocol. The receiving PEP places the data on a separate connection to the TCP receiver. To minimize the bandwidth usage on the satellite link, most PEPs also employ some form of compression." ([57], p. 87)

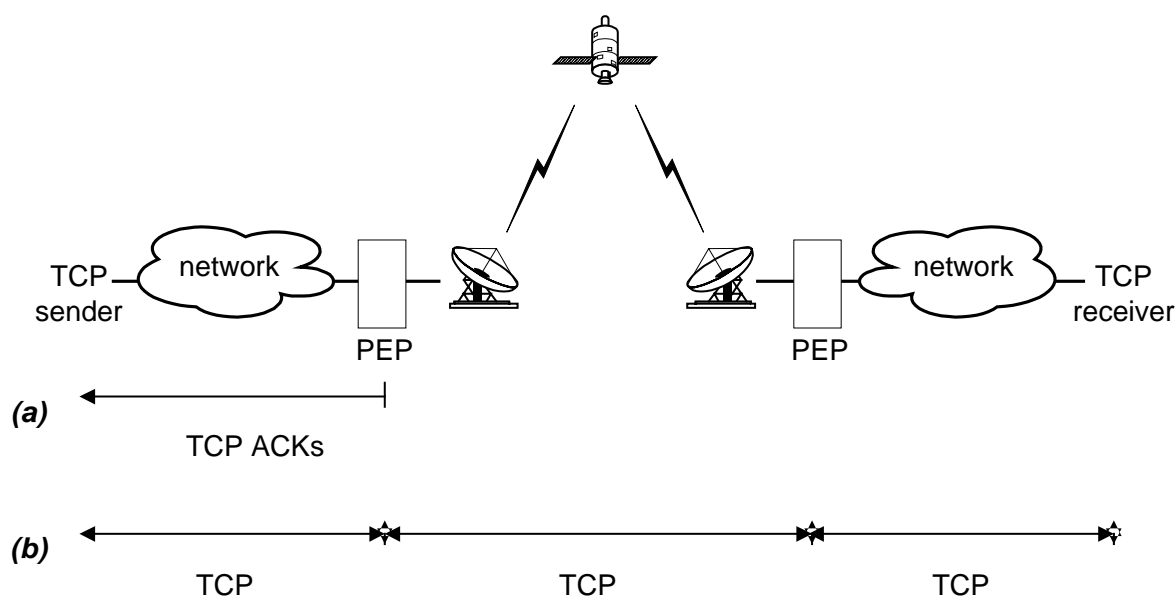


Figure 31: TCP PEP functions: (a) spoofing and (b) TCP split connections (adapted from [57])

In this document, the replacement of TCP over the satellite link (or over the entire path) is called *TCP-substitution*; it is described further below. It should be noted that Metz states that both Fourelle [91] and Mentat [97] have offered PEP solutions use split connections and satellite link-specific protocols ([57], p. 87). Further information about Fourelle and its products is included here in sections 6.2.1.1.6, 6.2.2.4.4 and 6.2.3.2.4. Further information about Mentat and its products is included here in sections 6.2.1.1.12, 6.2.2.4.2 and 6.2.3.2.2.

6.2.2.4 Alternatives to TCP

The second alternative by which to provide real-time, interactive and/or high bandwidth services via GEO sat environments — given the "deficiencies" of TCP in this environment — is to replace TCP as a transport protocol. In this document, this kind of approach shall be called *TCP-substitution*.

TCP-substitution can be carried out by replacing TCP in the protocol stack with a transport protocol more appropriate or optimized for networks that contain (GEO) satellite links. Most TCP-substitution approaches employ the use of other standard or proprietary protocols for packet transmission between gateways (or

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client-server proxy pairs) on each side of the satellite link(s). **This kinds of configuration is the best overall TCP-substitution approach, since it makes TCP-substitution completely transparent *outside* the gateways. Consequently, no special protocols need be installed / configured upon equipment or terminals outside the gateways.**

TCP-substitution can be carried out by a "clean" replacement of TCP, i.e., a replacement in which IP is still utilized as the network-layer protocol. Another kind of substitution of TCP-substitution can be performed in which both TCP and IP are replaced by some other kinds of transport- and network-layer protocols, perhaps even by some form of hybrid protocol (e.g., a protocol which integrates both transport and network-layers within a single packet-based implementation). Alternative use of IP is the subject of Section 6.2.2.5.

In the subsections which follow, four TCP-substitution protocols are described. The first is standards-based, the second is forum-based, and the latter two are proprietary. All protocols can offer *reliable transfer* — functionality which is a prerequisite for effective control signaling. The protocols also target more efficient utilization of network bandwidth.

6.2.2.4.1 Reliable multicast

Work upon reliable multicast is being performed by research groups affiliated with the IETF's Working Group (WG) for Reliable Multicast Transport. The basic principles underlying the approach, as well as a description of the work underway, is well available within the WG's charter [70]:

The WG will carry out protocol standardization by composing a set of RFCs that specify:

- building blocks: *A set of easily-separable coarse-grained modular components that are common to multiple protocols along with abstract APIs that define a building block's access methods and their arguments.*
- protocol instantiations: *Specifications that define the necessary gluing logic and minimal additional functionality required to realize a working protocol from one or more building blocks. These specifications will also include an abstract API that defines the interface between the protocol implementation and an application.*

To assist in this standardization process, the WG will also complete work on three documents. The first describes the design-space in which the three one-to-many transport protocols will be developed. The second explains the concepts of building-blocks and protocol instantiations. The third provides guidelines to authors of drafts that specify building-blocks and protocol instantiations.

The first of these documents (i.e., the design space document) is available as an RFC [56].

6.2.2.4.2 Xpress Transport Protocol (XTP)

XTP [72] is a forum-based standard from the XTP Forum. The XTP Forum [73] was founded as a non-profit organization in 1992, and Version 4.0 of the standard was released in March 1995 [74]. A working draft of the 4.0b specification of XTP (July 1998) is also available [75].

Of further significance, the SMPTE (Society of Motion Picture and Television Engineers [77]), is carrying out *active* work upon XTP, together with the XTP Forum. Currently, a working draft of the SMPTE XTP specification in SMPTE/ANSI/ISO format is available [76], dated August 2000. *This activity demonstrates*

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a move from a forum-based standard to an **international standard**. This condition alone is justification for a task to monitor the progress of SMPTE XTP and products which may eventually use this emerging standard.

As an alternative transport-layer protocol, XTP aims to alleviate "deficiencies" in TCP through four contributions: orthogonal protocol functions for separation of paradigm from policy, separation of rate and flow control, explicit first-class support for multicast, and data delivery service independence. The XTP version 4.0 specification [74] explains:

Separation of paradigm and policy

At the core of XTP is a set of mechanisms whose functionality is orthogonal. The most notable effect of this is that XTP clearly separates communication paradigm (datagram, virtual circuit, transaction, etc) from the error control policy employed (fully reliable through uncorrected). Further, flow and rate control as well as error control can be tailored to the communication at hand. If desired, any set of these control procedures can be turned off.

Separation of rate and flow control

Flow control operates on end-to-end buffer space. Rate control is a producer/consumer concept that considers processor speed and congestion. TCP does not provide rate control, and combats congestion with slow-start and other heuristics. XTP provides mechanisms for shaping rate control and flow control independently.

Explicit multicast support

Multicast is not an afterthought in XTP. Rather, each mechanism used for unicast communications is available for multicast use as well. The number of communicants is orthogonal to paradigm and policy.

Data delivery service independence

XTP is a transport protocol, yet with the advent of switched networks rather than routed internetworks, a traditional network layer service may not be appropriate in every instance. XTP requires only that the underlying data delivery service provides framing and delivery of packets from one XTP-equipped host to another. This could be raw MAC or IP or AAL5. XTP also employs parametric addressing, allowing packets to be addressed with any one of several standard addressing formats.

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Other features of XTP include:

- *implicit fast connection setup for virtual circuit paradigm*
- *key-based addressing lookups*
- *message priority and scheduling*
- *support for encapsulated and convergence protocols*
- *selective retransmission and acknowledgement*
- *fixed-size 64-bit aligned frame design*

(from Specification XTP Revision 4.0, [74], pp.1-2)

As explained in the quote above, XTP does not assume that IP is used as the underlying network-level protocol. XTP can be used as a TCP-substitution approach using gateways (or client-server proxy pairs) on each side of satellite link(s). XTP could also be integrated as a transport-level protocol within an end-terminal, for example, in cases where an end-terminal communicates directly with an earth station.

6.2.2.4.3 BST

BST is a proprietary protocol from Flash Networks [65]. It is used in Flash Networks' NettGain product-line. Flash Networks describes BST in the context of NettGain [66]:

NettGain uses a combination of multiple acceleration techniques to achieve consistently better performance over wireless connections. To achieve acceleration, prior to wireless transport, NettGain temporarily translates TCP packets into its own BST protocol for optimized transfer. At the far end of the link, BST packets are reconverted back to TCP. The conversion is completely transparent to users, without requiring any application modification. NettGain also applies real-time data compression to increase effective bandwidth on narrow wireless links.

From the information about BST available from Flash Networks' WWW site, it is not clear whether BST employs IP as the underlying network-level protocol. It is not unlikely that BST is some kind of hybrid protocol. Flash Networks' NettGain products can be used as a TCP-substitution approach using gateways (or client-server proxy pairs) on each side of satellite link(s). There is also some indication that certain BST / NettGain products can be used directly upon end-terminals such as laptops, mobile phones, and PDAs. The status of these products is unclear however, though the NettGain product for BST use directly upon the laptop (as an end-terminal) appears to be available. Further contact with Flash Networks is necessary in order to make clear determinations about product status.

6.2.2.4.4 Venturi Transport Protocol (VTP)

VTP is a proprietary protocol from Fourelle [91]. It is used in Fourelle's Venturi™ product-line. Fourelle describes VTP in the context of Venturi [67]:

Venturi reduces the amount of data transferred by compressing it and then transmitting it using an optimized IP-based protocol which Fourelle calls the Venturi Transport Protocol (VTP). The Venturi Server and the Venturi Personal Client or Venturi Workgroup Client software communicate to their respective applications using standard TCP/IP protocols, requiring no change to the standard application. At the same time, they transmit highly compressed data between them using VTP. To alleviate the delay caused by compression and decompression, the Venturi Server employs high-speed

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processors that devote significantly more CPU cycles to steadily streamlining data, thus removing strain from the network.

Here, it is stated that VTP uses "an optimized IP-based protocol"; hence, it is not completely clear whether BST employs standard IP as the underlying network-level protocol. Like BST, VTP could be some kind of hybrid protocol. Fourelle's Venturi "workgroup-level" products can be used as a TCP-substitution approach using client-server proxy pairs on each side of satellite link(s). Venturi "personal-client" product can be used as a TCP-substitution approach directly upon an end-terminal. Note: It is not made completely clear whether Venturi employs a TCP split connection approach, though this is implied by Metz (see section 6.2.2.3.3).

6.2.2.5 Alternatives to IP

The third alternative by which to provide real-time, interactive and/or high bandwidth services via GEO sat environments is to replace IP as a network protocol. In other words, to extract the IP payload without keeping the header. However, since the focus of this study is to *use* IP, not replace it, this topic is only briefly discussed below.

The concept of addressing is essential in services like VoIP for point-to-point telephony, fax and many other interactive services which target *specific* users or terminals. When it comes to providing the services above, the main feature of IP is to offer unique destination addresses (within the IP header). For broadcast services, unique addressing is obviously less important.

Currently the two best alternatives to IP are [59][60]:

- ATM, the encompassing telecommunication protocol designed for all kind of services
- MPEG-2, the multiplexing transport stream for digital TV

ATM over satellite is specified by ATM Forum in version 4.0. The protocol can be used in a variety of ways [61]. Either encompassing all protocol layers (ATM end-to-end), as a network protocol below TCP replacing IP, or even as a simple link layer protocol below IP. The latter is by far the most dominant approach.

MPEG-2 over satellite is mainly used for broadcasting digital TV, usually on top of a link layer protocol like ATM. However, MPEG-2 can in fact be used to carry IP packets within the transport stream; in other words, IP-over-MPEG-2-over-ATM [58]. Just like IP-over-ATM, this latter approach can obviously not be considered as an alternative to IP, since IP packets are transported all the same.

Historically other alternatives to IP have been devised, using packet based protocol such as X.25 or frame relay, for example. Although such approaches have been tested, the current benefits of this kind of approach are as questionable as ATM end-to-end. For compatibility reasons, "IP over everything, everything over IP" is the slogan of today.

The last alternative to IP is to transport the payload as a pure bitstream, without framing the stream into packets. This kind of approach is usually used for low bandwidth channels (e.g., from remote sensing satellites to earth, not communication satellites); most approaches also include some proprietary redundancy mechanisms for preserving the transport.

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Compared to the alternatives described here, IP is better suited for the kind of services focused upon in this study; alternatives to IP are not within the primary scope of this work.

6.2.3 Standards, Products and Maturity

Section 6.2.2 above has described technical aspects in general, with a specific focus upon the different approaches by which to cope with the inefficiencies of TCP in the effort to provide real-time, interactive and/or high bandwidth services via GEO sat environments. This section takes a look into the different products which are available which implement one or more of these kinds of approaches.

6.2.3.1 Improving TCP performance

6.2.3.1.1 IETF's TCPSAT

Two avenues of pursuit were mentioned in section 6.2.2.3. The first work presented was standards-based — work carried out by IETF's TCP Over Satellite Working Group (TCPSAT). A number of theoretical and experimental possibilities for direct improvement of TCP have been proposed by TCPSAT. This work has been carried out over for many years, though only under IETF auspices since ca. 1997. The work is very promising, but continues to require further maturation, large-scale testing and careful benchmarking. It is likely that some number of TCP acceleration products have used TCPSAT-related ideas and results in their products, and will continue to do so in the coming years. How this process will affect TCP developments in the IETF is difficult to predict.

6.2.3.1.2 Silicon TCP™

The second area of work concerning direct TCP/IP improvement was InterProphet's *Silicon TCP*-based product-line. Three products are currently available, targeting increased content storage and delivery rates. Further investigation is required here in order to determine:

- the degree to which these products could be used with great advantage across satellite links and
- the compatibility with technical solutions for emerging satellite systems.

6.2.3.2 Alternatives to TCP

Four TCP-substitution protocols were mentioned in section 6.2.2.3.3. These were:

- **Reliable Multicast**, from the IETF
- **XTP** (Xpress Transport Protocol), from the XTP Forum and SMPTE
- **BST**, from Flash Networks
- **VTP**, from Fourelle.

6.2.3.2.1 Reliable multicast

The IETF work upon Reliable Multicast is making good headway, proven by its first RFC [56]. Still, it may require between anywhere from 3-9 months or more before the building block-based protocol instantiations appear in RFC form. Products based upon these proposals *may* appear on the market sometime later, perhaps first from the companies represented by the RFC authors and contributors.

6.2.3.2.2 XTP

XTP has been specified for some time, and Mentat [97] markets two XTP-based products: Mentat XTP and Mentat's SkyX Gateway [98]. Mentat XTP is a high-performance, STREAMS-based, kernel-level

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implementation of XTP. It is available as a source code product on a variety of popular platforms and can be ported easily to any other operating system.

Mentat XTP *could* be integrated as a transport-level protocol within an end-terminal, but it is necessary to tailor applications to make use of this functionality. The Winsock API, used for networking in Windows applications cannot be used as interface to Mentat XTP. **Thus, it is not possible to directly integrate Mentat XTP as a transport-level protocol within an end-terminal, without making changes to existing applications.** Further, there is no known implementation of XTP for Microsoft Windows 95 & 98, only for NT and Windows 2000. It might be possible to develop a version of Winsock that interfaces with Mentat XTP. In general, however, the idea of using a TCP-substitution approach which requires installation of specific network libraries and application modifications upon end-user equipment is highly questionable.

Mentat's SkyX Gateway uses the SkyX Protocol, in place of TCP, over satellite links. Mentat's deep experience with TCP/IP may easily lead to the assumption that SkyX is a hybrid transport/network-level protocol, but this assumption has not been confirmed. It is not made clear whether the SkyX Gateway employs a TCP split connection approach, though this is implied by Metz (see section 6.2.2.3.3). On their WWW site, Mentat purports an impressive SkyX Gateway customer base [82], including Comsat, Gilat Communications, Intelsat and Hughes Electronics.

Mentat also offers references to results by independent testing organizations such as INTELSAT Technical Laboratories [79]; IGI and the Israel Inter-University Computation Center [80]; and NASA [78]. **Of importance here is the May 1999 IGI test in which the SkyX Gateway is compared to Flash Networks' SatBooster WWW proxy [81].** In that test, the SkyX Gateway significantly out-performed Flash Networks' product. Of course, that test was carried out ca. 1 1/2 years ago, and each company has had time to improve its products. Given the context of this study, it is highly relevant to acquire a new benchmark comparison of the two products.

6.2.3.2.3 BST

Information about the BST / NettGain products from Flash Networks is presented in section 6.2.2.4.3. Of importance here is the May 1999 IGI test in which the SkyX Gateway is compared to Flash Networks' SatBooster WWW proxy, see subsection 6.2.3.2.2 above.

6.2.3.2.4 VTP

Information about the VTP / Venturi products from Fourelle is presented in section 6.2.2.4.4.

6.2.3.2.5 TCP-substitution products: general comments

Products such as Mentat's SkyX Gateway, Flash Networks' NettGain Group Client and Fourelle's Venturi Workgroup Client employ TCP-substitution approaches which replace TCP in the protocol stack with other standard or proprietary protocols for packet transmission between gateways (or client-server proxy pairs) on each side of the satellite link(s). These kinds of product configurations make TCP-substitution completely transparent *outside* the gateways. Consequently, these products appear to be fully compatible with technical solutions for emerging satellite systems, since no special protocols need be installed / configured upon equipment or terminals outside the gateways.

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Products such as Mentat's XTP, Flash Networks' NettGain "Personal Client" and Fourelle's Venturi Personal Client *can* enable (product-dependent) protocol processing onboard a mobile end-terminal, without the need for an intervening gateway. This allows end-terminals the capacity to connect directly with an earth-station. For Mentat's XTP, the degree to which this requires special application configurations upon end terminals is described above in section 6.2.3.2.2. Determining whether similar implications exist for the other products requires further investigation.

All of these products advertize compatibility with existing satellite infrastructures, but it is highly recommended that end-terminal configuration requirements be investigated and factually tested prior to significant investments in any particular product.

6.2.3.3 Alternatives to IP

Since alternatives to IP are not within the primary scope of this work (see section 6.2.2.5), investigations into standards and products in this area was omitted from this study.

6.2.4 Special Issues

6.2.4.1 Degree of QoS achievable

QoS concerns:

- the kinds of quality aspects relevant to the *user's experience of services and their associated media* (see sections 6.3.2 and 6.3.3);
- the kinds of quality properties relevant to *signaling, transmission and processing of data traffic* (see sections 5.1 and 5.4);
- the kinds of protocols and mechanisms which can be applied to *influence and control* signaling, transmission and processing of data (see section 5.4.3, RSVP [51] and DiffSrv [52]); and,
- the kinds of *traffic and data guarantees which can be obtained* using such protocols and mechanisms (section 0).

Given the complexity and inter-relationships within the QoS arena, it is not possible to make general, yet meaningful statements about the degree of QoS which can be achieved within GEO satellite environments. It makes best sense to consider this subject in terms of *specific* services, in conjunction with the inherent *characteristics* and *requirements* of those services. Additionally, the user's own expectations and requirements must be taken into account. The reader is therefore referred to the discussions found in the sections noted above, where QoS and QoS-related matters are discussed in more well-defined contexts.

6.2.4.2 Security against eavesdropping

The proper manner by which to address different kinds of security threats is entirely dependent upon *explicit requirement specifications* for secure use of the system; depending upon system use, certain threats can be ignored while other threats must be thoroughly and correctly treated. This position is also stated by Allman, Glover and Sanchez [49] in regard to the threat of eavesdropping upon satellite links:

When using a broadcast medium such as satellites links to transfer user data and/or network control traffic, one should be aware of the intrinsic security implications of such technology.

Eavesdropping on network links is a form of passive attack that, if performed successfully, could reveal critical traffic control information that would jeopardize the

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proper functioning of the network. These attacks could reduce the ability of the network to provide data transmission services efficiently. Eavesdroppers could also compromise the privacy of user data, especially if end-to-end security mechanisms are not in use. While passive monitoring can occur on any network, the wireless broadcast nature of satellite links allows reception of signals without physical connection to the network which enables monitoring to be conducted without detection. However, it should be noted that the resources needed to monitor a satellite link are non-trivial.

Data encryption at the physical and/or link layers can provide secure communication over satellite channels. However, this still leaves traffic vulnerable to eavesdropping on networks before and after traversing the satellite link. Therefore, end-to-end security mechanisms should be considered. This document does not make any recommendations as to which security mechanisms should be employed. However, those operating and using satellite networks should survey the currently available network security mechanisms and choose those that meet their security requirements.

(from: Allman, Glover, Sanchez, [49], p.14-15)

6.3 Applicability of Voice, Video and Fax using IP over GEO Sat

Whereas section 6.2 focused upon the use of IP as a *general basis* for services within GEO satellite environments, this section focuses upon the applicability of an IP-based approach for three specific service-enabling protocols: H.323, SIP and T.38. The subsections below which cover this topic are organized as follows:

- Service-Enabling Protocols
- Service Characteristics
- Commercial Potential
- Threats / Expected Technical Problems and
- Possibilities for hybrid solutions using H.323 / SIP.

6.3.1 Service-Enabling Protocols

Most often, H.323 and SIP are associated with a “service” called VoIP. However, it is perhaps more useful to think of H.323 and SIP as *service-enabling protocols*. With this perspective, VoIP can be conceived as an enabling technology. That is, VoIP can be conceived as a technology providing functionality which can be embedded within higher-level services and applications — applications which can offer or utilize voice as an input / output medium. Examples of such applications include: point-to-point telephony, multi-party audio conferencing, voice-controlled applications, audio recording / playback, radio broadcasting, dissemination of (live) lectures / presentations, audio advertising, audio chat and more.

H.323 and SIP can also be used to implement an enabling technology referred to as “Video over IP”. Like VoIP, Video over IP can be used as a building block within services and applications which utilize video as an input / output medium. Examples of such applications include: “face-to-face” video-telephony, multi-party audio-video conferencing, video-on-demand, TV-style advertising, real-time instruction / assistance, and video recording / playback.

In comparison to H.323 and SIP as service-enabling protocols, T.38 is essentially specialized for one higher-level service: Fax over IP. For this reason, T.38 will receive little attention in the next two subsections below (i.e., the subsections which address service characteristics and commercial potential).

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6.3.2 Service Characteristics

Before discussing the commercial potential of services and applications which employ VoIP and/or Video over IP, it is necessary to consider the inherent characteristics of these services and their possible uses. The inherent characteristics of a service can include:

- cardinality, i.e., number of parties involved (e.g., one-to-one, one-to-many, many-to-many)
- party types (e.g., human—human, human—machine, machine—machine)
- primary information flow: bi-directional, uni-directional, n-directional (e.g., multicast)
- control flow paradigm(s) employed: request—response, notification, etc.
- types of media channels used
- *number* and *types* of media channels used simultaneously
- synchronization requirements between media channels
- *degree* and *rate* of interactivity / feedback
- traffic pattern (e.g., large data bursts, “smooth” stream of small packets, etc.)
- traffic symmetry (e.g., symmetric and near-symmetric, "lightly" asymmetric, "heavily" asymmetric, one-way).

6.3.3 Commercial Potential

VoIP, Video over IP and Fax over IP are three distinct enabling technologies, and at least VoIP and Video over IP can be employed in different ways. Using IP-based communication in GEO satellite environments, the overall quality of each of these enabling technologies — and consequently their respective commercial potential — is significantly influenced by the application context in which it is used. As mentioned earlier, the degree to which the quality of a service or application is experienced as excellent, satisfactory, tolerable or wholly intolerable, is dependent upon the nature of the service itself and the expectations/requirements of the user. For this reason, the perspective chosen here to review commercial potential and threats has three major aspects:

- to identify examples of possible commercial services and consider their inherent characteristics,
- to consider threats which stem from end-user tolerance and service perception (see *also* section 6.3.4.1) and
- to consider technical issues which threaten stability of the service (see section 6.3.4.2).

In Table 1, a collection of possible VoIP and Video over IP applications are presented. These are organized into six Application Groups:

- VoIP: Groups A-C
 - Group A: cardinality 1 – 1
 - Group B: cardinality 1 – n (“broadcast”)
 - Group C: cardinality n – n
- Video over IP: Groups D-E
 - Group D: cardinality 1 – 1
 - Group E: cardinality 1 – n (“broadcast”)
- Fax over IP: Group F.

For each of the individual applications in each group, qualitative information is provided about certain of their inherent characteristics. Most significant in this table, however, are the qualitative judgements about users’ tolerance to the effects of delay and noise within these various applications: it is precisely the user’s satisfaction or dissatisfaction with the behavior and performance of an application which will most strongly affect the application’s commercial potential.

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Some general observations can be made when reviewing the table. First, in non-broadcast application contexts, one can see an inverse relation between interactivity rate and tolerance for delay. That is, the higher the rate of interactivity, the lower the tolerance for delay. Users in broadcast contexts are basically indifferent to delay, since delay is essentially imperceptible during media transfer. However, users in broadcast contexts *can* be sensitive to delay when services are slow to start and slow to respond.

Another observation concerns the relationship between communication form / content, and tolerance for noise. That is, the less formal the communication form or content, the greater the tolerance for noise and errors (compare applications in Group C, for example). The table also illustrates that when the party types and direction of flow is "human <> human", data traffic tends to be symmetric over time, reflecting the flow of natural conversation and human interaction.

Lastly — though it is not reflected in the table — media synchronization requirements (i.e., lip- or gesture-sync) are generally very high in applications employing audio-video to display and present human users. When video content employs *audio narration* — rather than audio which is associated with a viewable speaker — media synchronization requirements are less stringent.

In section 7.2, more is said about the commercial potential of VoIP-based telephony services in GEO satellite environments.

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Table 1: VoIP / Video over IP applications and characteristics

	cardinality	party types / primary info flow	traffic symmetry	media types	media sync. reqmts.	degree / rate of interactv.	user's tolerance of delay	user's tolerance of noise / error	APPL GROUP
<i>standard telephony</i>	1 - 1	human <> human	symmetric	audio	none	very high	very low (a)	low (b)	A
<i>voice control</i>	1 - some few	human -> machine	"light" asymmetry	audio	none	low or high (c)	appl.dependent	very low (d)	
<i>audio recording / playback</i>	1 - 1	human <> machine	asymmetric	audio	none	low or med.	medium	med. or very low	
<i>radio broadcast</i>	1 - n	human -> human	asymmetric	audio	none	none	very high (e)	med. or very low	B
<i>lectures / presentations (live)</i>	1 - n	human -> human	asymmetric	audio	none	none	very high (e)	med. or very low	
<i>advertising (recorded)</i>	1 - n	machine -> human	asymmetric	audio	none	none	very high (e)	low (f)	
<i>audio chat</i>	n - n	human <> human	symmetric	audio	none	very high	low or very low	medium (g)	C
<i>standard multi-party calls</i>	n - n	human <> human	symmetric	audio	none	very high	very low (a)	low (b)	
<i>multi-party workshops (h)</i>	n - n	human <> human	symmetric	audio	none	med. or high	very low (i)	low or very low (i)	
<i>"face-to-face" video-telephony</i>	1 - 1	human <> human	symmetric	audio, video	very high	very high	very low (a)	low or very low	D
<i>real-time instruction</i>	1 - 1	human <> human	sym. / asym.	audio, video	high	med. or high	med. or very low	med. or very low	
<i>video recording / playback</i>	1 - 1	human <> machine	"heavy" asymmetry	audio, video	very high	low or med.	medium	med. or very low	
<i>video-on-demand</i>	1 - n	machine -> human	"heavy" asymmetry	audio, video	very high	none	very high (e)	very low	E
<i>advertising (recorded)</i>	1 - n	machine -> human	"heavy" asymmetry	audio, video	very high (f)	none	very high (e)	low (f)	
<i>Fax over IP</i>	1 - 1	machine <> machine	"light" asymmetry	data	none	very low or none	high or very high	none (j)	F

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Footnotes to Table 1

- (a) very low, since delays exceeding ca. 150 milliseconds begin to disturb the natural flow of oral, human communication
- (b) low, since there's a baseline, "standard" quality for voice calls which is well-known to all users of analog telephony
- (c) For voice control, interactivity is defined as a status response / notification which informs the user that the request has been successfully / unsuccessfully processed. The rate of interactivity is dependent upon the nature of the application, as well as the number of devices being controlled simultaneously; it is assumed that the interactivity rate demanded by the application limits the number of devices the human user is able to control simultaneously.
- (d) For voice control, it is likely that it is the voice recognition technology in the machine which cannot tolerate noise / error.
- (e) In broadcast contexts, users' tolerance of delay is due to the fact that they cannot perceive delay.
- (f) For advertising, the advertiser will likely demand high quality (low noise), while the listener / viewer is probably less concerned about quality.
- (g) Audio chat users would listen to / initiate conversations in an audio environment which likens a crowded room. Once conversation partners were identified, chat could be transferred to another, more private, audio channel (port); at that time, users would likely have tolerance for noise.
- (h) This use context concerns the execution of a work-oriented (rather than informal) discussion, where most parties contribute to the discussion.
- (i) Working in a multi-party audio environment requires a great deal of concentration, since one must both attend to the topic as well as the speaker's identity. Delays, noise and/or error in this context are extremely damaging to the overall work effort.
- (j) For fax, it is the machine which cannot tolerate error during data transfer. User's can often accept a significantly degraded version of the original document.

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6.3.4 Threats / Expected Technical Problems

6.3.4.1 Threats arising from users' experience of audio quality and interactivity

This section focuses exclusively upon VoIP, since it is the enabling technology of primary interest in this study.

The quality properties of GEO satellite-mediated connections are obviously different from those of earth-link mediated connections, especially for telecommunications services (Voice over IP). The delay induced by compression algorithms, network stacks and signal propagation time will be in the range 350 – 800 msec¹, considerably larger than ITUs recommendation of maximum 150 msec [62]. The caller (or client) may experience a response time of more than 1.5 seconds, which will considerably degrade the interactivity of the call/service. For a standard telephony service, this delay will seem especially long and disruptive for users unfamiliar with existing satellite-mediated telephony.

The jitter and loss in GEO sat environments seems to be smaller than for that for internet connections involving ca. 10-15 hops. Modern compression and packetization algorithms are designed to handle significant loss (above 1 %) gracefully. It is not expected that the amount of packet loss in satellite communication will have any significant impact on the user's experience of sound quality. Still, certain user groups may be dissatisfied in cases where they *expect* a service to deliver CD-audio quality, but the service is simply not configured to do so.

The effect of jitter is normally eliminated by buffering sound packets corresponding to a playout time equal to maximum expected jitter. As the jitter is expected to be around 1-4% of roundtrip latency, this technique can be utilized without adding noticeable delay overhead.

All audio compression schemes are lossy, in the sense that the decompressed signal differs from the original sound. This results in speech which is slightly harder to understand on the receiving side; still, the quality of current compression algorithms is fairly good and hardly noticeable when compared to standard GSM mobile phones. It should be noted that media packets passing through transcoding gateways will suffer from multiple compression loss, perhaps resulting in noticeable losses which affect the perceived quality of the media. **Still, the major threat to perceived speech quality and service response will arise from delay induced by signal propagation time.**

This kind of threat is independent of the VoIP protocol employed.

6.3.4.2 Threats arising from quality of setup and signaling

This section focuses upon H.323 rather than SIP, since H.323 is the protocol of primary interest in this study. Furthermore, there has not been time to thoroughly assess the threats arising from quality of setup and signaling using SIP.

¹ This value for delay is larger than that given in section 5.4.1, since it also includes audio compression and network stacks.

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The deviating quality characteristics of satellite-mediated internet links are expected to have some impact on the perceived quality of media data in IP-based voice traffic. An interesting question is whether the expected loss and delay for these links will have impact on the signaling used in H.323 and H.320.

H.323 uses the Transport Control Protocol (TCP) for peer-to-peer signaling. TCP is known to have performance problems on links with high loss or high roundtrip time (RTT), see section 6.2.2.2.

6.3.4.2.1 Bandwidth and TCP window size

The expected amount of loss in a GEO sat environment is not particularly high. Since TCP only is used for signaling, the bandwidth limitations due to the window size problem will not have any impact on H.323.

6.3.4.2.2 Delay and timers

One general problem with TCP is that the data acknowledgement timeout time is set to 120 sec initially. If the first packet is lost, this will take two minutes to discover by the protocol. With the relatively low expected loss rate, this will not happen very often. We can expect that the user or application handles this gracefully if it ever happens.

The question is then whether the high delay has any impact on the signaling and timeout values in H.323 and H.320 themselves. H.323 and H.320 and their underlying standards contain several procedures using timeout to recover from problems and errors.

H.323 uses H.245 and H.225 as control channel protocols (e.g., during call setup). H.225 includes the use of RAS and Q.931 control protocols.

6.3.4.2.2.1 H.323 and H.245

H.245 defines 9 timers for different procedures. The exact values for these are left for standards on higher level (H.321-324) to define. H.323 states that:

All timers defined in Recommendation H.245 should have periods of at least as long as the maximum data delivery time allowed by the data link layer carrying the H.245 Control Channel, including any retransmissions. ([17], section 6.2.8.5)

Whether or not the actual implementations of H.323 clients all conform to this is not known. The standard internet socket API does not convey the delivery time to the application, but there is sufficient functionality to implement an application level solution with acceptable accuracy.

6.3.4.2.2.2 H.323 and H.225

H.225 defines timers for RAS signaling. These are shown in Table 2, which summarizes the recommended default timeout values for the response to RAS messages and subsequent retry counts if a response is not received.

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Table 2: H.225.0 – Recommended default timeout values

RAS message	Timeout value (sec)	Retry count
GRQ	5	2
RRQ (Note 1)	3	2
URQ	3	1
ARQ	5	2
BRQ	3	2
IRQ	3	1
IRR (Note 2)	5	2
DRQ	3	2
LRQ	5	2
RAI	3	2
SCI	3	2
<p>NOTE 1 - The time-out value should be recalculated based upon both the time-to-live (which may be indicated by the Gatekeeper in the RCF message) and the desired number of retries.</p> <p>NOTE 2 – In cases where the gatekeeper is expected to reply to an unsolicited IRR with IACK or INAK, the timeout may occur if no reply to the IRR is received.</p>		

As seen in the table, some of these timeout values are defined to at least 5 seconds, others to at least 3 seconds. **Messages employing response times of less than 3 seconds could become delay-critical in network topologies where the RAS channel crosses both a satellite link and the Public Internet** (see Figure 30). Should the receiver (i.e., gatekeeper) be performing disk operations, for example, responses to the client/sender could arrive more than 3 seconds after the original request was sent.

The most important RAS message having a recommended timeout value of three seconds is the RRQ message. **Repeated failure of this message completely prevents an endpoint from registration with the network’s Gatekeeper².** Two retries are recommended for the RRQ message, but this provides no *absolute* guarantee that the message is received and processed in time.

Other sensitive RAS messages in the table are URQ ("Unregister Request") and IRQ ("Information Request"), since these have recommended timeout values of three seconds and each includes only one retry. The URQ message can be initiated by either the Gatekeeper of the endpoint; its purpose is to unregister the endpoint with the Gatekeeper. For an endpoint, it is necessary to unregister in order to change the alias

² It is explained in the table that the RRQ timeout value should be recalculated, but this recommendation can only be exercised once the *response* to RRQ (i.e., the RCF message) has been received.

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associated with its Transport Address, or vice-versa. Otherwise, the endpoint will remain registered with the Gatekeeper until its `timeToLive` value expires.

IRQ messages may be used by the Gatekeeper in order to request usage information from an endpoint for a particular call. This may be used, for example, in order to audit an endpoint to potentially discover the call capacity of that endpoint. The function of the IRQ message is certainly less critical than the RRQ and URQ messages; that is, it is extremely unlikely that H.323 connections will fail or be terminated due to IRQ timeouts.

H.225 specifies that gatekeepers “knowing” that they might be late in responding can issue the *Request In Progress* (RIP) RAS message, which causes the client to restart its timer at the original value. **Only gatekeepers that can be configured to issue RIP responses should be used in GEO sat environments.**

It is also important that the physical location of the Gatekeeper be taken into account, with respect to the configuration of the H.323 client. Consider Figure 30, for example. In this illustration, it is possible that user C (at the mobile satellite terminal) has an H.323 client which is internally-configured to register itself (RRQ) with a specific Gatekeeper somewhere within a private network. In an aberrant situation, the distance in *time* to the Gatekeeper *could* exceed the timeout value for this message, even during the second retry. Thus, one should thoroughly consider the ramifications of pre-configuring clients with fixed Gatekeeper addresses, before doing so.

6.3.4.2.2.3 H.323 and Q.931

Q.931 is a protocol used within H.225. H.225 includes use of a Q.931 setup timer:

The "setup timer" T303 (see Tables 9-1/Q.931 and 9-2/Q.931) defining how long the calling endpoint shall wait for an ALERTING, CALL PROCEEDING, CONNECT, RELEASE COMPLETE or other message from the called endpoint after it has sent a SETUP message. This timeout value shall be at least 4 seconds. Note that some applications may appear in networks which have inherently longer delays (for example, compare the Internet to a local enterprise network or intranet).([4], section 7.5)

Although the values used for this timer in actual implementations is not known, **it is possible that signaling failures arising from T303 timeouts could occur when Direct Endpoint Call Signaling is used** (but, if so, only very infrequently). **It is even less likely that such failures would occur when Gatekeeper Routed Call Signaling is employed** (since the messages in question are received first at the Gatekeeper, then *re-issued* to the sending / receiving endpoint). It is not known to what extent the frequency of any eventual failures may be significant to planned target system performance and/or application areas.

6.3.4.2.2.4 H.320

The H.320 standard builds on H.221 (among others) for signal framing and H.242 for control signal definitions. H.320 does not by itself define any timeout values. H.242 defines a few timers. All timers regarding response waiting time have a minimum value of 10 seconds. The summary of the H.221 standard contains the following phrase:

- *...[H.221 defines] a synchronous procedure. The exact time of a configuration change is the same in the transmitter and the receiver. Configurations can be changed at 20 ms intervals. [20]*

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When communicating through a H.323-gateway with partner(s) over a GEO sat link, this certainly cannot be fulfilled. It is reason to believe that this can be hard to achieve in ordinary ISDN networks, and that the phrase "*the exact time ... is the same*" ought to be taken with a pinch of salt. Even if H.320-systems may accept a rather loose definition of "*the same time*", achieving a sufficient degree of synchrony may be difficult when the roundtrip time exceeds one second.

6.3.4.3 Threats to commercial value of Fax over IP

This section exclusively addresses threats arising from the commercial deployment of fax over data networks; Fax over IP is a special case of such deployment.

One reason that fax still is in use is due to the partial, juridical validity of faxed documents. This is based on the fact that traditional, paper-based fax only can create "remote" paper *copies* of "real" paper documents. Consequently, any signature received in a fax is presumed to be an authentic copy of a non-forged real signature.

This condition is a major benefit of fax when comparing it to competing technologies such as e-mail and phone calls. Enabling computer-to-computer exchange of digital information representing a fax originating from paper will enable the receiver to save a fax-quality, digital copy of any handwritten signature in the document. Thus, other persons' signatures can easily be digitally copied and "pasted" into other documents by malicious persons. Ironically enough, making transmission of facsimile documents available on computers, on a large scale, can undermine the present juridical validity of fax, and thereby reduce one of its significant competitive advantages in the market.

6.3.5 Possibilities for Hybrid Solutions using H.323 / SIP

When it comes to interoperation between the H.323 and SIP protocols, both Cisco [63] and Lucent [83] have developed products which provide integration and interoperation of these technologies; other products may certainly exist as well. Both companies appear to integrate these technologies into their own, proprietary products [64] [84], rather than providing gateways for conversion to SIP from H.323 or vice-versa. This kind of integration scheme almost undoubtedly has its basis in each company's powerful legacy technology; in other words, it was certainly faster and easier to integrate these VoIP protocols into *existing* systems, rather than to build new products "from scratch". Investing in these kinds of products may lead one into a certain degree manufacturer-dependence. On the other hand, however, it is probably *certain* that these products perform well, given the proven strength of these manufacturers.

Concerning international standardization efforts in the area of H.323 / SIP interoperation, work has also begun within the IETF [48]. At this stage, the ongoing work is much in the area of requirements specification. Therefore, it will probably be *at least* 1-2 years before the first products which incorporate IETF standards for H.323 / SIP interoperation reach the market.

Other, more experimental work is also underway by Ackermann et al. [47]. Their approach is to use the Delivery Multimedia Integration Framework (DMIF) within MPEG-4 as a framework within which alternative kinds of VoIP protocols and interfaces can be integrated with one another. It appears that this work has begun quite recently; therefore, it is impossible to say whether or when products utilizing (aspects of) this approach may reach the market.

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6.4 Possibilities for QoS Provisioning in GEO Sat Systems

This subject is treated in section 5.4.3.

6.5 Recommendations for “Follow up” Activities

6.5.1 Highest Priority

6.5.1.1 RAS signaling and timeouts

Certain concerns exist in regard to eventual timeout conditions in protocols used by H.323, especially RAS signaling (see section 6.3.4.2.2). Recommendations include:

- to test with a prototype which supports RAS signaling, such that experiments can be conducted which investigate H.323 stability in the face of RAS timeouts;
- to determine critical values for delay within different use case topologies (see sections 5.3.2 and 6.3.4.2);
- to execute large scale simulations and perform statistical analysis of the results, in order to estimate the rate of H.323 failures resulting from timeout conditions.

6.5.1.2 TCP improvements

To deliver interactive IP-based services in GEO satellite environments, delay and bandwidth utilization must be seriously addressed. Gateways that *effectively* manage PEP functionality (see section 6.2.2.3.3) may greatly alleviate problems associated with TCP delay and bandwidth utilization. Mentat and Fourelle’s systems are both logically organized as PEPs. Both claim to improve bandwidth utilization, which is quite surely the case. **The degree to which they reduce TCP delays (e.g., via TCP split connections) is also important to determine³. Equally important to determine are the scalability and robustness of these systems,** should one wish to employ them.

Recommendations in this area include:

- to establish a clear, performance benchmark for the existing prototype configuration, both with and without TCP accelerator;
- if possible, benchmark Mentat and/or Fourelle’s gateways in place of a TCP accelerator and cross-compare performance.

6.5.2 Second Priority

Other approaches to improving TCP exist and are in progress. In this regard, recommendations include:

- to determine whether and how InterProphet’s Silicon TCP technology could be used with advantage in a network containing satellite links;
- continuously monitor developments upon standards-based TCP-substitution approaches: XTP (SMPTE), Reliable Multicast, IETF approaches.

Should one be interested in pursuing provision of interoperable H.323 - SIP services, recommendations include:

³ Clearly, delays associated with signal propagation over the satellite link have a theoretical limit.

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- to investigate products from Lucent and Cisco;
- to compare functionality, configuration, performance and cost of these products;
- to continuously monitor developments and new products in this area.

7. CONCLUSIONS OF THE STUDY

7.1 Technical Feasibility

Having studied the relevant standards and protocols, it is completely reasonable to believe that H.323, SIP and T.38 can be made to operate quite well within GEO satellite environments. Depending upon the level of failure-free operation one wishes to attain, however, certain concerns exist in regard to eventual timeout conditions in protocols used by H.323, especially RAS signaling (see section 6.3.4.2.2).

Recommendations about addressing this situation are presented above in section 6.5.1.1.

7.2 Service Quality and Commercial Potential

As indicated in sections 6.3.3 and 6.3.4.1, the major hinder impacting wide-scale exploitation of *highly interactive* VoIP- and Video over IP-enabled services is delay arising from signal propagation over the satellite link. For such services, this delay manifests itself as slow response or slow feedback from the service. For a VoIP-enabled standard telephony service, this condition is experienced by users as relatively long delays within conversational exchange, which have a tendency to disrupt the natural flow of interactive dialogue. Such delays will be less of a problem for users experienced with existing satellite-mediated telephony.

For other interactive VoIP- and Video over IP-enabled services, conditions of slow response can lead to users experiencing the service to be unsatisfactory or wholly intolerable. For *less* interactive services (e.g., broadcast), delay conditions may be negligible or wholly irrelevant to users.

Despite the conditions arising from delay, certain market segments and user groups will still be interested in using such services. Rough categorizations of these segments / groups can be :

- those who trade off service quality for price;
- those for whom there exists “no alternative means” for which to obtain such services;
- those for whom alternatives exist, but the immediate use of satellite-based services offers the “least effort”⁴.

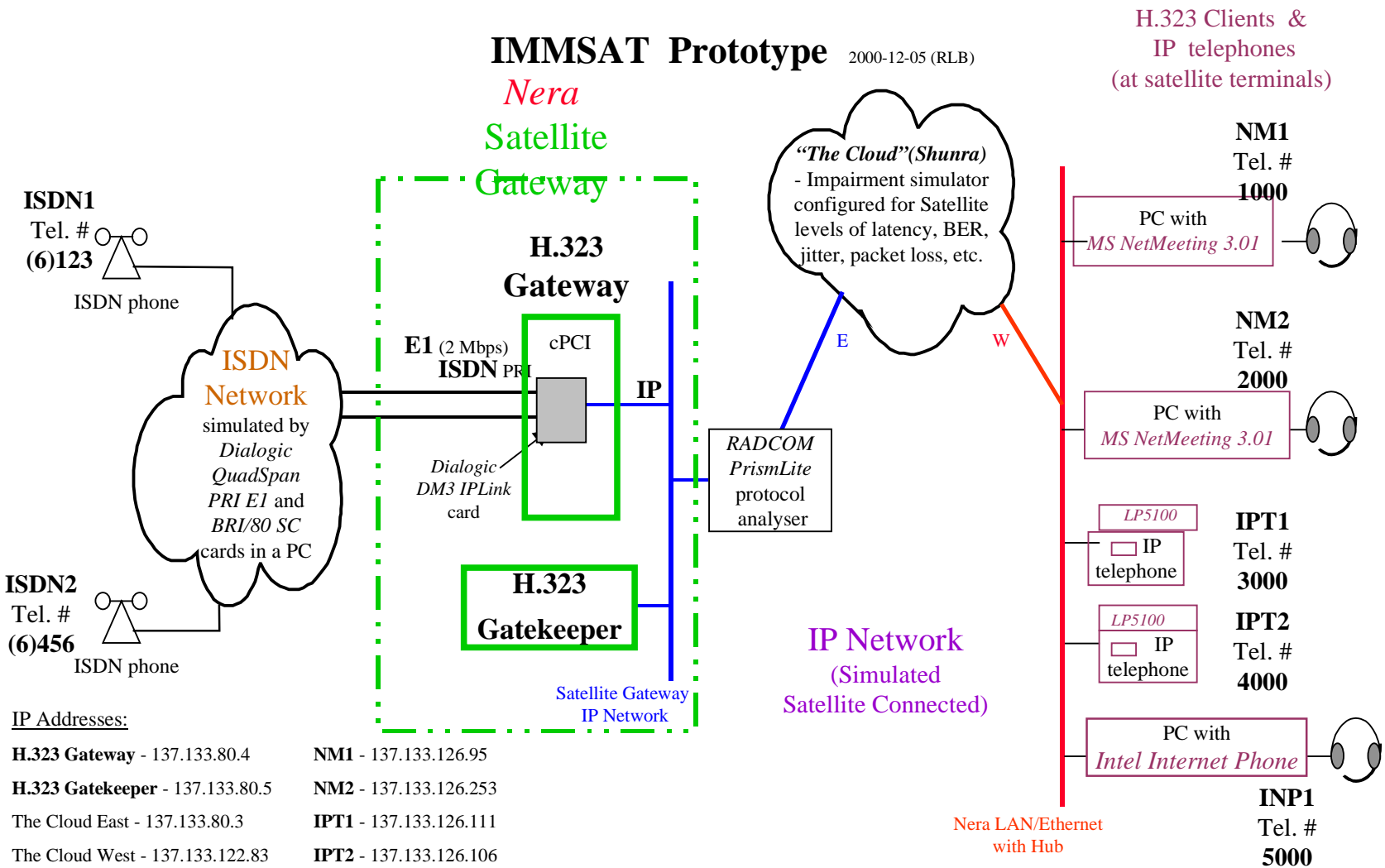
Final conclusions about the commercial potential of VoIP- and Video over IP-enabled services within GEO satellite environments rest upon the business model and role(s) of the organization planning to deploy / support such services.

⁴ In these cases, the other alternative means for communication are judged — *at that time and moment the service is needed* — as requiring more effort to set-up / establish / configure, etc., when compared to the quality requirements for particular service in question.

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IMMSAT Prototype 2000-12-05 (RLB)

Nera
Satellite Gateway



IP Addresses:

- | | |
|--|-------------------------------|
| H.323 Gateway - 137.133.80.4 | NM1 - 137.133.126.95 |
| H.323 Gatekeeper - 137.133.80.5 | NM2 - 137.133.126.253 |
| The Cloud East - 137.133.80.3 | IPT1 - 137.133.126.111 |
| The Cloud West - 137.133.122.83 | IPT2 - 137.133.126.106 |

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8. PROTOTYPE AND TESTING

8.1 Introduction

In order to validate some of the recommendations from the theoretical study, an H.323 based prototype with a simulated GEO satellite environment was assembled. A number of tests using VoIP equipment and protocols were carried out.

Due to unavailability of codecs in the H.323 Gateway equipment for video and limited feasibility and time for testing Fax, the prototype testing was focused on the VoIP service.

The major findings are summarized in this chapter. A detailed description of all aspects of the prototyping and testing, including all the test results, can be found in the IP Multimedia over Satellite Test Report[106].

8.2 Objectives

The main objectives of the IMMSAT prototype testing were:

- 1) *Verify that the H.323 protocols for the VoIP service can be used successfully in an environment that approximates the characteristics of a GEO satellite based IP network.*
- 2) *Find the parametric limits for latency, jitter, and packet loss/BER where the VoIP protocols stop functioning or provide an unacceptable service for placing calls.*
- 3) *Verify that the voice quality for the VoIP service is acceptable when operating in an environment that approximates the characteristics of a GEO satellite based IP network.*
- 4) *Find the parametric limits for latency, jitter, and packet loss/BER where the VoIP voice quality becomes unacceptable.*

8.3 Configuration

The IMMSAT Prototype configuration is shown in the diagram on the previous page.

The main system components of the H.323 based IMMSAT Prototype are a Dialogic H.323 Gateway and, an H.323 Gatekeeper. Enhanced versions of these two H.323 elements will be placed in the Nera Satellite Gateway in future Nera solutions providing VoIP service.

A simulator called “the Cloud” is used to simulate the satellite environment. Modifying the configuration values for latency, jitter, packet loss, and BER in “The Cloud” can change the satellite environment.

The terminals or user equipment are connected via Ethernet to “The Cloud”. The H.323 clients and IP telephones are normally connected to a satellite terminal/modem for access to the satellite based IP network. The H.323 clients/IP telephones that were used in the IMMSAT prototype testing included two Siemens IP telephones and two MS NetMeeting H.323 SW Clients (vers. 3.01).

Dialogic ISDN HW and SW were ported to a PC that supports ISDN PRI, BRI, and NT1 entities in order to simulate an ISDN network in the IMMSAT prototype. Two ISDN telephones are attached to the ISDN NT1

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entity that is connected to the simulated ISDN. This allows for calls to be setup between ISDN phones via the H.323 Gateway with any of the H.323 clients.

A Radcom PrismLite protocol analyzer was used to capture protocol messages that were exchanged between H.323 GW, GK, and Clients during testing. The Prism Lite provides a limited H.323 protocol decoding functionality that allows for protocol analysis of the most popular protocols.

8.4 Types of Testing

In order to achieve the above objectives, two primary types of tests were conducted: *call setup* and *audio quality*.

The objective call setup tests were aimed at meeting the first two objectives, whereas the subjective audio quality tests would try to fulfil the last two objectives.

All testing was performed using various pairs/combinations of H.323 clients, IP telephones, and ISDN telephones. For each combination of client/telephone, the following satellite environment parameter values was varied one at a time, independently of the others:

Latency
Jitter
Packet Loss
BER

A gatekeeper routed call signalling model was employed, although some early testing was done only with an H.323 gateway and no gatekeeper. Voice encoding and decoding was fixed to G.711, in the ISDN/H.320 network, and to G.723.1 (6.4Kb/s), in the IP/H.323 network. Transcoding is and was during the prototype testing always performed in the H.323 Gateway.

Parameters for a typical GEO satellite environment are:

- 520-580 ms one-way delay (of which the satellite propagation delay accounts for 250 ms)
- 10^{-6} average bit error rate (BER)
- 1% average packet error (assuming BER above and 100 byte packets on average)

8.5 Results

The results of the IMMSAT prototype testing are presented in the IP Multimedia over Satellite Test Report [106].

A summary of the results is presented below.

Regarding *applicability of H.323 protocols in a GEO Sat. environment*:

The call setup tests showed that the protocols worked under severely impaired conditions that were much worse than in typical GEO satellite environments.

Establishing calls could be done successfully and reliably for most client combinations, even with 3-4 seconds delay and 10-20% packet loss. The most critical situations occurred with *delay* for calls between two different types of IP terminal/client. Call setups began to fail when the delay at the satellite simulator was set to around 1.5 seconds for these client combinations. However since the prototype is based on the

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assumption that the GEO satellite cannot do “switching in the sky”, the messages of the call setup must travel over the satellite two times. With two hops at 1.5 seconds, the delay approaches the same limits as for the other client combinations.

For all combinations, BER had little influence on call setup. Note that in commercial implementation of satellite based IP networks, TCP/IP packets with bit error(s) are discarded anyway. An assumption was made for this testing phase, that since call setup has no *relative* timing requirements for packet arrival, varying the jitter parameter values will not impact the call success rate. Therefore, the jitter parameter was held constant.

Regarding *quality of H.323 based services such as VoIP*:

The voice quality tests included two aspects, *listening* quality and *conversational* quality.

For one-way listening quality, the tests showed that *jitter* was the most destructive parameter, especially when we configured other than a long jitter buffer in the IP terminal. For the Siemens LP5100 telephone, the voice quality began to degrade when the jitter parameter was set to 50 ms or more. The NetMeeting H.323 client fared better with a limit for jitter of around 150 ms.

Fortunately, in all tests, a normal distributed model for jitter was used instead of a uniform delay function. A normal distributed model is probably a more demanding model than what is actually observed in GEO satellite environments. Packet loss was the second most influential factor. Repeatedly, a 10% limit degraded the voice quality severely. Again, BER had little influence, and delay was not tested at all, as it was considered a *constant* not affecting listening quality.

For conversational quality, delay is obviously an important issue. Generally, circuit switched based telephone conversation is carried out in an environment which has less than 150 ms one-way delay [62], which is inherently impossible in GEO satellite environments. Thus, users have to adapt to the large delay conditions and the unnatural "single duplex" conversation mode.

All of the tests showed that it **is** possible to converse meaningfully even with a 3-4 second delay. However, many users might only use the service if the price is “right” or if VoIP is the only alternative for service. Conversational quality was also affected by what is suspected to be an incompatibility in silence suppression coding / encoding between audio codec implementations at the Dialogic GW and Siemens IP telephones. Calls from other H.323 clients to the Siemens LP5100 telephone through the gateway generated high amplitude “fax-like” noise, which made any conversation impossible. Calls originated from the LP5100 to other types of H.323 clients were noise free and of good quality. For all other client pairs, the quality was judged to be satisfactory or with only minor levels of noise.

Regarding *validity of recommendations from theoretical study*:

The tests showed that the recommendations still hold and were in fact backed up by the testing results. As expected, protocol timeout values proved to be essential, but not critical for call setup in GEO satellite environments. TCP improvements are still important, even though there were not sufficient resources or time to implement such improvements so that it could be tested in the lab prototype this time around.

An additional recommendation is proposed as a result of the analysis of the testing results – solutions should be investigated to avoid or compensate for the destructive effect of uncontrolled jitter that may occur when offering a VoIP service in a GEO satellite based IP network. Even though the extreme conditions tested in

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the prototype are not likely to be observed in commercial GEO satellite environments, the ability to handle jitter is crucial for offering an acceptable VoIP service when calling between different IP terminals.

Generally, all tests performed indicated that VoIP as a service is practical and usable, long before the limits of the protocols are reached and breakdown occurs. Also, the parameter limits are far outside (on the order of one magnitude) of that which is typical in GEO satellite environments.

8.6 Final Conclusion of the Prototyping and Study

The most important and final conclusion of the IMMSAT project - first derived from the work done in the study and then confirmed in the prototyping and testing - is that an H.323 VoIP service can operate quite well in GEO satellite based IP networks.

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