An Investigation into the Effects of GEO Satellite Environments upon H.323-based Voice over IP

Norwegian Computing Center
APPLIED RESEARCH AND DEVELOPMENT

RAPPORT/REPORT

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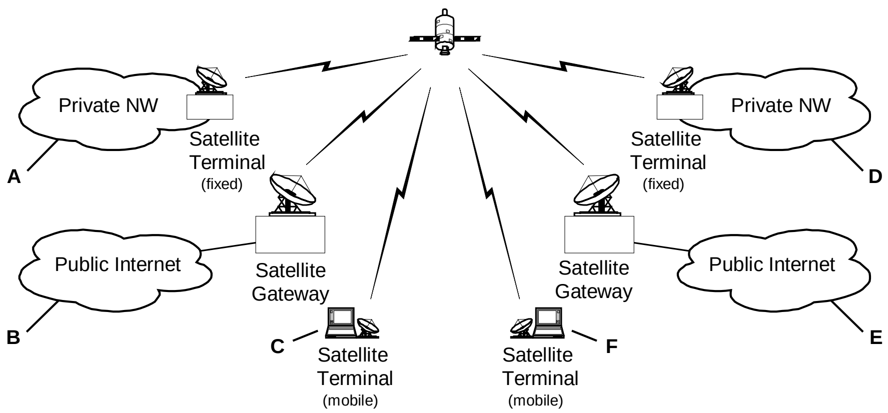

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- **Sammendrag**/Abstract:

Abstract

The work presented in this report was performed within the IMMSAT project, a project carried out during Q2-Q4 2000 by Norsk Regnesentral [21] and Nera SatCom AS [23].

IMMSAT comprised both a theoretical and an empirical study of technical aspects concerning "*Internet Protocol based MultiMedia Services over a Satellite Network*". Specifically, the project focussed upon technical aspects involved in the delivery of Voice over IP (VoIP), Video over IP, and Fax over IP within a GEO satellite-based environment.

Originally, the empirical work within IMMSAT intended to look into the effects of satellite characteristics such as latency, jitter, packet loss and bit error rate (BER) upon all three XoIP services. Due to limited feasibility and time for testing, however, prototype testing was focused upon an H.323-based VoIP service.

The primary objective during prototype testing was to validate, while simulating GEO satellite network characteristics, that H.323 protocols and an H.323-based VoIP service could successfully be used in a GEO satellite-based IP network. The primary types of VoIP tests carried out during prototype testing were *call establishment* and *audio voice quality*.

For these test types, the investigation proceeded with an effort to identify the parametric limits for latency, jitter, and packet loss/BER where:

- the H.323-based protocols could no longer provide an acceptable service for *establishing* VoIP calls (i.e., where call establishment becomes severely impaired and/or ceases to function in a timely manner) and
- the VoIP *audio voice quality* becomes unacceptable (i.e., where audio voice quality degrades such that usability of the service is severely and/or wholly impaired).

This document presents the outcomes of prototype testing, along with an accompanying analysis. Conclusions are also drawn about the robustness and quality for H.323-based VoIP services via satellite.

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1. Background

1.1 Trends

The general trend in computing today is digital convergence, with IP technology introduced everywhere, on all platforms and systems on the market. There is more widespread use of wireless and mobile services and networks, and an increasing degree of multimedia content is being delivered. At the same time, a variety of different "last-mile" broadband technologies are also being deployed.

The effects of this trend are readily apparent in satellite-based systems, since characteristics central to convergence are inherent in such systems: IP is one of the basic protocols to be employed, communication is wireless (and terminals often mobile), and the pipeline is broad. Many existing and future satellite-based systems are being designed or adapted with the aim to provide efficient and cost-effective media communication.

Both terrestrial and satellite-based systems can offer real-time, interactive and/or high-bandwidth services. Furthermore, the trend is to offer the *same* services in both systems. However, the technological differences within *geosynchronous* earth orbit (GEO) satellite-based systems (e.g., inherently large delay and bit error rate in the satellite link) makes systems employing such links better suited for low interactive, broadcast services. Whether satellite-based or not, systems planning to offer real-time, interactive high-bandwidth services face the technological challenge of reserving resources and providing end-to-end QoS guarantees across the network.

1.2 Commercial aspects

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:

- 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.

Some of the companies involved in this business are listed in Table 1 below.

Table 1: A selection of companies involved in media transmission via satellite

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/

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 listed use exclusively *GEO satellite constellations*. Furthermore, not *all* products, solutions and services available via 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).

1.3 Challenges

Bem, et.al. [5] provide a thorough introduction to the area and issues of broadband satellite multimedia systems, clarifying the basic physical and architectural distinctions between low earth orbit (LEO), medium earth orbit (MEO), highly elliptical orbit (HEO) and geosynchronous earth orbit (GEO) satellite systems. Their survey also provides clarifications concerning some of the technical and legal issues to be addressed. The technical issues therein include:

- *continued access to the services via fixed and mobile terminals*
- *continuity of service*
- *reduction of the power radiated by fixed and mobile terminals*
- *adequate quality of service (QoS)*
- *adequate capacity of the system.*

(Bem, et.al. [5], p. 3.)

Farserotu and Prasad [7] provide a more brief survey, and include concise descriptions of the basic issues and references to the latest work in areas such as enhancements to TCP/IP [1][2][8], enhanced QoS awareness, IP security over SATCOM, onboard processing, switching and routing, and service enabling platforms (i.e., middleware). Another concise introduction to basic issues in IP-over-satellite may be found in Metz [10].

2. The IMMSAT Project

The work presented within this report was performed within the IMMSAT project, a project carried out during Q2-Q4 2000. Norsk Regnesentral [21] and Nera SatCom AS [23] were the participating partners in IMMSAT; Nera was the main contractor, and Norsk Romsenter [24] was a partial sponsor.

IMMSAT comprised both a theoretical and an empirical study of technical aspects concerning "*Internet Protocol based MultiMedia Services over a Satellite Network*". Specifically, the project focussed upon technical aspects involved in the delivery of Voice over IP (VoIP), Video over IP, and Fax over IP[1] within a GEO satellite-based environment.

2.1 Scope

The IMMSAT theoretical study, reported in [3], provides general descriptions of the many signaling and transmission protocols relevant to this area. In addition, it addressed certain QoS issues for GEO satellite-based environments. It also presented recommendations and criteria for using IP as basis for XoIP services in GEO satellite-based environments. Specifically for VoIP, it examined both H.323 [11] from ITU-T and SIP (Session Initiation Protocol) [13] from the IETF.

Originally, the *empirical* work within IMMSAT intended to look into the effects of satellite characteristics such as latency, jitter, packet loss and bit error rate (BER) upon all three XoIP services. For VoIP, the intention was also to perform empirical testing and comparisons of H.323 and SIP under variations in (simulated) satellite conditions. Due to unavailability of video codecs from the vendor of the H.323 Gateway equipment and limited feasibility and time for testing Fax, however, *prototype testing was focused upon an H.323-based VoIP service.*

2.2 Objectives of the Prototype

In order to validate some of the recommendations from the IMMSAT theoretical study [3], an H.323 based prototype was assembled together with a simulator of characteristics of a GEO satellite-based environment. A number of tests using VoIP equipment and H.323 protocols were carried out.

One preliminary objective was to gather and document first-hand knowledge of the effects of satellite characteristics upon an H.323-based VoIP service. The primary objective, however, was to validate that H.323 protocols and an H.323-based VoIP service can successfully be used in GEO satellite-based IP networks.

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[1] Jointly, these services can be referred to as *XoIP services*.

Two primary types of VoIP tests were carried out during prototype testing:

1. call establishment and
2. audio voice quality.

The main objectives of the call establishment testing were:

- To verify that the *H.323 protocols* for the VoIP service can be used to establish calls in an environment that approximates the characteristics of a GEO satellite-based IP network.
- To find the *parametric limits* for latency, jitter, and packet loss/BER where the H.323-based protocols can no longer provide an acceptable service for establishing VoIP calls (i.e., where call establishment becomes severely impaired and/or ceases to function in a timely manner.)

The main objectives of the audio voice quality testing were:

- To verify that the *audio voice quality* for the VoIP service is acceptable when operating in an environment which approximates the characteristics of a GEO satellite-based IP network.
- To find the *parametric limits* for latency, jitter, and packet loss/BER where the VoIP audio voice quality becomes unacceptable (i.e., where audio voice quality degrades such that usability of the service is severely and/or wholly impaired).

3. Technical Foundation

To understand the underlying issues and analyses discussed in this report requires a basic understanding of H.323 and GEO satellite-based systems. The next two sections provide such an introduction. A more comprehensive H.323 tutorial can be found at [12].

3.1 H.323: Packet-based Multimedia Communications Systems

ITU-T recommendation H.323 [11] was originally entitled "*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. At present, most commercially available H.323 products adhere to version 2.

3.1.1 H.323 entities

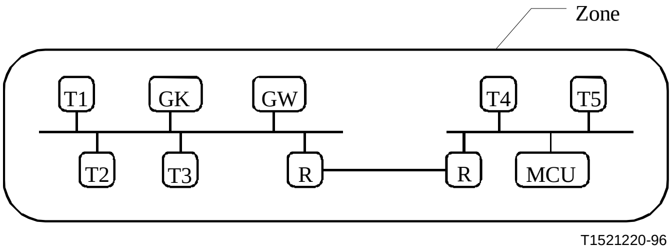

Figure 1: H.323 entities (from [11])

H.323 defines the following entities (see Figure 1):

- **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:** Logical "area" containing the Terminals, Gateways, etc. which are controlled by the same Gatekeeper[2].

3.1.2 Basic signaling procedures for H.323 call establishment

H.323 uses H.245 [15] and H.225 [18] as control channel protocols during call establishment. H.225 includes the use of RAS and Q.931 control protocols.

The provision of H.323-based communication, from call setup to termination, proceeds in 5 phases:

- phase A: call setup
- phase B: initial communication and capability exchange
- phase C: establishment of audio visual communication
- phase D: call services and
- phase E: call termination.

It should be understood that there exist timers and timeout values associated with messages transmitted during each of these phases; many of these timers are defined in

[2] The original definition for 'zone' was: a Local Area Network segment(s) used for H.323 communication and controlled by a single gatekeeper.

H.225 and its subprotocols. It is exactly these timers and timeout values which affect the successful establishment and termination of calls. Further details about timers and signal timing constraints are presented later in section 4.2.2.

In IMMSAT, the call establishment testing concerned itself specifically with the success or failure of phases A-C. These phases are described briefly below; descriptions of phases D and E (as well as detailed descriptions of phases A-C) can be found in [11].

In phase A, RAS and H.225 messages [18] are used to achieve initial call setup for each endpoint. Once these endpoints are connected, they establish the H.245 [15] call control channel (phase B). The procedures of H.245 are used over this channel for the exchange of terminal capability sets, such that the endpoints can determine which kinds and formats of media streams shall be sent and received by each endpoint. Master/slave determination is also resolved in this phase.

Phase C begins with the use of H.245 procedures aimed to open logical channels for the various media streams. Other call establishment procedures in phase C concern media stream address distribution, correlation of media streams in multipoint conferences and communication mode command procedures. For more details, see [11].

When phase C has been completed, a call may be said to be fully established. During IMMSAT testing, call establishment was judged successful when voice could be transmitted between the endpoints.

3.1.3 Basic call models

The call establishment phases A-C can be carried out according to several alternative *call models*. Call models are essentially distinguished from one another by the entities to and from which call signaling and control messages are exchanged.

When no Gatekeeper exists within a domain, endpoints are free to (attempt to) perform call setup and establishment messaging directly with one another. When a Gatekeeper does exist, however, other kinds of call models are usually enforced by the Gatekeeper. One simple kind of call model is that of *Direct Endpoint Call Signaling*. In this model, each endpoint exchanges RAS messages with the Gatekeeper, in order to gain admittance to the network. Once admitted, the remaining H.225 call setup and H.245 call control messages are exchanged directly between the endpoints. This kind of call model is depicted in Figure 2.

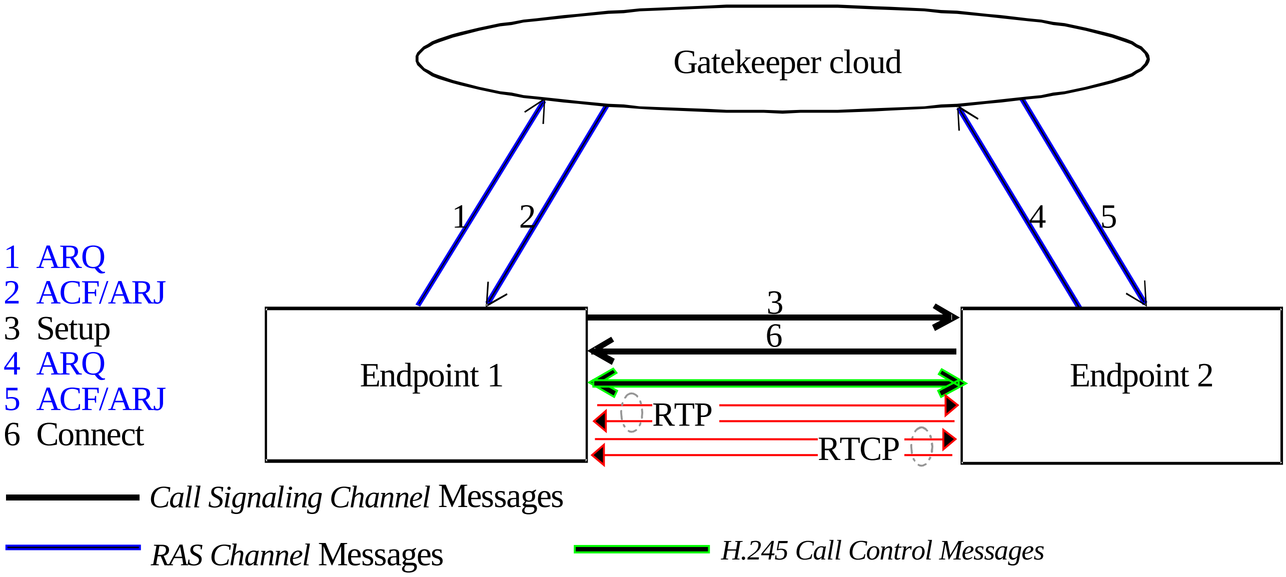

Figure 2: Direct Endpoint Call Signaling (adapted from [11])

The temporal ordering of the call setup messages is listed on the left-hand side of the figure. The RAS messages are shown as blue arcs, and the H.225 messages are shown as black arcs. The H.245 call control channel is shown as a green arc, and is established as the result of successful completion of phase B. This H.245 call control channel is used in phase C, in order to open the logical channels for various media streams, which are depicted as red arcs.

Another, more common kind of call model used in Gatekeeper-controlled Zones is that of *Gatekeeper Routed Call Signaling*. In this call model, all RAS and H.225 call setup messages between endpoints are routed via the Gatekeeper. The advantages to this kind of call model are greater overall control of the call (and call services), and greater robustness. This latter condition applies also to *Direct Endpoint Call Signaling via Gateway* models, and is explained in section 3.1.4.

In the Gatekeeper Routed Call Signaling model, the Gatekeeper is responsible for deciding whether the H.245 call control channel should also be routed via the Gatekeeper, or whether it should be established directly between the endpoints. Figure 3 depicts a call model for Gatekeeper Routed Call Signaling, with Gatekeeper-routed H.245 control.

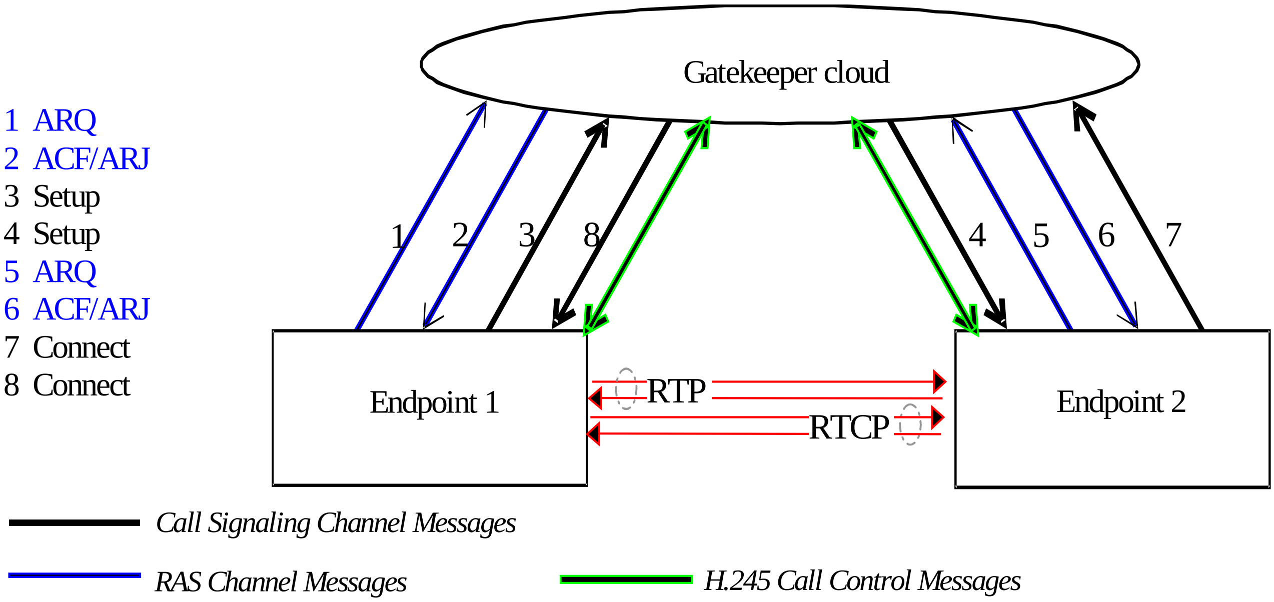

Figure 3: Gatekeeper Routed Call Signaling with Gatekeeper-routed H.245 control (adapted from [11])

3.1.4 Signaling and timing details within call setup

This section briefly introduces certain important signaling and timing details within the call setup phase (phase A, see section 3.1.2). These details apply to both call models described above. They are specifically illustrated below with respect to

a) a Gatekeeper Routed Call Signaling model, and

b) a Direct Endpoint Call Signaling model in which call setup signaling goes via a Gateway.

Call signaling within the call setup phase is slightly more complex than as illustrated in the call models depicted in Figure 2 and Figure 3. The call setup signaling which can occur within the Gatekeeper Routed Call Signaling model is shown in Figure 4; this diagram assumes that both endpoints are already registered with the same Gatekeeper.

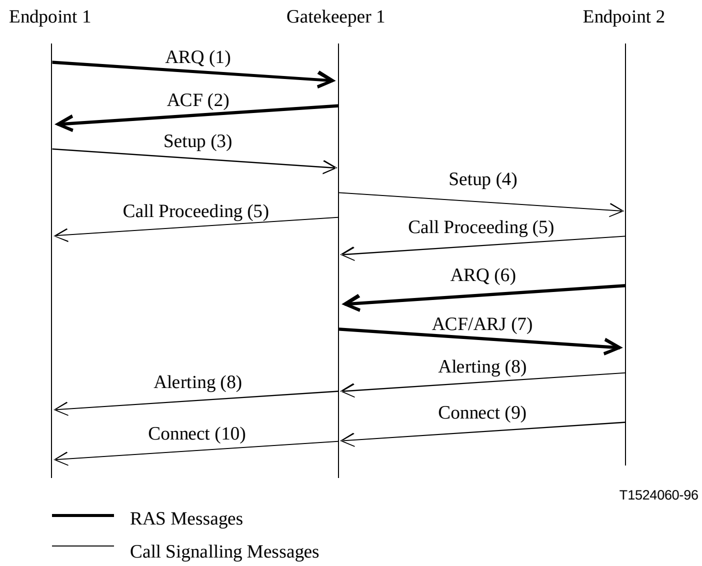

Figure 4: Gatekeeper routed call signaling: both endpoints registered, same Gatekeeper (from [11])

Figure 4 illustrates that *Call Proceeding* and *Alerting* signals may also be exchanged during the call setup process. The *Call Proceeding* signal can be used to inform the calling endpoint (e.g., endpoint 1) that call setup processing is underway. The *Alerting* signal is used to inform endpoint 1 that the terminal at endpoint 2 is ringing. In this figure, both the Gatekeeper and callee are illustrated as sending *Call Proceeding* and *Alerting* signals.

For a Direct Endpoint Call Signaling model in which call signaling goes via a Gateway, the call setup signaling pattern is similar, though slightly different. In this case, the RAS messages are exchanged between the endpoints and the Gatekeeper, while the call signaling messages are exchanged between the endpoints and the Gateway. In this situation, the Gateway is a callee endpoint with respect to endpoint 1, and a calling endpoint with respect to endpoint 2. This signaling pattern is depicted in Figure 5.

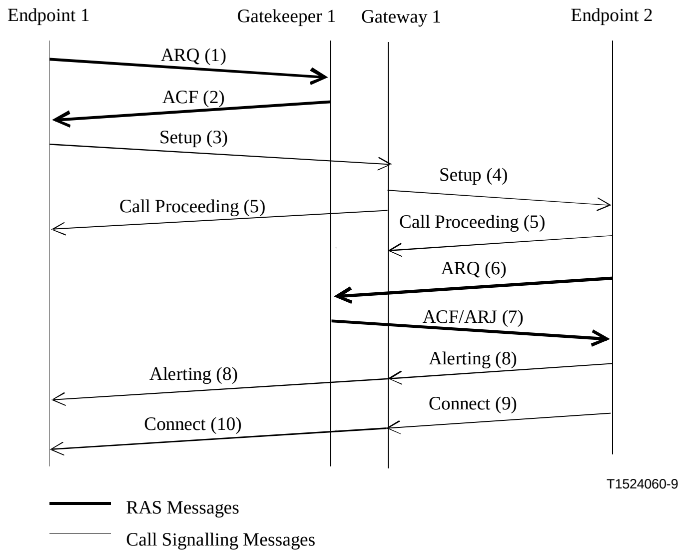

Figure 5: Direct endpoint call signaling via Gateway: both endpoints registered, same Gatekeeper (adapted from [11])

3.1.4.1 Timers

These extra signals (i.e., *Call Proceeding* and *Alerting*) are especially important with respect to timers used during call setup. Once the calling endpoint has sent a *Setup* message, it shall start the T303 timer and expect to receive an *Alerting*, *Call Proceeding*, *Connect*, *Release Complete* (or other message) from the called endpoint [18]. By default, the value for the T303 timer should be at least four seconds (see section 4.2.2.3).

Should the calling endpoint receive a *Call Proceeding* signal before T303 timeout, a T310 timer should be started. The T310 timer runs until the calling endpoint receives an *Alerting*, *Connect* or *Release Complete* signal. Typical timeout values for the T310 timer are about 10 seconds [22]. If an *Alerting* signal is received before a T310 timeout, a T301 timer ("establishment timer") should be started. This timer runs until a *Connect* or *Release Complete* signal is received. This timeout value shall be 180 seconds (3 minutes) or greater [18].

3.1.4.2 Mandatory vs. optional signaling

H.225 requires the callee endpoint to send an *Alerting* signal. In a Gatekeeper Routed Call Signaling model, H.225 also requires that a Gatekeeper forward the *Alerting* signal (e.g., in Figure 4, to forward the signal to the calling endpoint). H.225 also states that a Gateway should also forward the *Alerting* signal (as shown in Figure 5).

Of significance for call setup timing constraints, H.225 does *not* require that the callee endpoint send a *Call Proceeding* signal. That is, endpoint 2 is not required to send this message in either call model. If it *is* sent, H.225 states that Gatekeepers are required to forward it, and that Gateways should forward it (depending upon call model).

Interestingly, H.323 requires the Gatekeeper to send a *Call Proceeding* signal to the calling endpoint, whenever the Gatekeeper judges that it might require more than four seconds to respond to that endpoint. This same H.323 requirement applies to Gateways when call setup signaling transpires via a Gateway.

Thus — as required in all call models — the calling endpoint shall receive an *Alerting, Call Proceeding, Connect, Release Complete* (or other message) from the called endpoint within four seconds after it has sent a *Setup* message. In a Direct Endpoint Call Signaling via Gateway model, however, there is *no* guarantee that a *Call Proceeding* signal is sent from endpoint 2 to the Gateway. Without the reception of a *Call Proceeding* signal from endpoint 2, the Gateway will wait (by default) at most four seconds between the time at which the it sends *Setup* (4) and the time at which it receives *Alerting* (8); otherwise, the call setup procedure fails[3].

With respect to H.323-based VoIP delivered over a GEO satellite environment — an environment which inherently includes packet delay, distortion and loss — use of the *Call Proceeding* signal by the callee endpoint can be highly valuable in all H.323 call models. In short, use of this signal helps serve to alleviate, though not fully decouple, the end-to-end call setup timing constraints. That is, issuance of this message from callee endpoints helps mitigate the effects of multiple satellite hops during H.323 call setup.

3.2 GEO satellite-based network environments

Bem, et.al. [5] provides a concise summary of the definitions and basic characteristics of LEO, MEO, HEO and GEO satellite systems. *The focus of this report is GEO satellite-based environments.* Thus, this work primarily addresses problems and issues for such environments. Still, certain of the issues raised here are relevant for satellite-based environments as a whole. More complex issues and challenges such as on-board processing, on-board switching, intersatellite links, etc. [7][10] — challenges naturally associated with non-GEO environments — are not within the scope of this discussion.

3.2.1 Use case topologies

It is important to take into consideration the use case topologies for GEO satellite-based networks, since there exist several distinct alternatives which influence the overall quality and performance of the services they deliver. Certain ground-based switching alternatives are illustrated in Figure 6. These alternatives involve the use of satellite terminals and satellite gateways.

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. A *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.

[3] H.225 and Q.931 allow the *Setup* signal to be retried. Since TCP underlies the H.225 call signaling messages, however, many implementations choose to clear the call, rather than retry *Setup* [22].

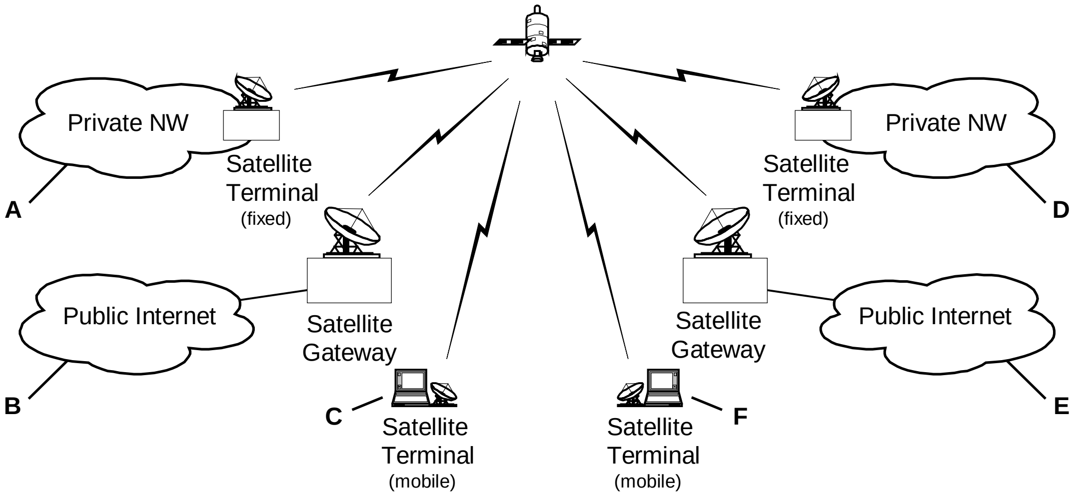

Figure 6: Alternative topologies for GEO satellite-based network environments

Figure 6 illustrates — on each side of the satellite link — three qualitatively different "paths" by which to access the satellite. The major qualitative difference amongst these paths is that of the end-to-end performance of the networks. For example, the Public Internet cannot provide any hard guarantees about the timeliness, smoothness, etc. of packet delivery from end-to-end. For private networks, it is at least more feasible to deploy and/or obtain such guarantees and, in turn, services which exhibit better overall performance.

3.2.2 Satellite hops and audio quality

Another factor which influences the overall audio quality of a (GEO) satellite-based service is the number of satellite hops which are necessary in order to realize that service. Consider within Figure 6 the case wherein a VoIP connection between C and F requires one type of audio format at one terminal and a second type of audio format at the other terminal. Since the GEO satellites existing 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.

In this kind of case, the audio media from each terminal must be transferred *twice* over the satellite link. The audio media from C is first 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 kind of "two hop" case, delays, as well as other quality aspects of the signal (e.g., noise) can significantly increase in magnitude.

4. Expected threats and problems

During the course of the IMMSAT project, the scope of the work became progressively restricted due to resource and prototyping limitations (see section 2.1). Ultimately, H.323-based VoIP was selected as the primary service in focus. Thus, the material presented in this section specifically concerns both possible and expected threats to audio quality and call establishment for H.323-based VoIP services provided over GEO satellite-based networks.

4.1 Threats concerning users' experience of audio quality

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 msec, considerably larger than ITUs recommendation of maximum 150 msec [17]. 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. Quite simply, delays of this magnitude (1.5 seconds) will *significantly* affect the manner in which conversation is accomplished. *Delays of even greater magnitude may render successful conversation impossible.*

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 when compared to the audio quality delivered through 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.

The jitter and packet loss in GEO satellite-based environments seems to be smaller than 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 audio quality. Still, certain user groups may be dissatisfied in cases where they *expect* a VoIP service to deliver CD-level audio quality.

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. *Large, uncontrolled variations in latency (high degree of jitter) would seriously degrade the users' capacity to perceive speech.*

4.2 Threats concerning call establishment

In addition to the impact on the perceived quality of voice in IP-based voice traffic that is discussed in section above, the impact on the signaling used in H.323 and H.320 [14] due to packet loss and delay for satellite-mediated internet links is another issue to be considered.

H.323 uses the Transport Control Protocol (TCP) for peer-to-peer signaling. TCP is known to have performance problems on links with high packet loss or high roundtrip time (RTT) [1].

4.2.1 Bandwidth and TCP window size

The expected amount of packet loss in a GEO satellite-based environment is not particularly high. Since TCP only is used for signaling, the bandwidth limitations due to the window size problem [1] will not have any impact on H.323 call establishment.

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 [15] and H.225 [18] as control channel protocols (e.g., during call establishment). H.225 includes the use of RAS and Q.931 control protocols.

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.* ([11], 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.

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.

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
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. 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.		

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 6).

The most important RAS message having a recommended timeout value of three seconds is the RRQ message. *Repeated failure of this message prevents an endpoint from registration with the network's Gatekeeper*[4]. Without successful registration, an endpoint in not able to place nor receive calls. 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 associated with its Transport Address, or vice-versa. Otherwise, the endpoint will remain registered with the Gatekeeper until its `timeToLive` value expires.

H.225 specifies that gatekeepers "knowing" that they might be late in responding to

[4] 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.

messages can issue the *Request In Progress* (RIP) RAS message, which causes the client to restart its timer at the original value. It is perhaps advisable that *only gatekeepers which can be configured to issue RIP responses be used in GEO satellite-based 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 6, 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 a (highly) aberrant situation, the distance in *time* to that Gatekeeper *could* exceed the timeout value for this message. Thus, one should thoroughly consider the ramifications of pre-configuring clients with fixed Gatekeeper addresses, before doing so.

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).*([18], 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 Call Proceeding is sent by the Gatekeeper to the calling endpoint if the GK judges that it may take more than 4 seconds for a reply from the other endpoint (see 3.1.4.2). It is not known to what extent the frequency of any eventual failures may be significant to planned target system performance for call establishment.

4.2.2.4 H.320

The H.320 standard [14] builds on H.221 [19] (among others) for signal framing and H.242 [20] 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.

5. IMMSAT Prototype Configuration

As mentioned in section 2.1, the scope of the empirical work within IMMSAT became focused upon the effects of satellite characteristics upon an H.323-based VoIP service. Interworking with and the interfaces to terrestrial networks such as the ISDN/PSTN were also of primary interest in IMMSAT.

The major network components and protocols/standards for such a network are shown in Figure 7.

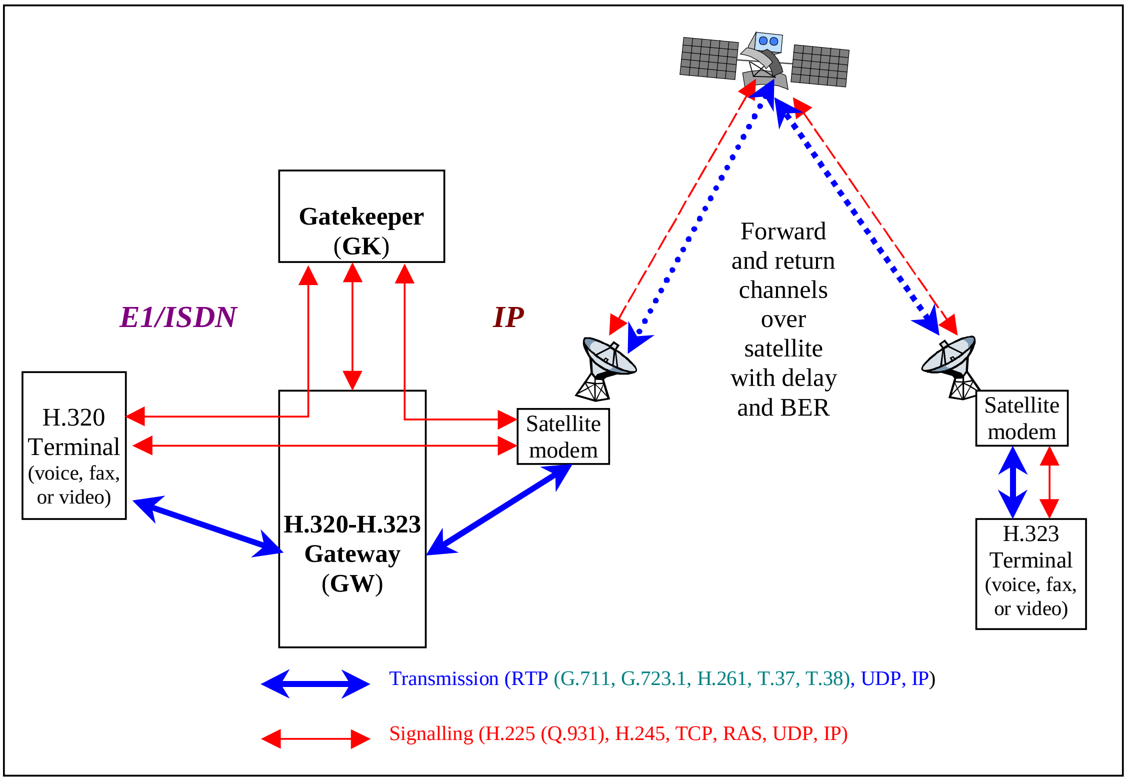

Figure 7: Network components for H.323-based XoIP over GEO Satellite

The satellite-based environment parameters in focus within IMMSAT were *latency, jitter, packet loss and bit error rate (BER).* In a typical GEO satellite-based environment, values for these parameters are:

- *Latency:* 520-580 ms one-way delay, of which the satellite propagation delay accounts for 250 ms; the remainder arises from packet processing within the involved network elements
- *BER:* 10^{-6} average bit error rate
- *Packet Loss:* 1% average packet error (assuming 10^{-6} BER and 100 byte packets on average).

This remainder of this section describes the main system components and network configuration for the H.323-based IMMSAT prototype.

5.1 Prototype elements

The main system components of the IMMSAT Prototype were a Dialogic H.323 Gateway and a demo version of an H.323 Gatekeeper created by RADCOM. The H.323 GW was based on IPLink HW and Dialogic system release 5.0 SW. It also included a Motorola cPCI platform and a Dialogic DM3 IPLink card (cPCI) [25].

A WAN simulator called "the Cloud" (version 2.1) [29] was used to simulate the satellite environment. Modifying the configuration values for latency, jitter, packet loss, and BER in "the Cloud" simulated changes in performance within the satellite environment.

The H.323 clients/IP telephones used in the IMMSAT prototype testing included two Siemens LP5100 IP telephones [28] and two Microsoft NetMeeting H.323 SW Clients (version 3.01).

Dialogic ISDN HW (QuadSpan) and SW [26] was also employed in order to simulate an ISDN network in the IMMSAT prototype. Two Siemens Profiset 30 ISDN telephones [27] were attached to the ISDN NT1 entity which was connected to the simulated ISDN. This allowed for calls to be establishment between the ISDN phones and any of the H.323 clients, via the H.323 Gateway.

In addition, the same Dialogic QuadSpan equipment on this PC platform was used to perform "playback" of pre-recorded voice files. One such file was used as a reference in the listening tests described below.

A Radcom PrismLite protocol analyzer [30] was used to capture protocol messages that were exchanged between H.323 GW, GK, and Clients during testing. The PrismLite provides a limited H.323 protocol decoding functionality which allows for protocol analysis of the most popular protocols.

5.2 Network Configuration

The network configuration of the IMMSAT prototype is given in Figure 8. The H.323 client types are indicated with **NM** for the Microsoft NetMeeting 3.01 clients, **IPT** for Siemens IP telephones, and **ISDN** for ISDN phones. These acronyms are followed by a number (1 or 2) distinguishing between different entities from same vendor.

The configuration provides for *two hop* satellite calls, as will be the case for calls between two H.323 clients/IP telephones. For such IP-to-IP calls in a satellite-based IP network, all signaling and transmissions must go from one H.323 client over the satellite to the Satellite Gateway (where calls are "switched") and then over the satellite again to the other H.323 client (see section 3.2.1).

The configuration also provides for *one hop* calls, which are the calls between ISDN phones and any of the H.323 clients/IP telephones, via the H.323 Gateway.

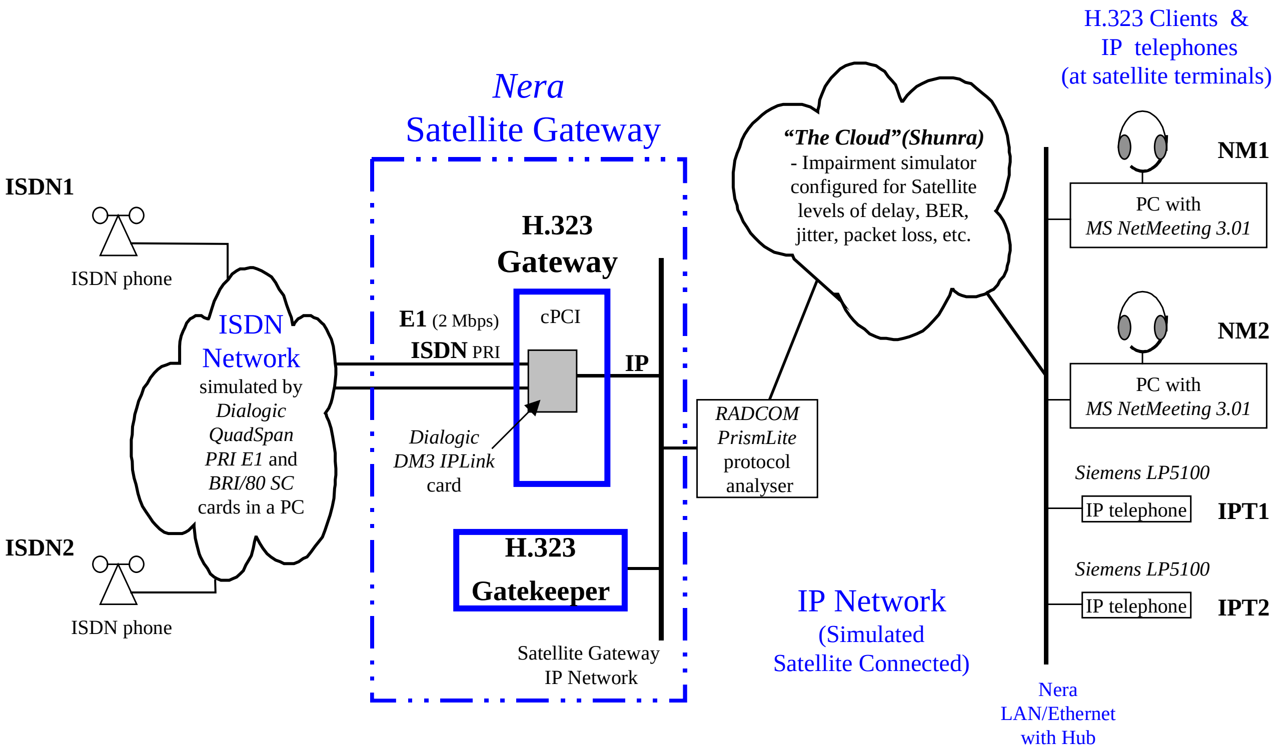

Figure 8: IMMSAT Prototype, including network and telephony configuration details

6. Test Cases: Configurations and Approach

As mentioned in section 2.2, the primary objective of the prototype was to validate that H.323 protocols and an H.323-based VoIP service could successfully be used in a GEO satellite-based IP network. To this end, two primary types of VoIP tests were carried out during prototype testing: *call establishment* and *audio voice quality*. The satellite-based environment parameters in focus were latency, jitter, packet loss and bit error rate (BER). With regard to the statistical model used during jitter simulation, a normal distribution was used rather than a uniform distribution. The two models were judged to be of equal significance; in addition, the normal distribution also offered greater variations from the norm.

All testing was performed using various combinations of H.323 clients, IP telephones, ISDN telephones, H.323 Gateway, and Gatekeeper (for most tests). Voice encoding and decoding was fixed to G.711 in the ISDN/H.320 network, and to G.723.1 (6.3Kb/s), in the IP/H.323 network. During testing, transcoding was always performed in the H.323 Gateway.

During the primary phases of call establishment and audio voice quality testing, a Direct Endpoint Call Signaling via Gateway model was employed (see Figure 5).

Some early testing was also carried out without the use of a Gatekeeper; the call model then corresponded to a simple Direct Endpoint Call Signaling call model (Figure 2). For further clarifications about these models, see section 3.1.3.

6.1 Test Strategy

During prototype testing, the satellite-based environment parameters in focus were latency, jitter, packet loss and bit error rate (BER). Since “the Cloud” could vary each of these parameters independently, a multi-dimensional parameter space was open for exploration.

Obviously, the process of testing call establishment and audio voice quality characteristics for *all* parameter combinations was not feasible. To be able to meet the objectives in a viable way, a strategy was chosen in which each of these various parameters was investigated independently, while the remaining parameters were held constant or nullified. Often, parameters held constant during a trial were set equal to their typical value. For instance, delay and jitter were often held constant with values set at 550 msec and a standard of deviation of ±15 msec, respectively. When jitter is modeled as a normal distribution, these values correspond to a transmission latency closely approximating 520-580 msec (see section 5).

The strategy was such that during a test instance or *trial* (e.g., a call establishment test for a specific client pair), a bounding value for each individual parameter was identified. In this investigation, a bounding value was a value for a *single* parameter (e.g., latency, jitter, etc.) at which the system failed to perform (e.g., call establishment failed). Once bounding values were identified for each parameter during a trial, efforts then concentrated upon investigating the "edges" of successful / unsuccessful performance.

More specific details about the approach and test process for the call establishment and audio voice quality testing is described further below.

6.2 Approach

6.2.1 Call establishment

Call establishment testing was performed using different client type *pairs* (e.g., NM to ISDN, NM to IPT, NM to ISDN, etc.), where the *direction* of the call (i.e., caller vs. callee) was clearly noted as a significant variable. As mentioned in the test strategy, testing first aimed to identify bounding values where call establishment began to fail due to long delay, large packet loss / BER, etc.

As it turned out, early yet significant experimentation with the impact of varying BER upon call establishment indicted that it was necessary to raise this rate to values greater than 1/50000, in order that call establishment should *occasionally* fail. Since such a rate is *far* outside the expected performance range for most networks, including GEO satellite, further investigation of this variable was terminated.

Thus, the majority of call establishment testing concerned investigating the effects of delay and packet loss. Once bounding values for these parameters were identified, two or three calls were made around areas of critical value for each parameter. Unfortunately, time available for testing yielded only small sample sizes for each client type combination.

During all call establishment testing, delay jitter was held constant, since it was assumed not to have a significant here. This assumption was based on the consideration that:

- delay is *much* more significant to call establishment (since long delays liken packet loss from a TCP perspective), and
- the typical value for jitter in a GEO satellite-based environment is at least an order or magnitude less that the typical value for delay.

6.2.2 Audio voice quality

The voice quality tests included two aspects, *listening* quality and *conversational* quality. To check the subjective *listening* quality of audio voice, Recommendation P.800 [16] was consulted. This Recommendation offers a number of alternative methodologies for acquiring subjective judgements about audio voice quality. Which methodology and opinion scale to select and employ depends upon the testing objectives and test environment. For IMMSAT, the scale chosen was one offered for *Quantal-Response Detectability Testing* (see [16], Annex C). This scale is presented below in Table 3.

Table 3: Quantal-Response Detectability Ratings

Detectability rating	short description	long description
0	inaudible	Noise / distortion completely undetectable
1	just audible	Noise / distortion can just be detected by listening carefully
2	slight	Noise / distortion detectable, but not disturbing
3	moderate	Noise / distortion slightly disturbing
4	rather loud	Noise / distortion causes appreciable disturbance
5	loud	Noise / distortion very disturbing, but call would be continued
6	intolerable	Noise so loud (or distortion so great) that the call would be abandoned, or operator would be asked to change the line

It should be noted that the time and resources available for listening quality testing were so limited that only a single subject was employed to perform the listening tests and to rate performance. Since only one subject was to rate listening quality performance, testing was made as "blind" as possible. That is, one person changed the values of the simulated satellite parameters (thereby affecting audio voice quality); the listener had no knowledge as to where in the parameter space the values were set[5]*.*

To be able to perform listening quality tests in as quiet an environment as possible within the lab, two H.323 clients (one NetMeeting client and one Siemens IP telephone) were placed in a "listening booth". The small room did not conform to the requirements of test cabinets such as in ITU-T specifications [16], since it was neither sound-proof nor of adequate volume. With the door closed, however, the room provided a far better environment for listening quality tests than the lab room itself.

[5] The listener was aware of which trial was being carried out (e.g., client type, number of frames and number of hops).

In order to employ a consistent audio reference, playback of a pre-recorded audio source (a voice file) was used. The H.323 Gateway was configured to trigger playback of the file from the QuadSpan platform (see section 5.1). When an H.323 client called the Gateway's IP address, the call was routed to "voice recorded announcement" equipment in the Dialogic QuadSpan platform.

As part of the listening quality tests:

- silence suppression was turned *off* in all prototype elements (whenever and wherever possible)
- audio frame size was 30 msec
- number of audio frames per packet was clearly noted as a significant variable (either 1 or 2 audio frames per packet)
- listening quality for both one and two hop calls was checked.

The tests performed for each trial:

- varied jitter and packet loss, while BER was held constant
- varied jitter and BER, with packet error loss held constant

For all listening quality tests, the base value for delay from which jitter was simulated was set to 550 msec. This decision was based on the consideration that *when listening on only <u>one</u> end of the call, delay does not to affect the quality of the voice listened to*. The effects of delay upon *conversational quality* have been well studied and were therefore not deeply investigated during IMMSAT testing. One summary of the effects of delay upon conversation can be found in [6]:

> *Delay does not affect the speech intelligibility but rather the character of a conversation, up to the point where no conversation is possible at all. Below 100 ms, most users will not realize the delay. Between 100 ms and 300 ms, users will realize a slight hesitation in the partner's response. Interruptions are more frequent and the conversation can get out of "beat" as the delay increases. Beyond 300 ms, the delay is obvious to users and they will frequently have to back off to prevent interruptions and "talk-over". Delay above 300 ms is not suitable for toll telephony, and the challenge is to keep it under 100 ms, for "best" overall quality.*

Given this existing knowledge and understanding, only quick, general checks on conversational quality were performed for each client combination. The checks were performed using expected values for the satellite parameters (see section 5). These checks were summarized using very simple ratings, i.e., "satisfactory", "some small noise/distortion", "noisy/distorted" or "very noisy/distorted".

7. Analysis of Results

This section presents, analyses and discusses the results of call establishment and audio quality testing carried out within the IMMSAT project. It must be said at the outset that since the total number of trials in each test series was so very limited, a judgement was made to analyze the results on the basis of apparent, rather than statistical trends.

Furthermore, *certain hypotheses and remarks below about equipment performance and documentation available during testing are in no way meant to be critical or injurious to their manufacturers and/or vendors*. These remarks are simply meant to help clarify the observed behavior of the prototype and the authors' capacity to understand it within a time-restricted testing and analysis effort.

7.1 Call Establishment

The results for call establishment testing are presented in Figure 9.

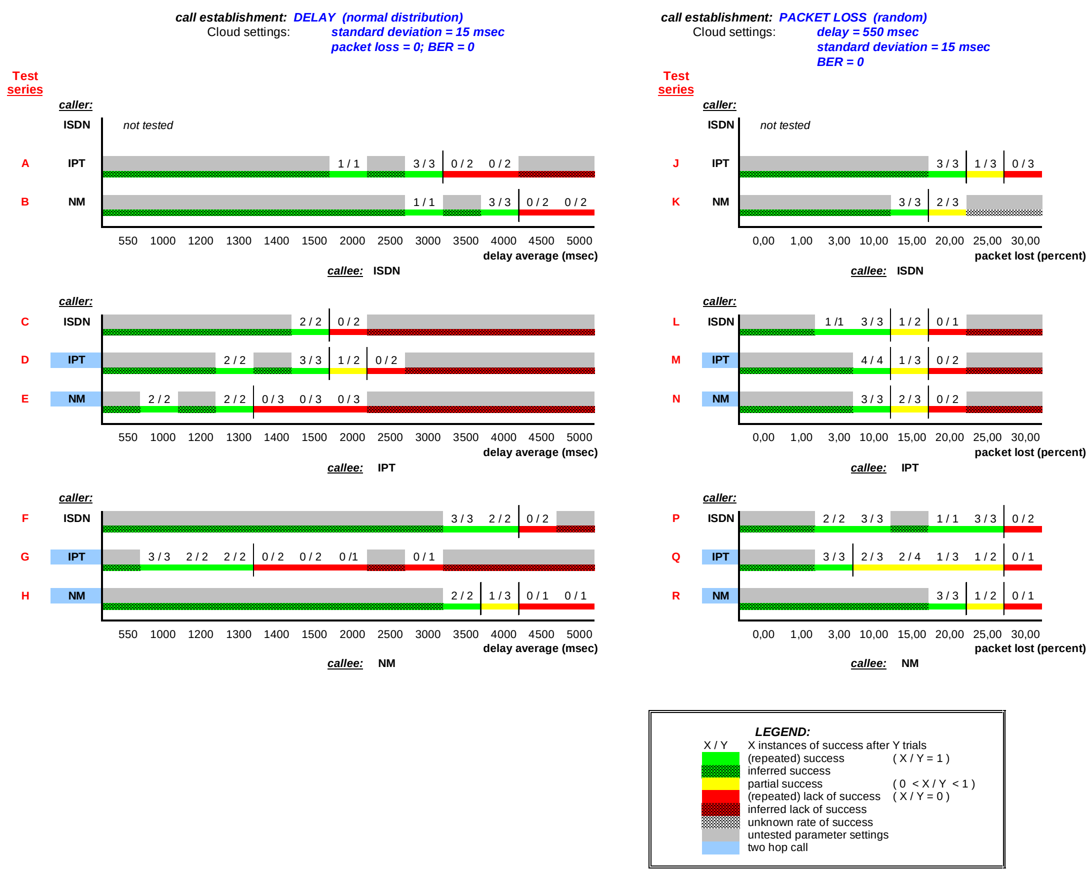

Figure 9: Results of call establishment testing for delay and packet loss

In the figure, two major sets of call establishment tests are summarized. Each major set included eight *test series*. These two major sets include:

- investigations of the effects of varying delay (test series A-H) and
- investigations of the effects of varying packet loss (test series J-R).

The relevant variables for each test series are: the type of terminal initiating the call (the *caller*), the type of terminal answering the call (the *callee*), whether the call involved one or two (simulated) satellite hops and the settings for the satellite simulator. When reviewing Figure 9, it is important to note that the horizontal axes employed therein are *not* linear.

7.1.1 Effects of delay

In regard to the effects of delay upon call establishment, there were two major observations. The first was that in the face of delay, call establishment performance did not seem to be influenced by the addition of a second (simulated) satellite hop. This can be seen in Figure 9 by comparing: test series C to D, test series C to E, and test series F to H. It is expected that this behavior is related to the characteristics of a Direct Endpoint Call Signaling via Gateway model. As explained in section 3.1.4, this kind of model helps serve to decouple end-to-end call setup timing constraints, such that the success of call setup should be little affected by the number of hops within the call.

With respect to this first observation, the comparison of F to G is *not* conformant. It is presumed that this result is a consequence of the second observation described below.

The second major observation is that increases in delay apparently seemed to impact calls employing IPT (as either caller or callee) *more adversely* than calls which did not involve IPT. This observation is depicted graphically in Figure 10a.

Trying to verify and explain this observation requires careful study of the effects of the (simulated) satellite hop upon the timing of signal exchanges within the call model employed (see Figure 5). A higher-level verification through logical deduction is presented here, instead.

IPT as a caller endpoint: Consider test series F, G and H. In each, the *second* leg of the call involves signaling between the Gateway and NetMeeting (i.e., NetMeeting is the final callee endpoint). Series F and H demonstrate that both first and second call legs *can* tolerate up to 4000 ms delay. Witnessing that series G begins to fail at 1400 ms implies that the failure lies on the first leg of the call in that series, i.e., the signaling between IPT and the Gateway. A reasonable though not verified hypothesis in this case is that timeouts occurred within the IPT terminal, the originating caller.

IPT as a callee endpoint: Consider test series B, E and H. In each, the *first* leg of the call involves signaling between NetMeeting and the Gateway (i.e., NetMeeting is the originating caller endpoint). Series B and H demonstrate that both first and second call legs *can* tolerate up to 4000 ms delay. In contrast, series E begins to fail at 1400 ms. This implies that the failure lies on the second leg of the call in that series, i.e., the signaling between the Gateway and IPT. It is interesting to see here that the other two series having IPT as final callee endpoint (series C and D) also demonstrated failure above 1500 ms[6].

A reasonable though not verifiable hypothesis in this case is that IPT did not issue a

[6] Failure in these series was first detected at 2000 ms; no other intermediate delay settings were used between 1500 ms and 2000 ms.

Call Proceeding signal to the Gateway, a situation which caused the Gateway to timeout and clear the call. As mentioned in section 3.1.4.2, callee endpoints are not required to issue the *Call Proceeding* signal; assuming this hypothesis is true, however, these test series seem to demonstrate the value of it being issued by the callee endpoint.

In concluding this section about the effects of delay, it must be mentioned here that the reason the hypotheses are not verifiable are several:

- the packet tracing equipment used during testing was configured with a basic protocol decode option and a preliminary software release, such that only the first 40 Ethernet frames of call set-up signaling for a call could be captured;
- signal processing time within the terminals could not be exactly determined;
- *during testing*, the documentation available from certain vendors for their hardware configuration files was quite poor; for example, the values for setup timers were integers, but the kind of *units* to be represented (e.g., seconds, tenths-of-seconds) were not specified; and,
- the Gateway apparently used only *two* timers to represent the T303, T310 and T301 timers called for in the H.225 and Q.931 standards[7] (see section 3.1.4.1); the relation of those two timers to the ITU standards was not explained in the system documentation available.

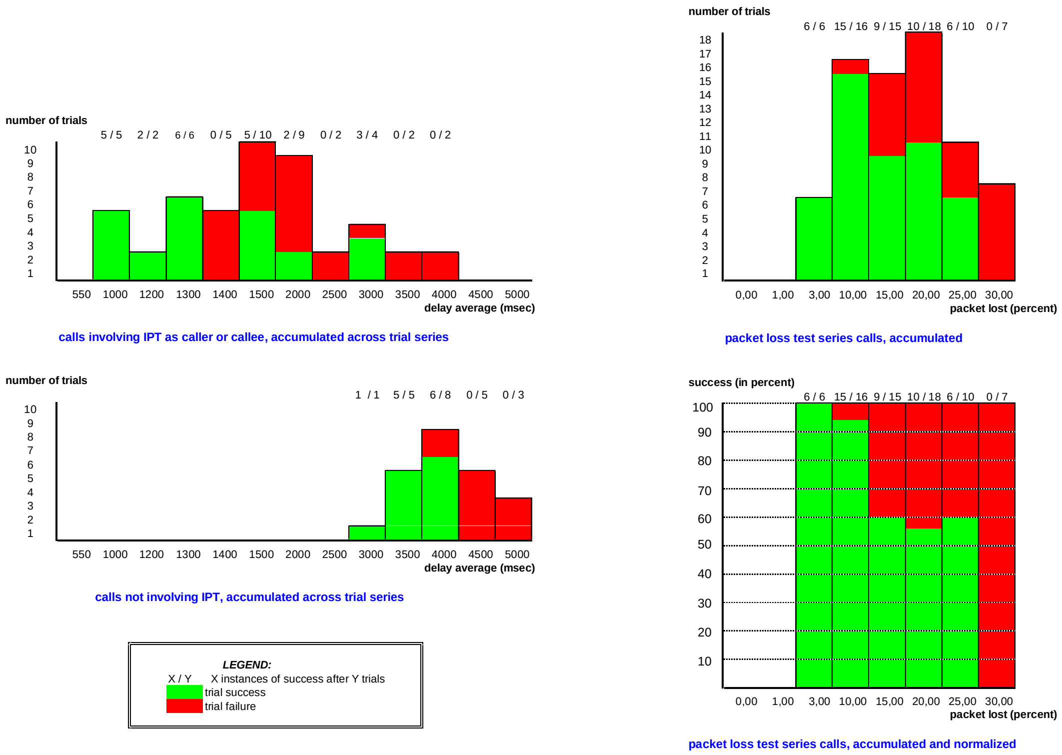

Figure 10: Satellite effects upon call establishment:
(a) Comparative effect of delay for ITP vs. non-IPT calls *(left column)*
(b) Effect of packet loss *(right column)*

[7] It was learned that this is not uncommon for certain equipment implementations [22].

7.1.2 Effects of packet loss

Analysis of the call establishment testing for packet loss did not reveal any apparent trends with respect to the variables under study (e.g., caller vs. callee type, hop count, etc.). The only apparent trend is that the success rate of call establishment seemed to significantly diminish when packet loss reached 15-20%. This behavior is illustrated in Figure 10b, where the results of all packet loss trials are accumulated. It should be mentioned here that some of the *successful* call establishment trials employing packet loss rates of 20-25% took one minute or more to establish.

7.2 Audio Voice Quality

The voice quality tests included two aspects, *listening* quality and *conversational* quality.

7.2.1 Listening Quality

The results for listening tests are presented in Figure 11.

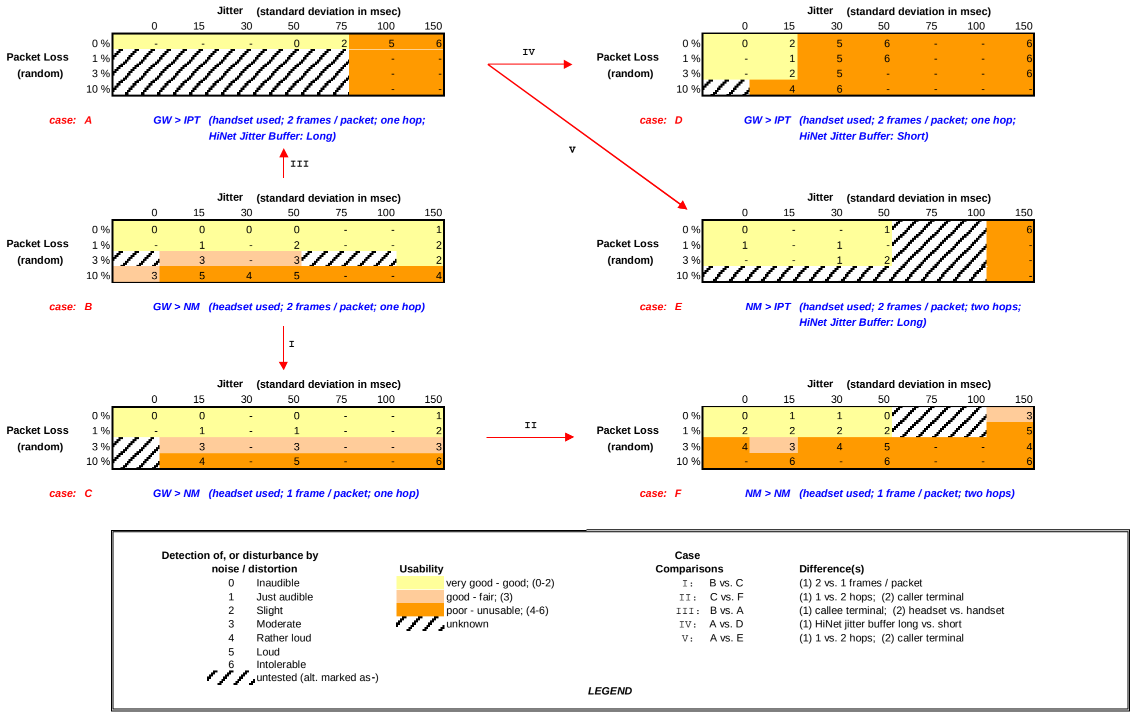

Figure 11: Results of listening tests for jitter and packet loss

In the figure, six major *cases* of listening tests are summarized: one matrix for each major case (case matrices A-F). The approach used for testing, including the testing scale, is described in section 6.2.2. In the figure, each case matrix is also colored according to a "usability scale" defined for IMMSAT. The colors essentially maps the seven point quantal-response detectability scale (see section 6.2.2) into a three point scale; this was done in order to ease perception of the significance of the listening test results. The legend in Figure 11 explains this mapping. Note that in certain case matrices, parts of the matrix are colored despite the fact that no trial was performed.

These instances of coloring were *inferred*, based upon the performance of other trials within the same matrix.

Each major case in the figure reports the effects of varying jitter and packet loss; testing of the effects of BER upon audio quality was omitted for the same reason as that of call establishment (see sections 7.1.1 and 6.2.1).

The parameters which *distinguish* each major case are: the type of terminal initiating the call (the *caller*), the type of terminal answering the call (the *callee*), whether the call involved one or two (simulated) satellite hops, whether the listening test was performed using a headset or a handset, and the number of audio frames per packet. For the cases in which IPT was the callee, an additional parameter was the setting for the HiNet jitter buffer; this buffer could be set to either 'Long' or 'Short'. The HiNet jitter buffer is a buffer internal to the Siemens IP telephones. The parameters which distinguish the trials *within* each major case are the settings for the satellite simulator.

When reviewing Figure 11, it is important to note that the axes employed for each major case matrix are *not* linear.

The easiest way to review the outcome of the listening tests and to describe its analysis is to view these results via comparisons amongst major cases A-F. These are summarized as "case comparisons" I-V in Figure 11's legend, and are depicted as red arcs in the figure. The legend succinctly describes the variation in parameter values for each case comparison. Note further that the case comparisons can be viewed as tree rooted at case B.

Case B depicts a listening test where the audio reference file was played back from the H.323 Gateway; NetMeeting was used as the callee. The other parameter settings for the case were: headset; 2 frames/packet; and one hop. As the figure shows, this configuration supported "good – very good" audio quality for jitter values as high as 150 msec and packet loss up to 1%. Audio quality began to noticeably degrade to "fair" when packet loss reached 3%. With packet loss at 10%, audio quality became "unusable" as soon as jitter appeared.

Comparing this performance to case C (comparison I), there is little difference to be seen. Thus, it appears that using two vs. one audio frames per packet is not of any significance.

Differences are noticeable, however, when comparing case C to case F (comparison II). Here, case F involves two hops, while case C involves only one. Comparison II illustrates that audio quality degrades from "good/fair" to "poor" when calls include two hops with ca. 3% packet loss per hop; this kind of result is expected, of course, due to the cumulative effects of packet loss, delay and jitter across multiple hops (see section 3.2.2). Worth noting in case F is the near-usability of the trial having 150 msec jitter and 0% packet loss per hop. This result seems to indicate that NetMeeting is internally operating with a relatively "large" jitter buffer. No user-level access to this parameter was available via the NetMeeting application, however, in order to confirm this hypothesis.

Consider now comparison III (case B vs. case A). The most significant difference from case B is that instead of NetMeeting, the Siemens LP5100 telephone is used as the callee terminal. The most apparent difference in their performance is that for the LP5100 telephone, jitter has a severely destructive impact upon audio quality as soon as it reaches 100 msec. This was the case even though the HiNet jitter buffer was set to ‘Long’. When that buffer was set to ‘Short’ (see case D, comparison IV), the effect of jitter was devastating.

In contrast to the LP5100 telephone’s poor performance in the face of jitter, the results seem to indicate that its audio quality was somewhat better than NetMeeting, given the same levels of packet loss. Though this is not explicitly shown in comparison III, one may choose to logically infer this by considering how well the LP5100 performed with respect to packet loss in case E — a two hop call (comparison V).

In summary, the one-way listening tests showed that uncontrolled jitter was the most *destructive* parameter to audio quality. Packet loss was also influential, though less dramatically; that is, a rate between 3-10% packet loss was required in order to render audio quality “poor – unusable”.

7.2.2 Conversational quality

Even when the intelligibility of audio is unaffected by its transmission medium, the exclusive effect of delay can have severe consequences upon the character of a conversation (see section 6.2.2 and [6]). Since this area has already been so well investigated, the tests for conversational quality were only quick, general checks. These checks were registered when the parameters for the satellite simulator were set to the typical values expected for latency, BER, packet loss, etc., as presented in section 5.

During IMMSAT testing, the calls to the Siemens LP5100 telephones through the H.323 Gateway were affected by what was suspected to be an incompatibility in silence suppression coding / encoding between audio codec implementations at the Dialogic GW and Siemens IP telephones. During periods of silence at the caller, a high amplitude “fax-like” noise was received at the callee (the LP5100s), which made any conversation impossible. Calls originating from the LP5100 to the other H.320/H.323 clients did not suffer from this problem.

In general, the conversational quality between clients was satisfactory; the two hop calls tended to included some (very) small amount of noise / distortion.

For the sake of experience, conversational quality was occasionally tested when delays were varied between 1500-4000 msec and other satellite parameters were held at their typical values. As expected, users could ultimately manage to successfully communicate. With such large delays, however, it was necessary for them to adopt an unnatural "single duplex" mode of conversation.

8. Conclusions

Generally, all tests performed using the IMMSAT prototype indicated that H.323-based VoIP services in a GEO satellite-based IP network are practical and usable. The effects of *typical*, GEO satellite environment conditions upon H.323-based VoIP services are expected to be negligible.

These general statements are further elaborated upon below, through treatment of call establishment and audio voice quality, respectively.

8.1 Call Establishment

The call establishment tests showed that the H.323 protocols operated successfully under severely impaired conditions — conditions which were at least an order of magnitude worse than in typical GEO satellite-based environments[8]. Establishing calls *could* be accomplished for most client combinations, even with 3-4 seconds delay and 10-20% packet loss. However, some of these more extreme trials took one minute or more to establish.

The call establishment tests also indicated that the client type on each end of the call can have great significance with respect to the amount of delay which can be tolerated during call establishment. Clients are usually pre-configured by the manufacturer and — when to be used in connection with satellite environments — may well require certain adjustment of internal timers in order to deliver best call establishment performance.

With respect to timers and signaling, there is also an issue as to what degree the implementation of the equipment accords with international standards. During the IMMSAT testing, it appeared that one of the clients failed to issue the *Call Proceeding* signal. The significance of this condition was described in section 7.1.1. Thus, persons selecting equipment for eventual deployment and use should be careful to ensure that the equipment chosen has first undergone stringent interoperability testing.

Lastly, it was interesting to see that both one and two hop call configurations performed equally well in the face of delay. It is expected that this behavior is due to the characteristics of a Direct Endpoint Call Signaling via Gateway call model, as explained in section 3.1.4.

8.2 Audio Voice Quality

The experiments showed that audio quality appreciably degrades when moving from a one hop call to a two hop call. This kind of result is expected, of course, due to the cumulative effects of packet loss, delay and jitter across multiple hops (see section 3.2.2). The experiments showed that client type also has an impact upon voice quality; use of headset instead of a handset also makes a difference here.

Still, jitter is the network characteristic that seemed to have the greatest impact on voice quality. Organizations planning to deploy VoIP equipment and services for a GEO satellite-based IP network should investigate solutions which can avoid or compensate for the destructive effect of *uncontrolled jitter* which may occur in these environments.

Even though the adverse conditions tested in the IMMSAT prototype are not likely to be observed in commercial GEO satellite-based environments, the ability to handle jitter is crucial for offering an acceptable VoIP service. Using a larger jitter buffer in certain system elements may help alleviate the problem, but an increase in jitter buffer size also implies an increase in overall delay.

The experience of *large* delays in a satellite-based voice service can be frustrating and confusing, even to users who have been informed of and prepared in advance for such conditions. Users who are familiar with large delays *can* adapt and reconcile themselves to the condition and use the service, even though conversations may be somewhat unnatural.

Users who are unprepared for delay conditions may also find a way to adjust to a satellite-based voice service, but will most likely find the voice service to be “different”

[8] See section 5 for typical delay and error conditions within GEO satellite environments.

and possibly inferior to voice services that they may have used previously. Whether a user will accept and use the satellite-based VoIP service will probably depend on whether any alternative voice services are available to the user and on the relative quality of the services. Probably the main factor that will determine “to what degree” satellite-based VoIP services are used is the price of the satellite-based VoIP services compared to the price of alternative voice services.

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