Overall picture of IP telephony

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1 Abstract
The main trend during the last five years in telecommunications, the convergence, has lead to the development of multimedia services in packet switched networks e.g. the Internet. After a short experimental phase the H.323 standard has been laid as a basis for multimedia services and applications including IP telephony. The present installed IP telephony systems use the H.323 protocol stack. Due to the complexity albeit flexibility of the H.323 the IETF is now finishing new rivalling standards the SIP and MGCP, which are aclaimed to offer better functionality and simpler implementation. Interoperability problems have sofar hindered a breakthrough of IP telephony. The driwing force in the telephony netwrok convergence, cheaper calls, has not compensated for the technical defiences.

2 Introduction
The term IP telephony is older than Voice over IP. IP telephony has earlier meant the use of telephones or hybrid equipment and PBXs over IP using gateways to overcome the barriers of different networks. Voice over IP points to a world of carrying voice over IP networks not necessarily needing any separate telephone like equipment nor PBXs. Software phones in PCs are an example of these new implementations. Today the two terms are more or less synonyms or IP telephony is a subset of VoIP.

2.1 A short history
Only ten years ago the Internet was something totally different it is today. Its use was restricted mainly to universities and research institutes. Its interface was text based and FTP was the main tool for exchanging information alongside with email and chat.

The first revolution occurred in 1993 with the World Wide Web. The colourfull new user interface appealed to thousands and thousands of new users and emerging search engines helped the users to find interesting new sites.

Year 1996 the first attemps were made to build an Internet telephony gateway. It consisted of a modem with speakerphone capabilities. The modem could only dial the destination number. At that time some sound board drivers were capable of simultanious play and record (full-duplex), but they lacked a telephone interface. The soundboard line-in jack had to be wired to the modem microphone and the modem speaker to the sound board line-out jack.

Some software was needed and the telephony freeware of those days the VAT came to help. By adding some code to interface the modem a crude one-line gateway prototype was invented. The potential of this primitive invention was huge. The development in the voice realm of Internet has since been immensely rapid and it has made a real contribution to the much advertised convergence of telecommunication.

However, the new possibilities also created new problems: Internet at that time was not ready for real time applications.

Anyhow IP telephony is growing very fast and it is estimated that by year 2002 nearly 20 % of the U.S. phone traffic will be carried over data networks.

By the World Wide Web the Internet had got its face, now it was getting a voice.

2.2 The overall situation now
The beginning of IP telephony has been lucky in that widely accepted standards have emerged in an early stage. Allmost all present implementations support the H.323 protocol family.

Standards should make it easy for the equipment of various vendors to interoperre. Unfortunately this has not been the case sofar in IP telephony. On the contrary the equipment and service implementations have mostly been proprietary in that the vendors have chosen a subset of the large and complex H.323 protocol stack that has met their immediate requirements. If you have bought an IP telephony system from one vendor you have been stuck to bying all future equipment from the same vendor. This lack of interoperability has been the major impediment for the wider deployment of H.323. For this reason fastest growth in VoIP will probably occur in
enterprise networks, where a uniform system and equipment base is easier to achieve. [3]

The capabilities negotiation phase could at least to some extent solve this problem, but unfortunately even it is often not implemented completely.

This interoperability drawback is now luckily fading. The IP telephony manufacturers are more and more acclaiming that their hardware will interoperate with other vendors systems. The International multimedia teleconferencing consortium IMTC has been set up with the primary goal of ensuring that various vendors products and services will interoperate.

Today the standardization situation is however not at all clear. To overcome the drawbacks of the cumbersome and difficult to implement yet flexible H.323 protocol family the IETF has created new protocols like the Session initiation protocol SIP and the Media gateway control protocol MGCP offer much more functionality than H.323 to VoIP.

SIP is simpler, it scales better and it leverages the existing DNS system instead of having created its own separate hierarchy of name services. By including a clients communication features within the invite request, SIP negotiates these features and capabilities of the call within a single transaction. The call setup delay can be as low as 100 ms depending on the network.

Thus the biggest question in VoIP today is which one of the standards will prevail. H.323 is now widely accepted and deployed, but many vendors have also announced support to the newcomer protocols. At this transitional stage we will probably see systems which support both protocol families.

This paper restricts to presenting an overview of the present prevailing technology, which anyway has laid the foundation of IP telephony and leaves the deeper presentation and comparison of the new standards to other presentations.

2.3 Characteristics of IP telephony

The characteristics of IP telephony are quite complex, especially compared to streaming video, where large buffers can be used to compensate for the imperfectness of the Internet regarding real time applications.

The main issues of IP telephony to be dealt with include:

- The human ears perception of echo and delay
- The voice compression and packetization technics
- Silent suppression and comfort noise generation
- The Internet shortcomings for packetized voice: delay, jitter and packet loss
- The according remedies: buffering, redundancy, time stamps and differentiated services
- Telephone signalling protocols and various call types

3 H.323

H.323 is an ITU-T standard that was first developed for multimedia (voice, video and data) conferencing over LANs and later extended to cover Voice over IP. This multimedia origin is partly the reason for its claimed complexity for mere VoIP. Its first version H.323v1 was accomplished in 1996 and the second version v2 was ready by 1998. It includes both point-to-point and multipoint connections.

H.323 is one of ITU-T’s mutually compliant videoconferencing standards. The others are:

- H.310 for broadband ISDN (B-ISDN)
- H.320 for narrowband ISDN
- H.321 for ATM
- H.322 for LANs with guaranteed QoS
- H.324 for public switched telephone networks (PSTN)

Clients of H.323 are able to communicate with clients of the other above mentioned networks.

The H.323 standard does not assume any QoS in the network.

3.1 Components of H.323

3.1.1 Terminal

Terminals are the LAN client endpoints providing real time two way communications. They have to support H.245, Q.931, Registration Admission Status RAS and Real Time Transport RTP protocols.

A H.323 terminal can communicate with an other H.323 terminal, a H.323 gateway or a MCU.

3.1.2 Gateway

A H.323 gateway endpoint is the interface between the Internet and the PSTN or some other network. It communicates in real time mode between H.323 terminals on the IP network and other ITU terminals on a switched network, or to an other H.323 gateway. The H.323 gateway is optional and thus is not needed in a homogenous network.

Gateways perform the translation between differing transmission formats like from H.225 to H.221. They can also translate between audio and video codecs. In
3 Overall picture of IP telephony

one single LAN the gateway is not needed, as the terminals in this case can communicate directly. The communication to other networks is done via gateways using the H.245 and Q.931 protocols.

3.1.3 Gatekeeper
The gatekeeper is the vital - yet optional - central managing point in its zone. When a gatekeeper is used all endpoints in its zone (terminals, gateways and MCUs) have to registered with it. It supports the endpoints of its zone by

- Address translation from an alias, such as an email address or a telephone number, to a transport address using a translation table, which it updates by registration messages
- Admission control denying or accepting access based on e.g. call authorization or source and destination addresses.
- Call signalling either by processing the signalling itself or with the endpoints. It may alternatively connect a call signalling channel between the endpoints and let them do the signalling directly.
- Call authorization using the H.225 signalling. The gatekeeper can reject calls due to time period or particular terminal access restrictions
- Bandwidth management, complying the number of calls with the bandwidth available
- Call management maintaining optionally a list of ongoing H.323 calls for e.g. Bandwidth management purposes
- Routing all calls originating or terminating in its zone. This feature enables billing and security. Rerouting to an other gateway in case of bandwidth shortage is also included in this option and it helps in developing mobile addressing, call forwarding and voice mail diversion services.

3.1.4 Multipoint Control Unit
The Multipoint Control Unit network endpoint makes it possible for three or more terminals and gateways to participate in a multipoint conference. The MCU consists of a mandatory Multipoint Controller MC and optional Multipoint Processors MP.

The MCU is an independent logical unit, but it can be combined into a terminal, a gateway or a gatekeeper.

The MC determines the common capabilities of the terminals by using the H.245 protocol, while the MP does the multiplexing of audio, video and data streams under the control of the MC.

In addition the MCU can determine whether to unicast or multicast the audio and video streams depending on the capability of the network and the topology of the multipoint conference.

In a centralized multimedia conference each terminal establishes a point-to-point connection with the MCU which then sends the mixed media streams to each endpoint. In the decentralized model a MC manages the communication compatibility but the terminals multicast and mix the streams.

3.2 The H.323 protocol stack
The audio video and registration packets of H.323 use the unreliable UDP protocol, while the data and control packets are transported by the reliable TCP protocol.

3.2.1 H.225 Call signalling
The call signalling channel is used to carry the H.225 control messages. In networks where a gatekeeper does not exist, the calls are signalled directly between endpoints using Call signalling transport addresses. In this it is assumed that the calling party knows the address of the called party.

If there is a gatekeeper in the network, the calling party and the gatekeeper change the initial admission message using the gatekeeper’s RAS channel transport address.

Call signalling messages can be passed in two ways

- In Gatekeeper routed call signalling the signalling messages are routed between the endpoints via the gatekeeper
- In Direct endpoint call signalling the endpoints change the messages directly

After the call signalling is completed the H.245 Control channel is established. When Gatekeeper routed call signalling is used, there are two ways to route the H.245 Control channel. Either the control channel is established directly between the endpoints or via the gatekeeper.

![Figure 1: The H.323 protocol stack](image)
3.2.2 H.245 Media and Conference control
After a H.323 call is established, H.245 negotiates and establishes all the media channels carried by RTP/RTCP.

The functions of H.245 are

- Determining master and slave. H.245 appoints a MC, which is in charge of central control in case a call is extended to a conference
- H.245 negotiates compatible settings between the endpoints after the call establishment. Renegotiation can take place anytime during the call
- Media channel control by which separate logical channels for audio, video and data can be opened or closed after the endpoints have agreed on capabilities. Audio and video channels are uni-directional while data channels are bi-directional
- Flow control messages provide feedback in case of communication problems
- Conference control keeps the endpoints mutually aware in a conference situation. A media flow model between the endpoints is also established

3.2.3 H.225 RAS Registration Admission Status
RAS defines communications between the endpoints and the gate keeper (in case one exists) by unreliable transport i.e. UDP.

RAS communications include

- Gatekeeper discovery is used by the endpoints to find their gatekeeper: endpoints multicast gatekeeper requests to find the gatekeeper transport address
- Endpoint registration is compulsory in case where a gatekeeper exists in the network. The gatekeeper must know all the aliases and transport addresses of all the endpoints in its zone
- Endpoint location. A gatekeeper locates an endpoint with a specific transport address to update its address database for example

3.2.4 H.248 Implementors' Guide
One reason for the poor interoperability between various implementations of H.323 has been attributed to the lack of an implementation guide. This problem is now being solved by the IETF Megaco project.

There is now a workgroup that is standardizing the H.248 Implementors' guide. At present it is specifying draft version 5 of the Implementors' guide. That version of the document is now at a first draft stage of a resolution of comments.

3.2.5 RTP
The Real time transport protocol RTP like RTCP are both developed by the IETF. They transport the audio, video and data packets of real time media over packet switched networks. They are annexed in the H.323 protocol.

The main tasks of RTP are packet sequencing for detecting packet losses, adjusting to changing bandwidth conditions by payload identification, frame identification, source identification and intramedia synchronization to compensate for the varying delay jitter of the stream packets.

3.2.6 RTCP
The Real time transport control protocol works in conjunction with the RTP. In a RTP session participants send periodically RTCP packets to obtain information about QoS, session quitting, participant identification (email addresses, telephone numbers etc.) and intermedia synchronization.

3.2.7 Q.931
Then main purpose of Q.931 is call signalling and setting up the call.

3.3 IP to IP call

<table>
<thead>
<tr>
<th>Message</th>
<th>Terminal A</th>
<th>Terminal B</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>Setup</td>
<td></td>
</tr>
<tr>
<td>2</td>
<td>Alerting</td>
<td></td>
</tr>
</tbody>
</table>
Table 1: Message exchange between terminals A and B during a call

<p>| | |</p>
<table>
<thead>
<tr>
<th></th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>3</td>
<td>Connect</td>
</tr>
<tr>
<td>4</td>
<td>termCapSet</td>
</tr>
<tr>
<td>5</td>
<td>termCapAck</td>
</tr>
<tr>
<td>4</td>
<td>termCapSet</td>
</tr>
<tr>
<td>5</td>
<td>termCapAck</td>
</tr>
<tr>
<td>6</td>
<td>masterSlvDet</td>
</tr>
<tr>
<td>7</td>
<td>masterSlvDetAck</td>
</tr>
<tr>
<td>8</td>
<td>masterSlvDetConfirm</td>
</tr>
<tr>
<td>9</td>
<td>openReq</td>
</tr>
<tr>
<td>10</td>
<td>openAck</td>
</tr>
<tr>
<td>9</td>
<td>openReq</td>
</tr>
<tr>
<td>10</td>
<td>openAck</td>
</tr>
<tr>
<td>11</td>
<td>endSession</td>
</tr>
<tr>
<td>11</td>
<td>endSession</td>
</tr>
<tr>
<td>12</td>
<td>Release</td>
</tr>
</tbody>
</table>

Voice messages can be sent in version v1 only after media channels have been established by sending first a connect message.

There is a ITU-T Mobility Ad Hoc Group working on mobile H.323 standardization.

### 4.1 Faster procedures

The Fast connect procedure was invented to overcome the above deficiencies. Fast connect solves the problems by

- Enabling uni- or bidirectional messages immediately after the Q.931 setup message
- Allowing a basic bidirectional audio only communication immediately after the connect message has been received
- Improving setup delays

An endpoint that uses the Fast connect procedure informs the calling party of all the media points it is prepared to receive or offers to send. This information is carried in the new fastStart parameter of the user to user Setup message. The description includes the codecs used and the receiving ports etc. This allows the early receiving of network prompts and improves also the setup delay.

The Fast connect procedure has been added as core feature in the ETSI TIPHON project, because it resolves the interworking problem with SCN.

Fast connect makes it possible to build simple limited capacity terminals that need only a minor part of the H.245 protocol.

H.323v2 offers an other solution with H.245 tunneling, where H.245 messages are encapsulated in Q.931 messages reducing the TCP connections to one. When H.245 tunneling is used, the Q.931 channel must remain open for the duration of the call. The Tunneling method can also clear the network generated messages problem and will thus probably replace the Fast connect procedure.

The above described procedures are rather fixes to H.323v1 problems than a simplification of the protocol.

The use of TCP causes at least one unnecessary SYN/ACK round trip. If the Setup message exceeds the maximum transfer unit MTU size, two or more TCP segments must be used. Most TCP implementations are network friends mandating a slow start, where the first TCP segment has to be acknowledged before the rest can be sent.

A remedy to this problem is a special H.323v3 mode that will use UDP instead of or simultaneously with TCP signalling.

3.4 IP to PSTN call

4 Enhancements to H.323

A major drawback - especially compared to the fast SIP protocol - in the first H.323 version was the long call setup time. One message round trip is needed for

- ARQ/ACF sequence
- Setup connect sequence
- H.245 capabilities exchange
- H.245 master slave procedure
- Setup of each logical channel

In addition a TCP connection has to be setup for Q.931 and H.245 channels and each TCP connection also needs an extra round trip for the TCP window synchronization. In a WAN environment one round trip can take 100 ms, which ends up in an unacceptable long setup delay especially when the gatekeeper routed model is used.

In a congested switched circuit network SCN, where a call cannot be setup, the network local exchange tries to send the caller a ‘your call can not be connected’-message. No connect is sent because the network informs the caller and not the endpoint.
4.2 Conferencing with H.323

A multipoint control unit MCU masters a multipoint conference. It consists of one multipoint controller MC and optionally one or more multipoint processors MPs.

4.2.1 Multipoint controller MC

The MC decides

- who is allowed to participate
- how new participants are introduced to an ongoing conference
- how the participants synchronize their operation
- who is allowed to broadcast media etc.

A gatekeeper or a terminal possessing sufficient resources can include MC functionality in it and even mix media locally to a limited extent.

4.2.2 Multipoint processor MP

When several participants of a multipoint conference are simultaneously sending audio, video or data, there has to exist a network element that can mix or switch the incoming media streams. The endpoint terminals seldom have the capacity to do this. This mixer/switch element is called the MP.

When video is sent, the MP might choose the pictures of the latest speaker. When audio is the content, the MP could sum the voices of the potentially simultaneous speakers.

In a centralized conference a MP mixes and switches the media streams, where as in a decentralized conference the terminals send their streams directly to all other participating terminals.

4.2.3 H.332

The conference type where all participants retain a full H.245 control connection with the MCU is called ‘tightly coupled’. This type is resource intensive and it is obvious that it will not scale to numerous participants.

The solution to large conferences is the H.332. A large conference mostly has a panel of active speakers (5 to 10) and a large more or less passive audience of which one speaker at a time can propose a question or a comment to the panelists.

The H.332 keeps ‘tightly coupled’ conference connections with the panelists and a multicast RTP/RTCP conference with the passive listeners. The listeners have to know especially the codec and the UDP port used. H.332 uses the IETF Session description protocol SDP to encode this information.

Due to the large number of participants in a panel conference, a constraint must be set: the codec should remain stable. No new participant should have the possibility to change the codec as this would mean new negotiations for all the others.

If a listener wants to speak, he must use the regular join procedure to attain the right to speak his mind.

4.3 Directories and numbering

Most home IP telephony users are connected to Internet by a dial-up link, where the IP address is allocated on demand and is thus not static. In the early stages the users of IP telephony software contacted a server with a preconfigured IP address.

H.323 makes this kind of solutions obsolete. A terminal has to register to a gatekeeper using a RAS message, which contains all the necessary information, especially the current IP address to contact the terminal by using an alias.

At present the Internet Domain name system DNS is used to resolve the IP address when an alias name is known. The DNS servers make up an addressing network, where an address can be resolved by quering proper DNS servers top down until one is found which has detailed information of the endpoint in question. In addition to alias/IP address pairs a DNS database has much more information. It can hold information of the gatekeepers of its domain in ras://-type txt records.

Once the gatekeeper is found, the caller knows to which transport address he shall send the setup message.

An important issue today for international IP calls from a PSTN network is the lack of a global IP telephony prefix. The solution has to scale to allow a large amount of users. The global prefix should tell the IP-callers network that the call that has to be setup is an IP call and should thus be routed to a home gatekeeper, which knows the location of the called party and can then resolve the phone address to a call signalling address.

It is clear that an IP call should be routed via an IP network avoiding the use of PSTN.

Several proposals have been made to define an IP telephony country code. The standardization process is not yet completed.

For example the use of DNS works well when IP address classes are used, but in the case of the ever more popular classless interdomain routing CIDR, the reverse address resolution is supported only by few servers and is thus not applicable.

4.4 H.323 security H.325

The aim of H.235 is to provide privacy and authentication to all protocols using H.245 including H.323. Even without H.235 H.323 calls are more difficult to listen than ordinary telephone lines, which can be
wiretapped. To break into a H.323 call you have to implement the codec algorithm.

With H.235 IP telephony becomes much safer than PSTN. The caller can even hide the telephone number of the endpoint it is trying to reach. However, the H.235 is not yet widely deployed.

The first purpose of security was to secure the media channels so that no outsider could listen to the ongoing call. Soon it turned out that users most of all did not want to be charged for calls they did not make and that no one could monitor the called phone numbers.

Providers wanted to authorize calls when they were set up, not when media or control channels were established. So the signalling channel had also to be authenticated and secured.

The network elements that have to know the contents of the H.225 and H.245 messages need naturally to be trusted by the endpoints. This authentication can be carried out by Transport layer security TLS or a challenge response exchange using some certificate.

H.323 does not specify the contents of the certificates, but provides a way to exchange them and verify the identities of the callers. The identity can be verified by several methods. A time stamp prevents replay attacks.

H.323 does not ensure privacy on the RAS link between an endpoint and a gatekeeper, but it does provide authentication.

The call signalling channel H.225 can be secured by TLS or IPSEC.

The control channel H.245 security method is negotiated in the call signalling channel during the initial set up process before any other H.245 messages are sent. Various methods are accepted to initiate the secure channel.

After the H.245 channel is ensured, the terminals negotiate the media channel encryption method by capability exchange. A new capability is defined for each codec and encryption mode pair.

Many encryption algorithms can be utilized e.g. DES, Diffie-Hellm and RSA.

5 Codecs

<table>
<thead>
<tr>
<th>Audio codecs</th>
<th>Title and date</th>
</tr>
</thead>
<tbody>
<tr>
<td>G.711</td>
<td>Pulse code modulation of voice frequencies at 56 or 64 kbps (11/88)</td>
</tr>
<tr>
<td>G.722</td>
<td>7 kHz audio coding at 64 kbps (11/88)</td>
</tr>
<tr>
<td>G.723.1</td>
<td>Dual rate multimedia speech coders at 5.3 and 6.3 kbps (03/96)</td>
</tr>
<tr>
<td>G.726</td>
<td>Speech coding at 16, 24, 32 or 40 kbps using ADPCM to encode a G.711 bit stream</td>
</tr>
<tr>
<td>G.728</td>
<td>Speech coding at 16 kbps using low-delay code exited linear prediction (09/92)</td>
</tr>
<tr>
<td>G.729</td>
<td>Speech coding at 8 kbps using conjugate-structure-algebraic-code-exited linear prediction (03/96)</td>
</tr>
<tr>
<td>H.261</td>
<td>Audiovisual video codecs at p * 64 kbps, where p = 1 – 30 (03/96)</td>
</tr>
<tr>
<td>H.263</td>
<td>Low bit rate video coding (02/96)</td>
</tr>
</tbody>
</table>

Table 2: Audio and video codecs used with H.323

The implementation of codecs is well developed and does not create any interoperability problems.

The mandatory speech codec is the G.711, which is a popular codec in telephony networks. It is not however quite suitable for Internet communication, where the subscriber loop bandwidths are much smaller. Today most H.323 terminals use use G.723.1, which is much more efficient using only approximately one tenth of the G.711 bandwidth. The G.728 and G.729 codecs are used for high quality audio with also very low bandwidth requirements.

Due to the burstiness and bandwidth hungriness of video communication efficient compression and decompression technics are of utmost importance. H.323 specifies two video codecs namely the H.261 and the H.263. Other codecs can also be used in case both endpoint support them.

Both the above mentioned video codecs use the discrete cosine transform DCT, H.261 with quantization and motion compression and H.263 with motion estimation and prediction

6 Applications and services

The vision of H.323 is interoperability between packet and circuit switched networks. H.323 also promises new value added services to the customers using circuit switched networks. These goals have not yet been achieved. Lower operational costs alone are not a reason good enough to switch to a new technology.

Several Internet telephony service providers ITSPs have met the expectations of good services in North America and Europe, but the global interoperability is still a big
problem. Furthermore the features and quality of service are often inferior to plain old telephone services POTS.

The main reasons for not meeting the quality goals are the poor interoperability of the endpoints, especially gateways, of various vendors and the limited scalability of H.323 communications. [3]

6.1 The architecture of H.323

The architecture of a protocol lays the foundation for the services and applications that can be built on it. The architectural model of H.323 differs essentially from that of the switched PSTN in that while PSTN is centralized the H.323 is decentralized.

The architectural model of H.323 is peer-to-peer, the protocol design is based on the ISO QSIG standard and the services can be built using a multi-tier approach. Use of the QSIG reduces the complexity to interact with the circuit switched PSTN networks that also use QSIG. The multi-tier model allows complex services to be built of building blocks of simple services.

6.2 H.450 Supplementary services

The supplementary services of H.323 rely on the H.450 series of recommendations. The key elements of it are protocol based on the QSIG, peer-to-peer signalling and a multi-tier approach of building services. [4]

H.323 architecture uses high level Application programming interfaces APIs’, so that software vendors do not have work with low level implementation details, which would decrease interoperability risks.

6.2.1 H.450 based on ISO QSIG

The installed base of private telecommunications networks that use QSIG is wide and thus the use of QSIG in H.450 greatly helps the inter-working with that base. The migration from PBX networks to H.323 multimedia networks is simplified as well. Simpler gateways are one more advantage of using a common standard the QSIG.

6.2.2 Based on peer-to-peer signalling

In this respect the H.323 network differs essentially from a circuit switched network. Like in the Internet, in H.450 the intelligence resides in the end and edge devices and the network simply routes the packets. The end device can be a PC or any IP phone and the edge device is a PBX or a consumer gateway at home location. The state of the calls is also distributed in the end and edge devices.

In the traditional circuit switch model the intelligence and the state of the call reside in the network. The ends and edges are simple phones that run a stimulus-response protocol.

In H.450 new services can be installed in the ends and edges like software packages in a PC. Any software house can develop services to this standard and sell them directly to the end-user. This simplicity and straightforwardness in deployment will certainly stimulate the growth of a service building software industry. It should be remembered that the potential incompatibility of services in end and edge devices will be catered for by the capabilities negotiation process.

In the switch model new services are installed in the switch and may result in upgrades in other parts of the network before they are available for the customer. The switch is more over not at all so open to packages of ‘outside’ vendors. Yet it has to be admitted that in the central model the deployment of new services can be simpler. On the other hand the switch is a single point of failure while a software PBX can be embedded in each desktop phone. In this respect the distributed model is more fault tolerant than the switch model.

6.2.3 The multi-tier approach

The modular nature of the multi-tier approach enables the creation of basic services out of building blocks of primitives. Compound services can then be created by utilizing two or more basic services. Finally applications can be built by using compound services.

Simple services are for example:

- Multiple call handling
- Call transfer
- Call forwarding
- Call park and pickup
- Call waiting
- Message waiting indication
- N-way conference

Examples of compound services include:

- Consultation transfer
- Conference out of consultation

Consultation transfer uses call hold, multiple calls and call transfer. Conference out of consultation consists of call hold, multiple calls and n-way conference.

In Consultation transfer the user can perform three operations:

1. Put a multimedia call on hold and retrieve it later
2. Call an other person and optionally alternate between the two calls, or
3. Transfer the call

In Conference out of consultation the user has also three options:

1. Put a multimedia call on hold and retrieve it later
2. Call another person and optionally alternate between the two calls
3. Merge the calls in one conference call

**Call center integration.** A call center gateway lets Web surfers with properly equipped multimedia PCs (typically with the right browser plug-in) connect to an existing Automatic Call Distributor (ACD) with Internet phone technology. This illustrates one of the major advantages of IP telephony — its ability to combine voice and data on a single line.

The main advantage the IP telephony brings to Call centers is skill based routing. An incoming call can be directed to a call taker that for example can speak the same language as the caller or is specialist in a field the caller wants help of. The call can also be directed to personal adviser.

Emergency services provide another example of an architectural conflict since, for example IP addresses have no correlation with geographic location.

7 An application example: IEPS

As an example of an application of IP telephony in the broad sense of the term application, this paper presents the basic requirements that IP telephony should take into account to support the International emergency preparedness scheme IEPS.

The ITU-T recommendation E.106 for emergency communications was first defined for PSTN and ISDN networks, but it was soon realised that this scheme had to be extended to cope with the next generation networks i.e. the Internet and especially IP telephony. In this regard the ITU-T Study group 16 is developing a new recommendation for International emergency multimedia service IEMS as an extension to E.106, to provide for enhanced emergency services over Internet based networks in the future.

The IEPS is needed when there is a crisis situation which causes abnormal telecommunication requirements for governmental, military, civil authorities and other essential users of PSTN. It allows authorized users to be able to access the International telephone service while the service is restricted due to damage, congestion and/or other faults. [6]

7.1 Overall functional requirements

The primary goal of IEPS is to support crisis management arrangements by increasing the ability of the essential users to communicate via the PSTN, ISDN, Public land mobile networks PLMN or IP telephony.

The basic requirements include:

- International and national preference schemes are independent yet compatible: one could be activated when the other does not need to be activated
- National preference scheme users may not get access to the international scheme, but authorized users of the international scheme must be able to use the national preference scheme
- In some national schemes IEPS features may be enabled permanently
- Calls originated by IEPS users should be given priority in the networks involved when IEPS is enabled
- There must not be any conflict between preference for a call from an essential user and a call priority for a non-essential user to an emergency service
- If call restrictions to certain specific destinations (countries or areas) have been set when IEPS is activated, these restrictions should not apply to IEPS users
- IEPS calls should be marked from end to end

[6]

7.2 Established Telecommunication services

The essential features of the E.106 for the IEPS in the well established circuit switched PSTN and ISDN networks include:

- Priority dial tone
- Priority call setup, including priority queuing schemes
- Exemption from restrictive management controls

In the United States the Government emergency telecommunications service GETS uses the High probability of completion HPC in SS7 signalling for marking emergency calls. It should be noted that HPC does not include pre-emption of existing calls. In the U.S. alternate carrier routing ACR is used in the GETS in case some inter exchange carrier is not available. GETS uses a non-geographic toll free universal access number.

Some countries use IEPS access lines where all calls have a priority, while in some other countries priority is applied on a per call basis only.
7.3 Next generation networks

The IEPS requirements of E.106 should also be fulfilled by newly emerging next generation networks especially the Internet. The packet switching technology provides a clearly different operational environment compared to the traditional circuit switched networks. Thus new aspects have to be considered but there also emerges the possibility of new innovative services based on the new features of packet switched networks.

Examples of the new features are:

- Quality, grade and class of service
- The flexibility of the emerging object oriented and distributed technologies

For IP telephony an IEPS indicator similar to that of the HPC has to be defined, but the IP indicator has to be applied throughout the call.

There is extensive work going on in the international, national and regional standardization bodies to define the next generation networks. It is of utmost importance that they shall now start the work on the adaptation of IEPS.

7.4 Quality

The quality of video in the Internet is poor and the audio quality is not high especially compared to the PSTN. Because H.323 is a higher layer protocol, it can utilize the quality mechanisms of lower level protocols like the IntServ/RSVVP and the Diff Serv. In fact the development of QoS in the Internet is a result of the introduction of multimedia services to the Internet.

OoS features have been built in all modern LAN equipment although some critics say that enough of ever cheaper bandwidth will cater for the new multimedia services and QoS will not be necessary. The codecs in use today squeeze an IP call to only about one 10th of the bandwidth of a traditional PSTN call and better codecs are on their way.

8 List of acronyms

<table>
<thead>
<tr>
<th>Acronym</th>
<th>Definition</th>
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<tbody>
<tr>
<td>ACR</td>
<td>Alternate carrier routing</td>
</tr>
<tr>
<td>API</td>
<td>Application programming interface</td>
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<tr>
<td>CIF</td>
<td>Common intermediate format</td>
</tr>
<tr>
<td>Codec</td>
<td>Compression/decompression</td>
</tr>
<tr>
<td>DCT</td>
<td>Discrete cosine transform</td>
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<tr>
<td>DiffServ</td>
<td>Differentiated services</td>
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<tr>
<td>DTMF</td>
<td>Dial Tone Multi-Frequency</td>
</tr>
<tr>
<td>GCC</td>
<td>Generic conference control</td>
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<tr>
<td>GETS</td>
<td>Government emergency telecommunications service</td>
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<tr>
<td>HPC</td>
<td>High probability of completion</td>
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<tr>
<td>IEMS</td>
<td>International emergency multimedia service</td>
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<td>IEPS</td>
<td>International emergency preparedness scheme</td>
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<td>IETF</td>
<td>Internet Engineering Task Force</td>
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<td>IMTC</td>
<td>International multimedia teleconferencing consortium</td>
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<td>IntServ</td>
<td>Integrated services</td>
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<td>IP</td>
<td>Internet protocol</td>
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<td>ISDN</td>
<td>Integrated services digital network</td>
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<tr>
<td>ISO</td>
<td>International Organization for Standardization</td>
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<tr>
<td>ITSP</td>
<td>Internet telephony service provider</td>
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<tr>
<td>ITU-T</td>
<td>International Telecommunication Union – Telecommunications Sector</td>
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<tr>
<td>MC</td>
<td>Multipoint controller</td>
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<tr>
<td>MCU</td>
<td>Multipoint control unit</td>
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<tr>
<td>MP</td>
<td>Multipoint processor</td>
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<tr>
<td>MCS</td>
<td>Multipoint communication service</td>
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<tr>
<td>MGCP</td>
<td>Media gateway control protocol</td>
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<tr>
<td>OSI</td>
<td>Open systems interconnection</td>
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<td>PBX</td>
<td>Private branch exchange</td>
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<td>PLMN</td>
<td>Public land mobile network</td>
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<tr>
<td>POTS</td>
<td>Plain old telephone services</td>
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<tr>
<td>PSTN</td>
<td>Public switched telephone network</td>
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<tr>
<td>QCIF</td>
<td>Quarter common intermediate format</td>
</tr>
<tr>
<td>QSIG</td>
<td>D-channel signalling protocol at Q reference point for PBX networking</td>
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<tr>
<td>RAS</td>
<td>Registration, admission, status</td>
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<tr>
<td>RFC</td>
<td>Request for comments</td>
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<tr>
<td>RSVP</td>
<td>Resource reservation protocol</td>
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<tr>
<td>RTCP</td>
<td>Real-time transport control protocol</td>
</tr>
<tr>
<td>RTP</td>
<td>Real-time transport protocol</td>
</tr>
<tr>
<td>SCN</td>
<td>Switched circuit network</td>
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<td>SDP</td>
<td>Session description protocol</td>
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<td>SIP</td>
<td>Session initiation protocol</td>
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<tr>
<td>TCP</td>
<td>Transmission control protocol</td>
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<tr>
<td>TIPHON</td>
<td>Telecommunications and Internet protocol harmonization over networks</td>
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<tr>
<td>TLS</td>
<td>Transport layer security</td>
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<tr>
<td>UDP</td>
<td>User datagram protocol</td>
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9 References


