Internet Telephony

An Internet Service Provider's Perspective

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INTERNET TELEPHONY:
An Internet Service Provider’s Perspective

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Abstract

The aim of this Masters Thesis is to propose to SYSteam Nät AB, a local Internet Service Provider (ISP) in Uppsala, Sweden, how to implement IP telephony in their existing IT-infrastructure as a service to their customers. Thus the perspective of the thesis will be that of a local Internet Service Provider. Three general areas are covered in the thesis: Market and Business Model, Technology, and Economics.

Important issues for SYSteam Nät AB as an established local broadband Internet Service Provider are to both retain present customers and to attract new customers. Some believe that offering value added services such as IP telephony could do this.

Implementation of IP telephony can be done in different ways to fulfil SYSteam Nät's requirements. The analysis leads to a proposal of how SYSteam Nät could implement IP telephony. This involves many multi-faceted business, technical, and financial issues; each aspect is examined in this thesis.

Sammanfattning

Avsikten med detta examensarbete är att komma med ett förslag till SYSteam Nät AB, en lokal Internetleverantör i Uppsala, om hur de, som en service till sina kunder, kan implementera IP telefoni i sin existerande IT-infrastruktur. Detta betyder att jag behandlar frågeställningen med en Internetleverantörs perspektiv.Tre huvudområden behandlas i examensarbetet: Marknad och Affärsmodell, Teknik och Ekonomi.

Som en etablerad lokal leverantör av Internet via bredband är det viktigt för SYSteam Nät AB att både behålla nuvarande kunder och att attrahera nya kunder. En del tror att man skulle kunna åstadkomma detta genom att erbjuda värdehöjande tjänster som IP telefoni.

Implementering av IP telefoni, som svarar mot SYSteam Näts krav, kan utföras på olika sätt. Analysen, som leder till ett förslag hur SYSteam Nät skulle kunna implementera IP telefoni, involverar många mångfasetterade frågeställningar av affärs-, teknisk- och ekonomisk natur. Var och en av dessa aspekter behandlas i rapporten.

Acknowledgements

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Chapter 1

Introduction

1.1 Problem Statement

The aim of this Masters Thesis is to propose to SYSteam Nät AB, a local Internet Service Provider (ISP) in Uppsala, Sweden, how to implement IP telephony (Internet Protocol telephony) in their existing IT-infrastructure as a service to their customers. Thus the perspective of the thesis will be that of a local Internet Service Provider. Three general areas are covered in the thesis: Market and Business Model, Technology, and Economics.

SYSteam Nät wants to review (1) if they should include IP telephony in their offerings to their broadband customers in Uppsala and if so, (2) how to implement it. First priority should be given to existing broadband customers among tenant-owner societies (“bostadsrättsföreningar”) and customers living in student housing and second priority to new broadband customers.

System Nät has at the outset put forward the following requirements on possible implementations of IP telephony:

- Easy and cost effective implementation,
- User-friendly,
- Should give increased customer value without subsidies, e.g. costs to be paid by customer, and
- Low initial costs for System Nät.

Key issues to address are:

- To register as tele operator/operator or not (regulatory issue);
- To develop their own IP telephony solution, or to buy, license, or collaborate with a third party;
- Collaboration with a Voice over IP (VoIP) gateway operator versus operating their own gateway; and
- Collaboration with a VoIP service provider versus operating their own service.

1.2 Introduction to IP telephony

For the purpose of this thesis VoIP is utilized as a technology to transport voice traffic on an IP network, whereas IP telephony is technology that places voice clients and voice applications as well as voice traffic onto an IP network [8]. Internet telephony is a term for telephony partially or entirely on the Internet.

Traditional circuit-switched telephony over the Public Switched Telephone Network (PSTN) uses a fixed 64 kbit/s full duplex connection, which has guaranteed quality and high availability [4]. IP telephony on the otherhand is packet-switched, i.e., voice traffic is transported as IP packets over IP networks, such as the Internet, using different paths and assembled at the destination. Factors such as delay, packet-loss, and jitter affect the quality of real-time communication such as voice and video communication. It is important that packet-loss is kept to a low rate and that most packets arrive on time and in the right sequence to maintain acceptable voice quality. Techniques such as forward error correction and traffic shaping can be used to lessen these problems. While the Internet is a best effort-network and doesn't offer any quality guarantees, a private IP network can easily offer a specific Quality of Service (QoS) within the (private) network as the network can be dimensioned for the traffic expected/produced. Thus IP telephony within the ISP's network **can** offer guaranteed QoS.

One aspect of IP telephony is that it is possible to control the quality of the call through using different Coders/DeCoders (CODECs) that increase or decrease the bitrate according to the available bandwidth. A lot of CODECs exist and in January 2005 the Swedish company Global IP Sounds' internet Low Bitrate Codec (iLBC) became the first speech CODEC to receive standardization approval from the Internet Engineering Task Force (IETF) [9].

Automatic CODEC negotiation (via SIP's SDP – see section 2.3) makes it possible to negotiate which CODEC to use during the setup of a call, instead of fixing the choice of the CODEC in advance. The use of voice activity detection together with different CODECSs makes the use of bandwidth more efficient. Efficient use of bandwidth is important for the access links to the Internet, as network congestion is most likely to occur on these links (see section 2.6.1).

The most common modes of operation and traffic examples for IP telephony are:

1) Computer to computer

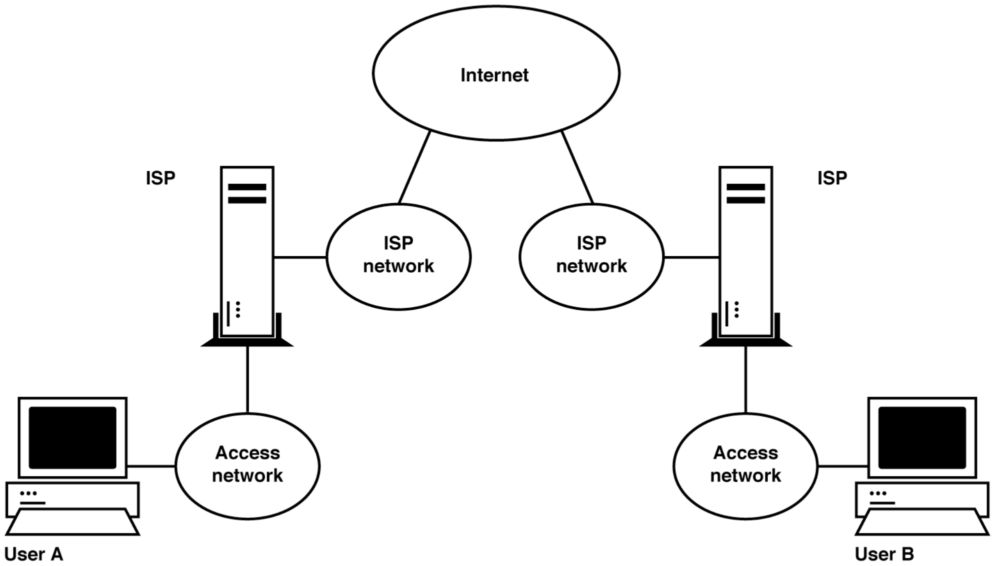

Figure 1.1: Computer to computer IP telephony.

The phone call is transported entirely over IP networks. Both computers need software phones, i.e., so called softphones.

2) Computer to telephone

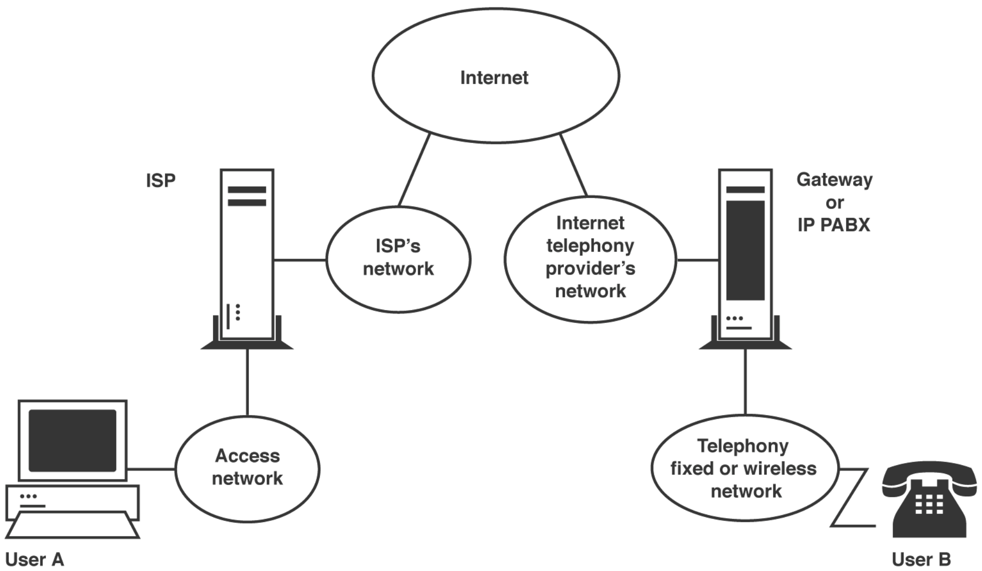

Figure 1.2: Computer to telephone or telephone to computer.

The phone call is transported as IP traffic to a gateway, where the IP packets are translated and transformed to PSTN traffic. Or the call can take the reverse path, as in the next case.

3) Telephone to computer

The call is transported as PSTN traffic to a gateway that transforms the call to IP packets and transports it as IP traffic to the computer.

4) Telephone to telephone

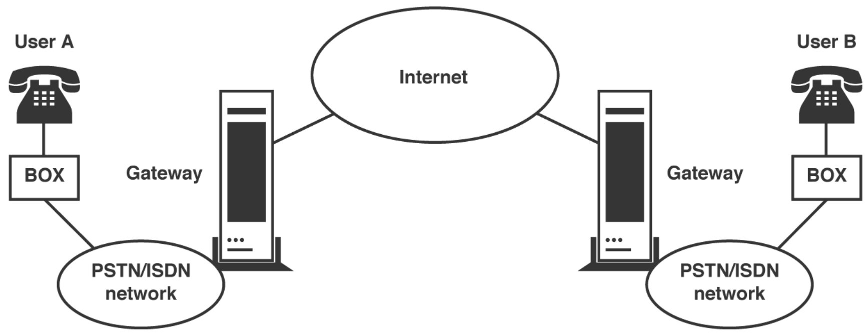

Figure 1.3: Telephone to telephone IP telephony using adapter boxes.

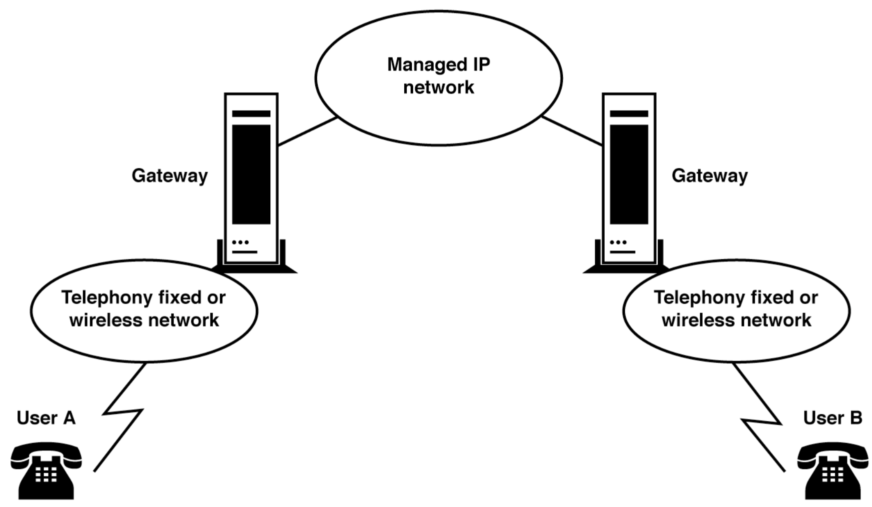

Figure 1.4: Telephone to telephone IP telephony using gateways.

The call is transported as PSTN traffic to a gateway; transformed to IP packets and transported as IP traffic to another gateway that transforms the IP packets back to PSTN traffic.

Today there are two major competing protocol families that have been developed to handle signalling for real-time sessions over packet-networks.

- The International Telecommunication Union (ITU) developed the H.323 standard. It includes a number of components that use signalling concepts from the traditional telephony industry. H.323 uses the sub-protocol H.225 for call control and signalling and the sub-protocol H.245 for media control and transport. A number of software components are defined by H.323.The H.323 terminal is an endpoint on a network that may provide real-time communication (voice, data, and video) with other H.323 Terminals, H.323 Gateways, and H.323 Multipoint Control Units. The H.323 Gatekeeper is optional and provides admission control and call-level, pre-call services to endpoints. The H.323 Gateway handles protocol conversion services and routing. The H.323 Multipoint Control Unit supports conference calls for both voice and video [4].

- IETF's Session Initiation Protocol (SIP) is a text-based protocol similar to Hyper Text Transfer Protocol (HTTP), which initiates voice, video, chat, interactive games, or virtual reality sessions between terminals. SIP handles signalling and control to establish, modify, and terminate sessions and carries other protocols such as the Session Description Protocol (SDP) which describes the media content of sessions and the Real-time Transport Protocol (RTP) which carries the actual voice, video, or other media content.

Regarding the choice of protocol the following is stated in [11]:

> "The choice of protocol need not be a technical decision, but rather a business choice. Companies that are heavily dependent on the PSTN for services may be best served by a methodical approach toward integration, using H.323 where it fits. Companies that are very Internet driven, with large IP networks, may find that SIP offers a superb solution for their internal telephony requirements. The key to the right choice is flexibility today and extensibility for tomorrow."

Given this, the choice of SIP for an ISP is obvious.

1.3 Historic overview

VoIP was first demonstrated in the early 1980s when Bolt, Beranek, and Newman in Cambridge, Massachusetts, set up the “voice funnel” to communicate with team members on the West Coast as part of its work with the Advanced Research Projects Agency (ARPA). The voice funnel digitized voice, arranged the resulting bits into packets, and sent them through the Internet [12]. The history of IP telephony in the market expanded in 1995 when the Israeli company VocalTec released their softphone InternetPhone® that enabled computer-to-computer IP telephony. In 1994 a M. Sc. Thesis project at Kungliga Tekniska Högskolan (KTH) [59] proved the feasibility and the advantages of replacing circuit switched speech communication with packet switching in a Local Area Network (LAN) environment, with the help of a gateway and a user interface. The first gateway between IP networks and the PSTN was released in the market in 1996. The Israeli company DeltaThree offered a telephone-to-telephone communication service over IP networks in 1997. In Sweden the operator Glocalnet offered IP telephony as early as 1999 [13].

Since 1995, IP telephony has developed rapidly to become a successful commercial service. TeliaSonera announced in March 2005 [60] that they plan to introduce a new generation of IP-based communication systems for both present and future fixed and mobile telephony services. In the most aggressive scenario the AXE-network for fixed telephony of today would be terminated and replaced by a converged network for integrated mobile and fixed telephony in five years (2007-2012). Today there are a number of operators in Sweden and many more in the rest of the world offering IP telephony.

Chapter 2

Background

2.1 SYSteam Nät AB – a broadband Internet Service Provider

SYSteam Nät AB, Uppsala (formerly known as Udac) has 25 years experience in installation, service, and documentation of cabling systems. The number of employees is currently 15. The company is a wholly owned subsidiary of SYSteam AB, a Swedish group offering sophisticated IT-based solutions to European customers. The group has around 1000 employees and a turnover in 2003 of 1046 MSEK. The number of employees in Uppsala is around 100.

SYSteam Nät is an authorised system integrator for general cabling systems from AMP and Avaya; and also a Premier Reseller for communication products from Cisco.

SYSteam Nät AB offers:

- Installation of general cabling systems,
- Fibre optic installations,
- Broadband installations in residential buildings,
- Operation of broadband networks in residential buildings,
- Internet access, and
- Wireless communication.

SYSteam Nät AB operates as a local broadband ISP in Uppsala.

2.2 An Internet Service Provider's challenge

The competition between broadband ISPs is increasing and broadband subscriptions in many markets including the Swedish market are commoditized. Today, ISPs need to differentiate their broadband offering in order to maintain their customers. Many ISPs believe it is necessary to offer new services, which raise the end-user value of their broadband subscription. Beyond basic call capabilities, ISPs have started to offer complementary multimedia services such as "TV" over broadband. Most of these services (e.g., call handling features, video telephony, video conferences) are provided by and charged for by the operator.

It is important to recognise that the telecom industry is changing and that communication is rapidly converging to IP as the standard. An example of this is the success of Skype™ [15] – which is by far the most successful peer-to-peer (P2P) VoIP solution available today (see section 3.8). "Skype could be a major step towards a change in the business model of the telecom industry – a model in which basic voice services are offered free of cost and are subsidized by revenues generated from value added services. Overall, Skype represents a massive discontinuity in the telecommunications industry, driving the convergence of voice and data much faster than originally anticipated" remarks Marc Vollenweider [14].

“Today is the moment for delivering IP telephony services!” That was a clear message at a conference (“IP-telefoni – Konkurrens på riktigt”) on February 2, 2005 organised by The Royal Swedish Academy of Engineering Sciences (IVA). Mr. Kennet Rådne, Head of Corporate Product & Services in Telia made a presentation, “VoIP-a step towards the integrated world”, containing following estimate of VoIP development in Europe [16]:

- 50 % of global traffic minutes will be carried on IP telephony networks by 2006 (source: IDC)
- Predictions are that no public switched voice services will be left by 2020 (source: Forrester Research)
- IP Centrex like services will comprise 20 % of business lines by 2005 (source: Gartner)

Broadband competitors to SYSteam Nät AB either have or are in the process of developing IP telephony services:

- Blixtvik Internet och Telefoni AB [37].
  IP telephony in operation (in collaboration with IP-Only) via an Analogue Telephone Adapter (ATA) (see Section 2.9).
- Bredbandsbolaget [38].
  IP telephony in operation via an ATA for their own broadband customers.
- Com Hem AB [39].
  IP telephony in operation for cable-TV customers.
- Tele2 AB [40].
  IP telephony via an ATA, regardless of ISP.
- Telia i Sverige AB [41].
  IP telephony (in collaboration with Hotsip [62]) with softphones.

2.3 Session Initiation Protocol (SIP)

SIP is an application-layer signalling-control protocol used to establish, modify, and terminate interactive multimedia sessions, such as voice, chat, video, interactive games, and virtual reality in an IP network. It is a text-encoded protocol based on HTTP and Simple Mail Transfer Protocol (SMTP) [4, 5].

The IETF Multiparty Multimedia Session Control (MMUSIC) Working Group began the development of SIP and since 1999 this has continued within the IETF SIP Working Group. The working group [17] developed SIP according to the following model:

1. Services and features are provided end-to-end whenever possible.

2. Extensions and new features must be generally applicable, and not applicable only to a specific set of session types.

3. Simplicity is the key.

4. Reuse of existing IP protocols and architectures, and integration with other IP applications, is crucial.

Request For Comments (RFC) 3261 [3], which defines SIP, states five facets to establish and terminate communications:

User location	determination of the end system to be used for communication
User availability	determination of the willingness of the called party to engage in communication
User capabilities	determination of the media and media parameters to be used
Session setup	"ringing", establishment of session parameters at both called and calling party
Session management	including transfer and termination of sessions, modifying session parameters, and invoking services

SIP does not provide services, QoS, or transport for large amounts of data and is therefore used in conjunction with other protocols to provide this [3, 5]. SIP can use any transport layer protocol, such as User Datagram Protocol (UDP) or Transmission Control Protocol (TCP), for its underlying transport. The SDP [18] is used to describe multimedia sessions, e.g. the type of media, ports, and CODECs. SDP is encoded in Multipurpose Internet Mail Extensions (MIME) and carried as a message body inside a SIP-message. RTP [19] is used to transport real-time data (e.g., voice, video, etc.) in the sessions and Real Time Control Protocol (RTCP) provides QoS feedback.

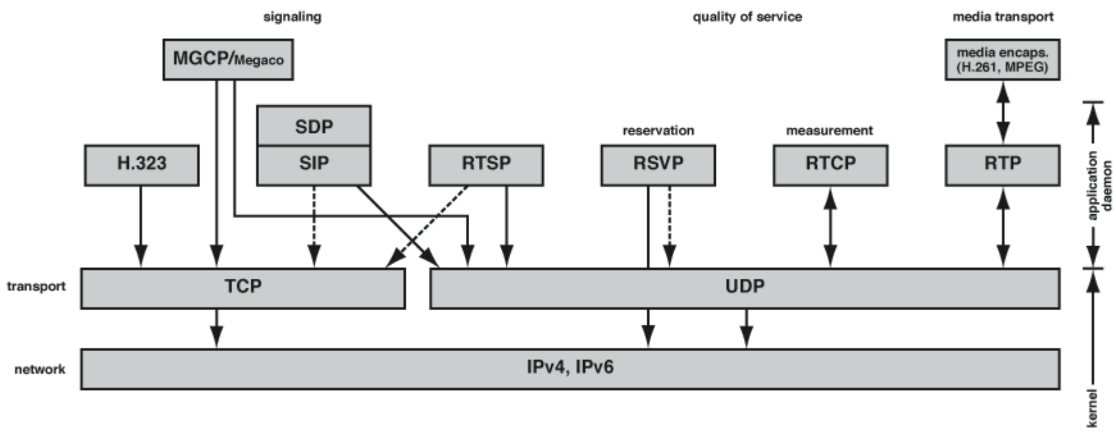

Figure 2.1: Open Systems Interconnection (OSI)-model showing how the different protocols are related.

2.3.1 SIP request types and response codes

There are six SIP methods defined in the basic SIP specification: INVITE, ACK, BYE, CANCEL, REGISTER, and OPTIONS [20]. Additional methods are extensions to SIP and defined in separate RFCs and Internet drafts. Response codes consist of six classes of numerical codes, differentiated by the first digit:
1xx: Informational
2xx: Success
3xx: Redirection
4xx: Client Error
5xx: Server Error
6xx: Global Failure

2.3.2 SIP addresses: URIs and URLs

Uniform Resource Identifiers (URIs) are used to designate participating parties in a session. URI's are independent of the location (a specific host) of a named object [5]. A Uniform Resource Locator (URL) describes the network location of a resource available on the Internet.

A SIP URI has a form similar to an email address, i.e., user@host. The user part of the address can be a user name or telephone number and the host part can be a domain name or network address. The older URI scheme is of the form: sip:erik@sipuser.com [20]. A newer scheme is a secure SIP URI of the form: sips:erik@sipuser.com [3], this uses Transport Layer Security (TLS) over TCP to provide transport security. Two different SIP URIs are used depending on whether the purpose is to contact a **user** or a user's device. When contacting a particular **user** the Address-of-Record (AOR) is used to identify the user and Domain Name System (DNS) Service Location (SRV) records are used to locate SIP servers for the domain. Alternatively, a Fully Qualified Domain Name (FQDN) is used to identify and contact a specific user's device supporting SIP.

Both User Agents and SIP Servers, can perform SIP address resolution by: DNS SRV lookup, Electronic Number Mapping (ENUM) lookup, or Location Server Lookup. Address resolution occurs at the start of the session and the results can be cached for future requests. This provides both mobility and portability, (although also loss of the possibility to cache the lookup results), as strengths of the SIP protocol [5].

2.3.3 Example of SIP session setup and teardown

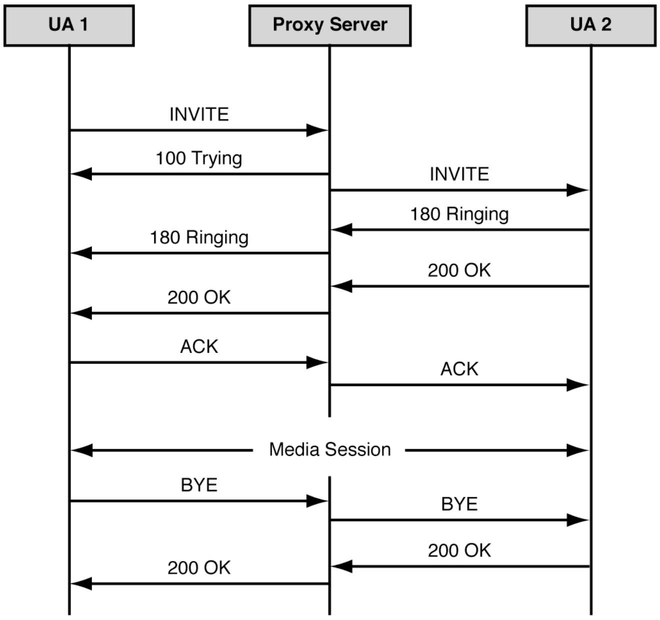

Figure 2.2: SIP session setup and teardown message flow.

2.3.4 Elements in a SIP Network

A SIP network consists of three types of elements: SIP User Agents, SIP servers, and Location Servers [5]. SIP User Agents are end devices such as: SIP phones, a software SIP client on a computer or handheld, or a gateway to other networks, typically a PSTN gateway. They can originate SIP requests and send and receive media. A SIP User Agent contains two parts: the User Agent Client (UAC) part initiates requests and the SIP User Agent Server (UAS) part responds to received requests.

SIP servers consist of Proxy, Redirect, or Registrar servers [20]. A SIP Proxy server is either outbound or inbound calls/session initiation and receives requests from a SIP user agent or a SIP proxy and forwards requests towards the intended destination. A SIP Redirect server receives requests from user agents or proxies and returns redirection information to contact an alternative URI. A SIP Registrar server receives SIP registration requests and updates Location Servers with SIP User Agent information. A Location Server is a database that contains user, routing, and location information.

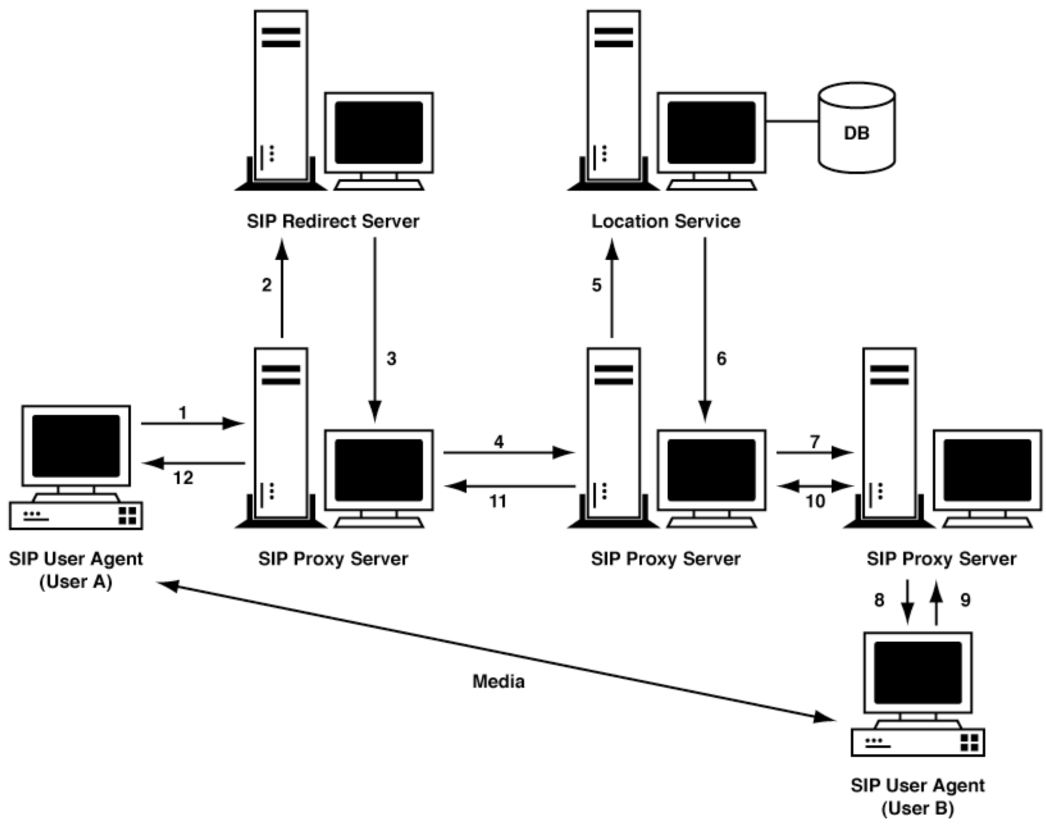

Figure 2.3: Elements in a SIP network.

2.3.5 Session Description Protocol (SDP)

SDP isn't really a protocol; it is more of a text-based session description format. It is used by SIP to convey information about a media session, e.g. type of media, the transport protocol in use, format of the media, multicast addresses, port numbers, etc. [18]. SIP carries SDP encoded in MIME as a message body in a SIP message. SDP supports use of IPv6 addresses.

A next generation protocol, SDP Next Generation (SDPng), is being developed by MMUSIC Working Group [61]. It is designed to address SDP's flaws.

2.3.6 Real-time Transport Protocol (RTP)

RTP is an Internet Standard defined in RFC 3550 [19]. RTP provides end-to-end network transport functions suitable for applications transmitting real-time data, such as audio, video, or simulation data, over multicast or unicast network services. RTP does not address resource reservation and does not guarantee quality-of-service for real-time services. The data transport is augmented by a control protocol (the Real Time Control Protocol – RTCP) to allow monitoring of the data delivery in a manner scalable to large multicast networks, and to provide minimal control and identification functionality. RTP and RTCP are designed to be independent of the underlying transport and network layers. RTP provides sequence numbers for ordered delivery and also time-stamps in packets that allow compensation for random delays by using play out buffers at the destination.

2.3.7 A summary of SIP

Encoding	Textual
Use in 3G Partnership Project (3GPP)	Yes
Call set-up delay	1.5 Round Trip Times (RTTs)
Complexity	Moderate
Extensibility	High
Architecture	Modular
Instant messaging support	Yes
Firewall support	Comparable to H.323
Addressing	URL
Transport Protocol	UDP (mostly), TCP, TLS, SCTP
Inter-domain call routing	Hierarchically by DNS
Service standardisation	None. “Standardise protocols, not services” [23]

Table 2.1. A summary of SIP adapted from [23]

2.4 Firewalls and NATs

Network Address Translation (NAT) is a method by which IP addresses are mapped from one address realm to another, providing transparent routing to end hosts [24]. NAT’s primary use is to create private IP networks that use internal IP addresses that are not part of the public Internet address space and are not routed over the Internet [5]. NATs are widely used because of lack of public IPv4 addresses or to avoid reconfiguring IP devices when changing service provider. Use of IPv6 addresses is thought to reduce the need for NATs in the future.

Firewalls are used where private IP networks interconnect to the public Internet. A firewall acts as a one-way gate, allowing requests to go from the private network into the Internet, and allowing only responses to those requests to return, but blocking most requests originating in the Internet destined for the private network. However, some incoming traffic is allowed to specifically enabled services.

When a SIP User Agent (UA) uses UDP as the transport protocol to initiate a session, a server outside the firewall will receive the SIP message, but the response will be blocked, since it is not associated with an outgoing request, over a TCP connection. When using TCP, a SIP session will be established, but the RTP media packets sent by the called party will be blocked by the firewall. The result is a one-way media session [5]. Firewalls and NATs cause problems for SIP traffic wanting to traverse it unless the device is SIP aware.

SIP works on port 5060, but SIP only handles the set up of a session. The session description in SDP tells what media, IP addresses, and ports to use for the media streams. These ports are ephemeral and dynamic. However, the firewall doesn’t know if an audio RTP stream should be let through, denied, or dropped [25] unless it understands the SIP messages (i.e., it is SIP-aware). This is because IP addresses and port numbers are inside SIP and SDP, and NATs generally only modify the IP headers, this can only work if the NAT is SIP aware. Several solutions have been developed to address these problems, but they each have their limitations.

- Simple Traversal of UDP through NATs (STUN) is a protocol that allows applications to discover the presence and types of NATs and firewalls between them and the public Internet. It also provides the ability for applications to determine the public IP addresses allocated to them by the NAT [26]. However, STUN doesn't enable incoming TCP connections through a NAT.

- Another solution for NAT problems is to use a SIP Application Level Gateway (ALG), which analyses SIP traffic and exchanges proposed addresses. An ALG is an application specific translation agent that allows an application on a host in one address realm to connect to its counterpart running on a host in a different realm transparently [24].

- Universal Plug and Play (UPnP) is an architecture, which builds on Internet standards and technologies, for pervasive peer-to-peer network connectivity of PCs and intelligent devices or appliances. UPnP enables devices to automatically connect with one another and work together. The UPnP NAT traversal solution, developed by the UPnP Internet Gateway Device (IGD) Working Committee, achieves automatic NAT traversal for real-time communications, etc., by providing methods for the following:
    - Learning public IP address,
    - Enumerating existing port mappings,
    - Adding and removing port mappings, and
    - Assigning of lease times to these mappings.

  Device vendors can add software or firmware to their device to support UPnP and then other devices and software can communicate with the NAT device using this same technology. The solution removes the need for writing and maintaining a database of ALGs to traverse NATs [63].

An example of a product that supports SIP and solves the firewall and NAT problems is the Intertex IX66 router/firewall [27].

2.5 Security protocols

There are different approaches to provide security for IP telephony and SIP-based calls. Two possible choices are IP Security Protocol (IPSec) or Secure Real-time Transport Protocol (SRTP) together with Multimedia Internet KEYing Protocol (MIKEY).

2.5.1 IPSec

The IETF has developed the IPSec suite, a set of IP extensions that provide security services at the network level. IPSec technology is based on modern cryptographic technologies, making possible very strong data authentication and privacy guarantees [28].

Because it secures the **network layer**, the IPSec protocol suite guarantees security for any application using the network layer. IPSec makes possible the realization of a secure Virtual Private Network (or secure VPN), a private and secure network carved out of a public and/or insecure network. Secure VPNs are as safe as isolated office LANs or Wide Area Networks (WANs) run entirely over private lines, and are far more cost effective [28].

IPSec Architecture

The IPSec suite provides three overall components: an Authentication Header (AH) for IP that lets communicating parties verify that data was not modified in transit and that it genuinely came from its apparent source, an Encapsulating Security Payload (ESP) format for IP that encrypts data to secure it against eavesdropping during transit, and protocol negotiation and key exchange protocol, Internet Key Exchange (IKE), that allows communicating parties to negotiate methods of secure communication [28].

2.5.2 SRTP

The Secure Real-time Transport Protocol (SRTP) was designed to create an efficient security solution for the Real-time Transport Protocol (RTP), which can work in constrained environments [29]. SRTP can provide confidentiality, message authentication, and replay protection to the RTP traffic and to the control traffic for RTP, i.e., RTCP. SRTP provides a framework for encryption and message authentication of RTP and RTCP streams. SRTP defines a set of default cryptographic transforms, and it allows new transforms to be introduced in the future. With appropriate key management, SRTP provides security for unicast and multicast RTP applications [30].

2.5.3 MIKEY

A security protocol such as SRTP cannot be used unless the communicating parties have agreed upon a set of parameters that SRTP needs (e.g., the encryption algorithm that should be used, keys, etc.); this includes one or more fresh cryptographic keys that can be used during the session. This raises the question of how to securely distribute this information prior to communication, which is generally solved by running a key management protocol. One such key management protocol that can distribute parameters and keys for SRTP is the Multimedia Internet KEYing (MIKEY) protocol defined in RFC 3830 [42]. MIKEY is executed in a single roundtrip. The authentication of the peers can be achieved either by using pre-shared keys or using digital signatures. The SRTP keys are in general not negotiated, but one party selects the keys and distributes them to the other party/parties. The reason for this is to make it possible to use MIKEY in a group communication scenario [29].

2.6 Quality of Service (QoS)

Network Quality of Service usually concerns the following areas:

- Bandwidth (minimum) – throughput
- Delay (maximum) – delay
- Jitter (delay variation) – delay jitter
- Information Loss (error effects) – packet loss
- Availability (reliability) – uptime, mean-time to failure, mean-time to repair
- Security (encryption, authentication)

Traditional IP networks only offer a best-effort service with no guarantees. Since IP-telephony is a real-time application it is important that data is delivered with the same time relationship as it was created. A number of factors have to be considered when providing IP telephony: bandwidth, delay, jitter, packet loss, availability, and security.

VoIP delay is the amount of time it takes for speech exiting the speaker's mouth to reach the listener's ear [4]. It is important that the total delay doesn't exceed the threshold required for interactive speech, which is around 180 ms, above that conversation becomes more strained and finally breaks down [6]. The ITU-recommended delay bound for speech traffic is 150 ms [43]. Jitter or delay variation is the variation of packet interarrival times.

In voice communications, packet loss shows up in the form of gaps or periods of silence in the conversation, thus leading to a "clipped speech" effect. One rule of thumb is that when the packet loss rate exceeds 20%, the audio quality of VoIP is degraded beyond usefulness, in part due to the bursty nature of the losses [6].

2.6.1 QoS options

Today network congestion most likely occurs on the **access links** to the Internet. Thus options to achieve QoS from a practical perspective are [5]:

- Provide adequate access bandwidth over the access line,
- Provide a simple QoS mechanism on the access line, such as using Type of Service (ToS) bits from the user's router to the first router in the ISP's network, or
- End-to-end QoS using standard IETF protocols for QoS combined with SIP signalling.

Different traffic handling mechanisms are defined for IP. Two of the most used are Integrated Services (IntServ) and Differentiated Service (DiffServ).

The Integrated Services (IntServ) QoS model

The IntServ QoS model enables a user to request a particular type of network service using the Resource Reservation Protocol (RSVP). RSVP uses policy-based control to determine if a user is authorised to request these resources from the network. Admission control is used to govern allocating and reserving of resources and path set up.

The IntServ model provides three levels of priorities: best effort, controlled load (for tolerant applications), and guaranteed service (for intolerant applications) [11].

RSVP requires that each routing node is aware of the QoS mechanism used and that the user application supports it. The problems with the IntServ model are poor scalability and the complexity of RSVP implementation.

The Differentiated Service (DiffServ) QoS model

The DiffServ QoS model categorises traffic into different classes of service (CoS). Similar services are then aggregated for similar handling and routed over paths configured to support that particular CoS. Packets are classified at the edge of the network, then forwarded according to a designated treatment for that CoS. To classify packets into a CoS, packet marking redefines the packet's ToS field and uses 6 bits of this field as a coding scheme. Per-hop behaviours govern how an individual class or aggregate is handled [11].

DiffServ scales to a very large network size and is simpler to implement than IntServ.

Multi-Protocol Label Switching (MPLS)

MPLS, in combination with effective traffic management facilities, provides the best approach for managing QoS in today's access networks and allows ISPs to deliver differentiated IP-based services [91]. MPLS is an efficient encapsulation mechanism. In MPLS, data transmission occurs on Label-Switched Paths (LSPs) as the "labelled" IP packets enter the IP access network and are forwarded from node to node along the path from source to destination, this moves traffic faster overall [98, 99, 100]. To support QoS requirements, network facilities must provide traffic management services such as admission control, traffic classification, traffic marking, and traffic conditioning. MPLS has a capability to allow network administrators to bypass potential points of congestion and direct traffic away from the default path selected by today's Interior Gateway Protocol (IGP)-based networks and deliver precise control of network traffic and behaviour. MPLS offers a practical QoS solution for ISPs that want to deliver differentiated, IP-based services [91]. MPLS provides a platform to begin deploying voice, video and data over a single network [99]. Using dynamic bandwidth allocation and packet classification, time sensitive, mission critical, and best effort applications can be simultaneously supported while providing full QoS [100].

Authentication, Authorisation, and Accounting (AAA)

AAA is necessary to decide who you are, if you are allowed to ask for a QoS service, and how much you should be charged [6]. To read more about AAA see RFC 3702 [32].

2.6.2 Service Level Agreements (SLAs)

A Service Level Agreement (SLA) is a contract between; for example, a network service provider and a customer that specifies, usually in measurable terms, what services the network service provider will furnish. Some metrics that SLAs may specify include [79]:

- What percentage of the time services will be available,
- The number of users that can be served simultaneously,
- Specific performance benchmarks to which actual performance will be periodically compared,
- The schedule for notification in advance of network changes that may affect users,
- Help desk response time for various classes of problems,
- Dial-in access availability, and
- Usage statistics that will be provided.

An SLA also contains a remedies section stating what will happen if an ISP fails to meet the terms of the agreement [80]. The SLA defines service features – such as availability, provisioning time and quality characteristics [49]. Via the SLA, the quality of the service is measurable and accountable. Network statistics are necessary in order to report SLA compliance to customers. Critics indicate that what many ISPs call an SLA is really just a description of the service you have bought and paid for from the ISP who is essentially promising to run its network as best it can [81]. The hard part of being an ISP is dealing with end-to-end problems such as customers having problems connecting to the ISP's network, when the problem may be somewhere else in a global chain of networks. The customer expects the local ISP to act as his representative to the rest of the world and get his packets through, on time, and without loss or corruption. However, no chain is stronger than the weakest link. This means that SLAs *between* ISPs are important.

2.7 ENUM

ENUM is a telephone number mapping function that translates a telephone number to an Internet address [2]. This is useful for a PSTN user with a dialling pad that wants to reach an IP telephony user. IETF's E.164 Number Mapping standard uses the Domain Name System (DNS) to map standard ITU-T international public telecommunications numbering plan (E.164) telephone numbers to a list of one or more Uniform Resource Locators (URL). ENUM allows the entity assigned this telephone number to control which URIs are associated with the number [2]. SIP uses these URL's to initiate sessions [6]. The use of the DNS for mapping E.164 numbers is described in detail in RFC 3761 [33].

The Swedish National Post and Telecom Agency (PTS) suggests that ENUM should be introduced in Sweden in a recent report about ENUM [7]. The report also covers how ENUM affects different IP telephony traffic cases.

2.8 Regulatory issues

2.8.1 The Electronic Communications act

The Swedish Electronic Communications Act (SFS 2003:389) [1] is a regulatory framework for electronic communications networks and services. It is based on EU directives, and took effect in Sweden on July 25, 2003. It affects IP telephony by stating several obligations for **publicly available** telephone services in the areas of emergency calls (112), number portability, and legal intercept.

The Electronic Communications Act does **not** define the term telecommunications operator, as was the case in the preceding legislation, the Telecommunications Act. It defines the term **operator** as one in possession or in other ways in charge of a public communications network or auxiliary installation. In the case of IP Telephony, both those who handle content and those in charge of signalling can be seen as operators [1, 2].

The Electronic Communications Act also contains following definitions:

Public telephony network	an electronic communications network used for the provision of publicly available telephony services and which enables the conveyance of speech, telefax messages, data communications, and other forms of communications between network termination points.
Telephony service	an electronic communications service that implies the possibility to place or receive calls through one or several numbers within a national or international numbering plan, including emergency calls.

With The Electronic Communications Act comes a notification duty, e.g., Chapter2, Section1, states: "A public communications network, usually commercially provided, or other publicly available electronic communication services may only be provided following notification to the authority appointed by the Government (the supervisory authority)" [1, 2].

2.8.2 Obligations

The Electronic Communications Act states in chapter 5, section 7, the following general obligations for an operator who provides a publicly available telephony service [1, 2]:

1. "make sure that the service and the public telephony network to a fixed network termination point satisfies reasonable demands on functionality and technical safety, as well as durability and availability at extraordinary events during peacetime,

2. contribute to enabling interruption-free conveyance of emergency calls, free of charge for the user,

3. to the extent technically feasible, provide location data to the party receiving emergency calls,

4. under conditions that are fair, cost-based, and non-discriminatory satisfy every reasonable request to hand out subscriber information, not subject to secrecy or a duty of confidentiality according to law, to one who conducts or intends to conduct subscriber information activity,

5. provide a subscriber free of charge with a detailed telephone bill that concerns the use of a public telephony network to a fixed network termination point or thereto belonging to publicly available telephony services, unless the subscriber has requested that the bill should not be so detailed,

6. make sure end-users in other countries within the European Economic Area can reach Swedish numbers, whose numbering structure lacks geographic significance, if it is technically or economically viable and the called subscriber has not chosen, for commercial reasons, to limit their access to calls from certain geographic areas,

7. consider the needs of persons with disabilities for special services, and

8. calls that are free of charge for the calling subscriber may not be stated in a telephone bill.

The Government or the public authority appointed by the Government may issue regulations on the manner in which the obligations shall be satisfied and on matters concerning exemptions from the obligations."

2.8.3 Emergency Calls (112)

Chapter 5 section 7 states that those who provide publicly available telephone services are obligated to [1, 2]

- "contribute to enabling interruption-free conveyance of emergency calls, free of charge for the user,
- to the extent technically feasible, provide location data to the party receiving emergency calls,"

With regard to this, two definitions are significant [2]:

Emergency calls	calls to public emergency services through a number within a determined numbering plan for telephony.
Location data	data which is treated in an electronic communications network and which provides the geographic position of a user's terminal equipment.

SOS Alarm is responsible for receiving emergency calls (112) in Sweden. They have 20 SOS Alarm centres spread about the country.

There is no problem for IP telephony to convey the emergency call using for example a PSTN-IP gateway. The problem for IP telephony is to switch the emergency call to the closest or preferably the most available (i.e. least busy) emergency service centre and to resolve the location and identity (to minimise prank calls) of the IP telephony caller [2].

When placing a call to the emergency number 112 within the PSTN in Sweden, the actual B-number (the A-number is the telephone number of the origination of the call and the B-number is the telephone number of the destination of the call) is determined by the Local Exchange (LX), based on a county ID, depending on the geographic location of the LX. A transit switch (Foreign Exchange, FX) then switches the call, with the help of the (new) B-number, to the correct emergency service centre. The centre receives the (new) B-number together with the call and therefore knows the geographic area (the county ID) from where the call originates [2].

An emergency call placed from an IP network will be switched to an emergency service centre based on the geographic location of the PSTN-IP gateway that handles the call. Unfortunately the IP telephony caller might not be in the same county or even country as the PSTN-IP gateway. A conclusion is that an emergency call could be re-routed to a suitable centre after communication with the caller [2] or that the information from the SIP message can be used to determine the appropriate centre.

The A-number of an emergency call originating in the PSTN is forwarded via Signaling System 7 (SS7) to the emergency service centre. This tells the centre the address of the subscriber in a (telephony) directory and enables dispatching of aid to the right location without further communication [2].

An IP telephony operator would need to keep a record of its subscribers to locate the emergency call or the necessary information could be in the SIP message. Even if a subscriber's call goes through the same PSTN-IP gateway, it doesn't mean it has the same (physical) origin [2].

2.8.4 Number Portability

Obligations on number portability are stated in Chapter 5, section 9 [1, 2]:

> "A party providing publicly available telephony services shall ensure that the telephony network allows a subscriber to retain his or her telephone number on changing service provider. If the subscriber so requests, telephone numbers used for such service shall be transferred to another party in order that such party shall provide these services.
>
> Telephone numbers whose number structure has geographic significance only need to be transferred for provision of telephony services within the same area code district."

The legislations concerns only numbers and not addresses, although extending the scope raises some interesting points[2]. In IP-telephony number portability is even more complicated than in traditional telephony. In SIP-based IP-telephony mapping between a number in E.164-format and an IP-address is done in two parts [2]:

1. E.164 mapping to SIP URL
2. SIP URL to an IP address

Below two ways are described regarding how to retain a user's E.164 number, when moving from a traditional telephony operator to a SIP operator (via a PSTN-IP gateway) [2].

1. The user's portable E.164 number is transferred to the new operator through mapping the number to a non-portable E.164 number that is operator specific. The operator then uses ENUM to map the operator specific E.164 number to a user associated SIP URL. The Swedish Number Portability Administrative Centre (SNPAC) then handles the translation between the two E.164 numbers, as in traditional telephony.

2. SNPAC could support mapping of a user's portable E.164-numbers into a SIP URL. This would avoid requiring the SIP operator to allocate a new E.164 number for each user.

2.8.5 Legal intercept

An obligation concerning secret telecommunications interception is also included in the Electronic Communications Act. Chapter 6 section 19 states that an activity shall be pursued in such a manner that decisions concerning secret telecommunications interception and secret telecommunications monitoring may be executed without the execution being revealed, if the activity concerns the provision of [1, 2]:

1. "a public communications network not solely intended for the conveyance of signals via wire for broadcasting of sound radio programs to the public or other activities, as specified in Chapter, 1, Article 1, third paragraph, first sentence of the Fundamental Law on Freedom of Expression,

2. services within a public communications network, which consists of:

    a) a publicly available telephony service, to a fixed network termination point, which admits the conveyance of local, national, and international calls, telefax, and data communication with a certain stated lowest transfer rate, which admits functional access to the Internet,

    b) a publicly available electronic communications service to a mobile network termination point.

    The content and information about the telecommunications messages subject to interception or monitoring shall be made available so that information may be easily dealt with. A telecommunications message refers to sound, text, pictures, data or other information conveyed by aid of radio transmission or light emission or electromagnetic oscillations utilising a specially devised conductor.

The Government or, if authorised by the Government, the supervisory authority may issue regulations concerning matters addressed in the first and second paragraph, and in individual cases exemptions from the obligations of the first paragraph."

The separation of media content and signalling in IP telephony, causes IP telephony operators, who only handle signalling, to be unable to disclose any media content, since they don't handle this traffic. Furthermore, a SIP-user can encrypt the SDP-content of the signalling to make it difficult or impossible for the operator to disclose the addresses that are used in the media-traffic [2].

2.8.6 To be or not to be a teleoperator/operator

As outlined above there are several obligations for an operator who provides a publicly available telephony service – a teleoperator. These obligations have business, technology, and financial consequences leading to advantages and disadvantages of being one or not being one. Being one means the business model is either: build-it yourself or outsource. Not being one means the business model is resale (see Section 3.4 Business models) or not providing a public telephony service. Chapter 3, 4 and 5 deals with these issues in more detail.

2.9 VoIP gateways and ATAs

VoIP gateways or IP-PSTN gateways work as interfaces between the PSTN and the Internet or other IP networks. Incoming synchronous voice calls from the local PSTN are terminated at the gateway; the voice signal is encoded, and encapsulated into packets and sent as IP packets. Incoming IP voice packets are unpacked, decoded, buffered, and sent as synchronous voice to the local PSTN [2, 6]. VoIP gateways can also provide other basic functions, such as authentication and billing, and global directory mapping (i.e. translations between IP addresses or SIP addresses and the E.164 telephone numbering scheme.) [6].

An Analogue Telephone Adapter (ATA) is a device used to connect a standard analogue telephone to a computer or network so that the user can make calls over IP networks such as the Internet. There are several types of analogue telephone adaptors. All ATAs create a physical connection between a phone and a computer or a network device; some perform analogue-to-digital conversion and connect directly to a VoIP server, while others use software for either or both of these tasks [64].

Chapter 3

Market and Business Model

3.1 SYSteam Nät's current situation

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SYSteam Nät AB operates as a local broadband Internet Service Provider (ISP) in Uppsala. The customers today are mainly residential customers: tenant-owner societies and students in student housing. The major part of the students living in student housing in Uppsala are connected to SYSteam Nät via different networks (See section 4.1). A much lower penetration is at hand for tenant-owner societies. Exact numbers can't be revealed due to confidentiality reasons.

SYSteam Nät offers broadband to residential customers as Internet access with up to 10Mb/s full duplex over Ethernet, with a dynamic public IP address for 250 SEK/month. A central firewall service is optional and free of charge. The connecting fee is 495 SEK. The contract is a rolling 3-month contract with a 1-month period of notice. E-mail and web homepage are not offered. Internal security is provided through a private Virtual Local Area Network (VLAN) and external security through authentication when using the Internet through a central firewall service.

Tenant-owner societies need to have at least 20 flats subscribing in order to have a contract with SYSteam Nät. The company also offers Internet access to companies located in buildings connected to SYSteam Nät's dark fibre net. The offered bandwidth varies from 1 to 100 Mb/s over Ethernet.

3.2 SYSteam Nät's potential market

As SYSteam Nät wants to give priority to offering IP telephony to tenant-owner societies and students in student housing in Uppsala, thus I will only look at the potential market for these two categories.

3.2.1 Tenant-owner societies

According to statistics for the year 2003, Uppsala had 28,693 flats in tenant-owner societies, consisting of 2,613 in (semi-) detached houses and 26,080 in apartment buildings [44]. Around 995 flats in apartment buildings were under construction in 2004 [45]. This means that the total number of flats in tenant-owner societies in Uppsala should be around 30,000 at the end of 2005.

3.2.2 Student housing

There are around 9,900 rooms or flats in student housing in Uppsala owned by around 20 companies and foundations. Studentstaden i Uppsala AB is the largest owner with circa 6,200 rooms and flats [46]. Including ongoing construction, I estimate the total number of rooms and flats in student housing to increase to around 10,300 by the end of 2005.

3.2.3 Conclusion

The maximum market potential for 2005 is according to above approximately 30,000 customers in tenant-owner societies and 10,300 customers in student housing. The accessible market today, i.e. customers with possible access to SYSteam Nät's broadband, consists of around 3,000 dark fibre customers and 8,000 students through UpUnet-S (Uppsala University Computer Network for Students) - a theoretical accessible potential of 11,000 IP telephony customers *without* regard to technical limitations. Thus, the accessible market is 27 % of the maximum market potential.

This translates to following total market potential in value:

- A maximum market potential of 40,300 subscribers at 265 SEK/month, equivalent to ~11 MSEK/month or 128 MSEK/year (265 SEK/month is the average cost for the job stressed single and the teenager family subscribing to Company X [70])
- Accessible market potential of 11,000 subscribers at 265 SEK/month, equivalent to ~3 MSEK/month or 35 MSEK/year.

3.3 The Market in Sweden

"IT-barometern Hösten 2004" [47], concludes that the total market for IP telephony in Sweden (business and residential) is worth approximately 500 MSEK, with a yearly growth of around 10-15% during the coming next years. The main driver is lower technology and traffic costs.

PTS reported in February 2005 [70] that the number of IP telephony subscribers in Sweden nearly has tripled since midyear 2003, from 20,000 subscribers in June 2003 to 56,000 in June 2004. The leading player is Bredbandsbolaget that introduced IP telephony services during the second quarter of 2002.

Stelacon, a Swedish research and strategy company, published in March 2005 [68] a market research report showing continued good growth of both the broadband market and the IP telephony market. There is a big interest for TV via broadband. Introduction of triple play (Internet, Telephony, and TV) will be an opportunity for broadband ISP's to attract new customers. Stelacon believes that approximately 100,000 households will have broadband TV at the end of 2005. Today there are few users of broadband TV. At the end of 2004, approximately 100,000 households of a total of around 4,7 million (equivalent to 2.1%) had adopted IP telephony, a number that is predicted to grow to 350,000 at the end of 2005. So far only 10,000 or 1.3% out of 800,000 companies (nationally) have started to use IP telephony. In more than 90 % of the cases it's only a test of IP telephony involving some employees. IP telephony is still considered an advanced technology with certain shortcomings.

As a comparison to the market in Sweden, it is reported that by the end of 2004 there were just over 1 million VoIP residential subscribers in the U.S. The leading service provider is Vonage with around 400,000 subscribers. However, 2005 is expected to be the breakout year for VoIP services with subscriber growth accelerating and reaching just over 3 million by the end of the year. The number of subscribers is predicted to grow to over 16 million in 2008 [73]. Another study expects that the number of U.S. subscribers to residential VoIP services will grow to 27 million by the end of 2009 [76].

3.4 Business models

Broadband ISPs, like SYSteam Nät AB, have an interesting customer base for delivery of VoIP services. However, these companies must act quickly in order to avoid the competition cherry-pick customers from their existing customer base. The local broadband ISPs must try to create more value by bundling services and offering better QoS than competitors. They will always have an advantage in the initial sale to the customer of the broadband subscription, because of advertising, geographic preference, single-bill requirements, or simply brand loyalty [48].

There are four models for offering VoIP services to end customers: **build-it-yourself**, **outsource**, **resale**, and **recommending a suitable provider**. **Build-it-yourself** is of course a serious undertaking, technically challenging - especially without a skilled staff. What also makes this alternative unattractive for most companies is the billing and Operations Support System (OSS) platform, which are well known to any telephony veteran. **Resale** is the easiest method, but also the most dangerous from a long-term view, as the customers are “owned” by a third party company, who only pays a commission back to the ISP. Branding in this case is difficult. Customers could easily discover that they are not working with the ISP at all and as a consequence skip the ISP and go direct to the VoIP provider. There is also a risk of customer dissatisfaction both with the facility provider and the VoIP reseller. **Outsourcing** seems to be the most balanced solution from a risk and investment perspective. In this case the OSS and telephony back-office platform as well as termination and perhaps even network resources are handled by a company which has multiple other customers served by a scalable system. The existing staff that is already employed by the ISP can handle customer support such as on-site service and hot-line support (advice and trouble-shooting). The ISP “owns” the customer, thus avoiding customer confusion over branding and support, which may occur in a resale situation. Another reason for local broadband ISPs to outsource VoIP services to third party providers is to avoid high capital expenditure and recurring expense for experienced staff [48]. Finally there is also the possibility of just helping customers to freely choose an IP telephony provider by **recommending** suitable providers.

3.5 Customer requirements and attitudes

The customer requirements vary, depending on the needs of their applications and the services they buy from the ISP. Some of the common requirements are [49]:

- Connectivity,
- Throughput,
- QoS (guaranteed bandwidth, minimal delay, and minimal jitter),
- Security,
- High availability of the service,
- Service-level agreement reports, and
- Multi-services using a single connection.

A survey carried out in the U.S. in January 2005 states that there are several barriers to consumer adoption of VoIP [65]. A sample of 1,473 U.S. adults and 335 business decision makers participated in the study. Some of the barriers relate to attitudes toward the new service. Among those who are aware of VoIP, but do not use it, the results were:

- I just don't know much about VoIP (47 %),
- VoIP providers have failed to show me a convincing story (36 %),
- I'm waiting for VoIP to become more mainstream (34 %),
- Recommendations would help me move toward VoIP (27 %),
- Potential savings are just not worth the hassle (25 %),
- VoIP seems too complicated (equipment, installation) (22 %), and
- VoIP is an unproven technology (22 %).

Other barriers stem from the perceived drawbacks of the technology. Among those who are aware of VoIP, but do not use it:

- VoIP might not support 911 calling in my area (62 %),
- VoIP could be subject to security and privacy issues (60 %),
- In a power failure situation, VoIP would not work (58 %), and
- The quality of my calls could be worse than with traditional phone service (52 %).

A research exercise in the U.S. [71] focused on understanding the value of next-generation broadband voice services to customers. Twelve services were examined in detail to determine which ones had the greatest impact on demand. Click-to-Dial, Caller ID, and Video Calling occupy the top three spots in terms of their impact on broadband households demand. Single Voice Mailbox (for the wireless phone and home phone) occupies the fourth spot, and Call Screening and Routing takes the fifth spot. Perhaps the most interesting finding of all is that a broadband phone service, without any added next-generation services, has little value to broadband customers. Less than 5 % of the consumers say they would pay for a broadband phone service that does not include any additional next-generation services. This lead the research company to believe that existing offerings in the marketplace, without next-generation communication management and multimedia services, are selling only on the value proposition of the services they are bundled with, such as long distance discount packages.

3.6 ISP goals

ISPs have a number of goals to meet in order to make their business profitable [49]:

- Short development and deployment time of IP-based services,
- Scalable and stable backbone (core) network to support a large number of customers,
- Ease of network operations for maintenance purposes and for upgrading/modifying customer requests,
- Multiple services using the same network,
- Sub-supplier support, and
- Fast time to market.

3.7 Broadband competitors

3.7.1 Blixtvik Internet och Telefoni AB

Blixtvik is an Uppsala based broadband and IP telephony provider to companies, properties, and private persons in Uppsala, the Stockholm area, and Karlstad. IP telephony with an ATA is offered in collaboration with IP-Only. The initial fee is 375 SEK, the monthly fee is 62.50 SEK with ordinary call charges, and the ATA cost is 1,995 SEK [50].

3.7.2 Bredbandsbolaget (B2Bredband AB)

Bredbandsbolaget is the second largest broadband provider in Sweden after Telia with 335.000 customers representing a 24 % market share and the largest when it comes to broadband via cable networks [101]. IP telephony is offered with an ATA. Neither a computer nor an Internet subscription is necessary. The monthly fee is 99 SEK, without any initial fee. Ordinary call charges for all calls, except for free calls between their own subscribers. [38]. IP telephony was launched in 2003, and serves 80,000 customers today. Bredbandsbolaget recently launched IP-TV, thus offering full triple play. The Norwegian company Telenor signed on March 23, 2005 agreements with the shareholders of Bredbandsbolaget to acquire the company. The acquisition is expected to be completed during June 2005, subject to approval from the Swedish competition authority [101].

3.7.3 Com Hem AB

In April 2004 it was announced that an agreement between Cygate Måldata and Com Hem was signed regarding supply, implementation, and maintenance of all technical equipment in a triple play venture (broadband, TV, telephony) based on communication via the cable-TV net.

Com Hem offers telephony via an ATA. There is no initial fee, and three types of subscriptions are offered: 60, 80, 125 SEK/month with ordinary call charges. In February 2005 it was announced that an IP telephony offer was introduced in Eskilstuna as the third city after Linköping and Lund. [39]. Customers in Uppsala received a telephony offer in May.

3.7.4 Tele2 AB

Tele2 is supplying IP telephony via an adapter, regardless of the user's ISP.The starting package including an ATA costs 699 SEK, the monthly fee is either 0 SEK or 99 SEK/month, the later offering calls within Sweden free of charge. Other calls have ordinary call charges [40].

3.7.5 TeliaSonera Sverige AB

Telia is a part of the TeliaSonera Group since December 2002. TeliaSonera operates under the Telia brand in Sweden and Denmark. Telia has announced that during 2005 it will offer broadband, IP telephony, and TV, i.e. triple play or quattro play if mobile telephony is added, to 70 % of Sweden's households [51]. The initial target group for IP telephony is families with teenagers and students who have moved from their homes to study at universities [52]. Telia introduced, during the beginning of 2005, digital TV via it's broadband network to certain addresses in the 15 largest cities in Sweden [53]. Telia has gradually, step-by-step, started to become a strong competitor to Bredbandsbolaget in the fibre network market. The company succeeded during the past 8 months to connect 100,000 households who contracted for fixed access of 10 Mb/s. Today Telia has around 30,000 LAN-customers [54].

TeliaSonera announced in March 2005 that they plan to introduce a new generation of IP-based communication systems for both present and future fixed and mobile telephony services [60]. In the most aggresive scenario the AXE-network for fixed telephony of today would be replaced by a converged network for integrated mobile and fixed telephony services in five years (2007-2012). The new network will have a capacity of 8-9 million subscribers. TeliaSonera will make the first decision concerning the migration in 2006. A prerequisite is the new global standard, Next Generation Networks (NGN), currently developed by 3GPP and the European Telecommunications Standards Institute (ETSI), for integrated mobile and fixed networks, that will handle all kinds of real time services such as, telephony services, data services, and TV services.

On May 3, 2005 TeliaSonera announced that they were going to start tests in Denmark with a new concept where mobile telephony is integrated with IP telephony at home. "The aim with the tests is to gradually provide an integrated mobile and IP-based telephony solution where the customer only needs a single wireless phone for all of his or her telephone needs. The phone will serve as an IP phone within the four walls of a home. When the customer leaves home, the phone will automatically switch over to the mobile network. TeliaSonera will monitor and evaluate alternative technologies for integrated mobile telephony and IP telephony solutions. The possible launch in different markets depends on the availability of terminals and the general market conditions "[94].

Telia's broadband telephony, according to Kenneth Rådne [16], has the following characteristics:

- Access independent, based on IP at ≥ 128 kb/s,
- A platform from Hotsip [62]
- SIP-based,
- Software client based,
- Service integration with PSTN, mobile, Wireless Local Area Network (WLAN), and
- A part of a wider concept – integrated communications.

The charges are:

- Initial fee,
- Monthly fee, and
- Ordinary call charges – not flat rate.

3.7.6 Uppsala Stadsnät AB (Uppsala Metropolitan Area Network)

Uppsala Stadsnät AB, a daughter company to TDC Song, is a local company with a Metropolitan Area Network (MAN) accessible for approximately 75 % of all households and 90 % of all companies in Uppsala. Their network is linked to several regional, national and international networks. They have a broadband concept named SWOP, where the subscriber can choose one of five providers (Bahnhof/One.se, Tele2, Blixtvik, Ready, and Ownit) for Internet access. Uppsala Stadsnät AB also offers IP telephony from four providers (Tele 2, AllTele, Blixtvik, and Ready) and will soon also offer broadband TV from Canal Digital. Thus, Uppsala Stadsnät will within a short period be able to provide triple play to customers [92].

3.8 Skype

"Skype is the Global P2P Telephony Company ™ that is changing the telecommunications world by offering customers free, superior-quality calling worldwide" [15]. The Skype Group (Skype Technologies S.A.) is headquartered in Luxembourg. A Swede, Niklas Zennström and a Dane, Janus Friis created Skype. These persons were also founders of KaZaA, a program using peer to peer (P2P) to allow sharing of computer files (music, videos, games, software etc.). It is claimed to be the most downloaded Internet software so far [14].

Skype is a peer-to-peer (P2P) Internet telephony software, in which the call is routed directly between the computers of the users. Calls to other Skype users globally are free of charge. The income comes from SkypeOut, which allows users to make calls from PCs or Personal Digital Assistants (PDAs) to conventional fixed-line phones or cell phones. Skype software also allows file transfer across platforms and instant messaging to other Skype users. Currently, Skype does not target enterprise clients. Skype has, compared with other telephony providers, significant advantages such as low cost, better sound quality, improved security, and easy and low installation costs [14]. Critics point out that Skype's technology is still a closed, proprietary environment with access only for their own customers. Skype may, for survival, have to peer with other networks or risk losing customers to other, more open, networks [55]. More than 1.2 million people were using SkypeOut on April 15, 2005 [95].

On January 6, 2005 an interesting report was published with the title "Impact of Skype on Telecom Service Providers" [14]. The report concluded: "VoIP players not using P2P technology in the retail segment might not survive on their current business model, since P2P technology has an inherent cost advantage and is likely to retain a quality advantage as well. Skype's competitiveness in the enterprise segment will have to be seen, once Skype introduces its business solutions. In conclusion, Skype represents a major disruptive force for all telecom service providers and will result in a faster than expected convergence of voice and data traffic. Skype will accelerate the already underway change in telecom business models, to one where basic voice services are offered free of cost and are subsidised by revenue generated from value added services."

Skype had registered more than 114 million downloads (free of charge) and 9 billion call minutes served as of May 25, 2005 [15]. Skype might reach between 140-245 million retail users by the end of 2008 depending on the market [14]. Skype announced on April 15, 2005 the public beta launch of SkypeIn and Skype Voicemail. SkypeIn customers can receive inbound calls to their Skype client from ordinary fixed telephones or mobile phones. SkypeIn customers choose a country and area code and are assigned a regular telephone number. Users may purchase up to three numbers from their home country in Denmark, Finland, France, Hong Kong, Sweden, the U.K., and the U.S. during the beta period [95]. You can buy a phone number from Skype in a country where you don't live. This means that if your important contacts are in the U.S. and you live in Sweden, you buy a 12-month subscription to a U.S. phone number and area code, and your U.S. contacts can call you for much less. The price is 30 euros for a 12-month subscription including free voice-mail. Skype is working together with a partner on a router-based product to reach out to a larger group that doesn't want to have their computers running all the time [67]. Skype will integrate its software either into a cordless handset or a base station, which users can then connect directly to their router. Skype claims that its call quality is far superior to traditional calls using the PSTNs narrowband technology. Skype is using wideband CODECs, with a much broader frequency range.

Niklas Zennström predicts [66] that telephony will become a software application that you simply run on your device. This will completely change the rules, e.g. you can't charge for a call, just as you can't charge for other software tasks. Skype will make money offering a free service but charging for value-added services, such as SkypeOut and later for the Skype phone. Taking an increasingly pragmatic approach to its business Skype is now using affiliates to sell its voice mail service and other premium products. Already, some 1,800 organisations have agreed to sell Skype's products in return for 2-10 % of the revenue they generate [102]. Registered sites link to both Skype's home page and the Skype Store, where customers can purchase Skype's premium services from the affiliate website. The referral period credits publishers for all purchases made from return visits to the Skype Store for up to 30 days after the original click [103].

3.9 Microsoft

The global market intelligence and advisory firm IDC reports that Microsoft is poised for a major VoIP push [97]: "After years of quietly building a voice over Internet protocol (VoIP) strategy on a number of fronts, Microsoft is now stepping up its VoIP efforts and is poised for major initiatives in both the enterprise and carrier space. The centrepiece of this increased activity is SIP-based collaborative applications being developed for the Microsoft Office Live Communications Server (LCS) 2005 product. In the past, Microsoft has kept a low profile around its development and marketing strategy for VoIP. But all of that is about to change. The company has developed partnerships with major IP-PBX vendors, such as Siemens and Alcatel, to help jump-start the move into enterprise IP telephony. These partnerships represent an important step toward strengthening Microsoft's position in the VOIP space. On the flipside, however, many vendors in the IP PBX market worry about Microsoft as a competitor, especially in the area of the high-end collaborative applications. Real-time collaborative applications such as instant messaging and Web conferencing increasingly have VoIP features and this association will dramatically enhance VoIP's stature."

3.10 An ISP's IP telephony customer proposition

As an example of an end-user value proposition, the following is quoted from Hotsip [56]:

- "Get more value out of your broadband connection.
- Options for making phone calls cheaper (e.g. low-cost or free PC to PC calls).
- Voice quality that is better than received in the PSTN (better than in cellular networks).
- Inexpensive video telephony of high quality.
- More convenient call handling, such as click-to-call and control of incoming phone calls.
- A second telephony line (with its own telephone number), which is inexpensive to install and own.
- Get your own private personal number to bring with you for example when moving.
- Easy to use service with web-based self-provisioning available 24 hours a day."

Thus, there are many convincing reasons for an end-user to ask an ISP or others for an IP telephony solution.

3.11 Pricing residential VoIP

Normally there is an initial setup or service connection or service activation fee. Some companies as a promotion offer the first month of service for free, but are still charging an initial setup fee. Most companies have a monthly service fee, but a few are offering a yearly service fee, with a corresponding additional discount for paying in advance. Traffic fees for telephone calls are charged (particularly for calls terminated in the PSTN) and for video telephony, and video conferencing services. Finally there is a list of value added features such as PC to PC file transfer, games, additional phone numbers, and personal call handling control using presence-based routing, which many subscribers seem willing to pay extra for [57].

An offer from an IP telephony operator, using SIP, might be [58]:

Free of charge:

- Download of client software to PC,
- Calls to other subscribers of this operator and other SIP-users "on line",
- Voice mailbox with e-mail notification or e-mail transfer of the voice mail,
- Number representation a.k.a. "Caller ID",
- Transfer of calls, and
- Video telephony.

Possible extra charges:

- One or more telephone numbers, and
- To receive calls from the PSTN or mobile net.

Extra charges:

- Calls via the PSTN or mobile net,
- An adapter for use with a normal phone, and
- An IP phone (including setup).

Requirements: 128 kb/s or greater access link

3.12 IP telephony costs for a consumer

PTS has published a report [70] in which they have calculated the average cost for IP telephony from different providers. They did this for six different customer groups. The two extremes were the job stressed single and the teenager parents. The monthly cost for the first category varies between 80 SEK (Tele2 broadband) and 250 SEK (Com Hem Large). Corresponding figures for the second category were 155 SEK (Rix broadband) and 385 SEK (Wx3). Tele2 broadband telephony and Rix broadband telephony were generally speaking the cheapest alternatives. The report concludes that households with relatively low consumption of telephony can save money with changeover to IP-telephony especially if supplied via LANs or cable-TV networks. If the communication instead is via Asymetric Digital Subscriber Line (ADSL), where the consumer in most cases still has to pay a subscriber fee to TeliaSonera, the same cost saving is not available. Subscribers with low consumption gain from lower fixed monthly costs, while volume consumers have a smaller part of the total cost in fixed fees. Furthermore, pricing of international calls is of importance.

Research by Market Cohesion [72] shows that VoIP services rarely provide users a cost saving over conventional telephone services. “Calls between users are free, so if a very large proportion of calls are between users – say, relations or, in the case of businesses, branch offices – then those calls will be without charge. However, typical users unable to make such arrangements will not benefit”. In fact, for most users VoIP will prove to be more expensive than choosing appropriate packages from conventional providers [72]. On the basis of their cost analysis model, Skype would be significantly more expensive for the typical family. So the question is then why would so many users use Skype, Vonage, etc. My conclusion is that the cost analysis model used may not give full credit to these suppliers.

Chapter 4

Technology

4.1 SYSteam Nät's current situation

SYSteam Nät AB is a local broadband ISP (see sections 2.1 and 3.1) located in Uppsala, Sweden. They offer broadband to residential customers as Internet access with up to 10 Mb/s full duplex over Ethernet, with a dynamic public IP address. Internet security is provided through a private VLAN and external security through authentication and a central firewall service. Their own core network is a Gigabit Ethernet network today connecting, through dark fibre links, around 1100 active users in tenant-owners' societies and student housing. There is also Transit Access (TA) via SYSteam's office to Skanova and Uppsala stadsnät (Uppsala Metropolitan Area Network) with a capacity of 200 Mb/s. SYSteam Nät's office is linked to Uppsala University's network, UpUnet-S (Uppsala University Computer Network for Students), with around 8,000 students connected. Students via the university network, UpUnet-S, have access to the Swedish University Computer Network (SUNET) and other parts of the Internet [83]. UpUnet-S is a Metropolitan Area Network (MAN), while SUNET is a Wide Area Network (WAN). The university is an ISP in this case and delivers WAN-links and net logon machines. Different real estate owners and foundations that own the student housing buildings own the passive and active parts of the network infrastructure, while SYSteam Nät is responsible for operating the network and supplying support. Uppsala University needs to upgrade WAN-links and the owners of the buildings need to upgrade the active equipment, to be able to deliver appropriate QoS. The General Conditions for SYSteam Nät's residential broadband customers contain certain rather generic SLAs [90].

SYSteam Nät AB has so far had no other IP telephony activities than implementation of some tests with softphones (see section 6.1).

4.2 IP telephony solutions

4.2.1 General

The following best practices are recommended when designing an IP telephony network solution [84]:

- Deploy open-standards based solutions to reduce the likelihood of massive redesign down the road,
- Provision sufficient bandwidth on WAN links,
- Prioritize voice traffic over data, and
- Consider the adoption of an SLA with the ISP to specify network availabilities, throughput, and average round-trip delay.

Designing an IP telephony network solution includes these steps [84, 85]:

- Perform an assessment of the current network,
- Determine the additional bandwidth and performance that will be required for new voice services,
- Design a network topology,
- Do a capacity analysis,
- Do a signalling network design,
- Implement end-to-end QoS, and
- Upgrade the network as needed.

A complete end-to-end broadband IP telephony solution consists of Customer Premises Equipment (CPE), gateways, softswitches, Operational Support System (OSS), and Business Support System (BSS).

CPE — Customer Premises Equipment is the telephone or other service provider equipment that is located on the customer's premises (physical location) rather than on the provider's premises or in between. It can be owned by the customer or by the provider [86].

Gateways — Gateways are used to interconnect dissimilar networks or applications. An IP-PSTN gateway performs translations between the two types of networks, (See section 2.9).

Softswitch — Softswitch (software switch) is a generic term for any open application program interface (API) software used to bridge PSTN and VoIP by separating the call control functions of a phone call from the media gateway (transport layer) [87].

OSS — An Operations Support System is a set of programs that help a communications service provider monitor, control, analyse, and manage problems with a telephone or computer network. Sophisticated systems are needed for such activities as ordering and keeping track of network components, tracking usage, billing, and reporting [88].

BSS — A Business Support System is needed for billing- including order entry, customer self services, customer care, trouble ticketing, and customer relationship management [89].

4.2.2 Build-it-yourself

ISPs that pursue a "build-it-yourself" strategy, i.e. an ISP that chooses to build and operate its own VoIP infrastructure, must establish themselves as teleoperators, deploying softswitches, servers, and media gateways and obtaining collocation space and establishing interconnection agreements with multiple service providers. Following is a list of requirements for an ISP that chooses to build and operate its own VoIP infrastructure [74]:

- Softswitch management,
- PSTN interconnection,
- Voice transit and termination,
- Traffic analysis and capacity planning,

- Bandwidth management,
- Local number portability,
- Variable-rate or distance sensitive billing,
- Emergency number support (112),
- Operator services/directory assistance,
- Installation and Customer Premises Equipment (CPE) support,
- Regulatory compliance, and
- Testing equipment and integration expenses.

A Financial Business Case for a "build-it yourself" strategy for SYSteam Nät is presented in Section 5.1.1.

4.2.3 Outsourcing

ISPs can optionally work with a third-party vendor for some or all of its VoIP services. There are several companies in the Swedish market offering a variety of solutions that enable ISPs to outsource all or part of their VoIP services. As an ISP's VoIP requirements change, for example with increased penetration, or increasing focus on providing commercial services, a third-party vendor can often provide a migration path to adjust the mix of specific service capabilities that are outsourced. Partnering with a company also means technology risk avoidance as IP network technology continues to undergo rapid and continuous change [74].

Together with SYSteam Nät I have participated in a meeting with a possible Swedish collaboration partner. Due to confidentiality reasons I refer to this company as "Company X". The company cooperates with Citylink AB, from which they buy (all) call minutes. Citylink claims to lease Sweden's most modern commercial multiservice fiber-optic network for fixed telephony and data communication [82]. This network has eight connection points from north to south Sweden, where the traffic is switched locally. Citylink takes care of all SS7 communication to the rest of the world. The physical connections are Integrated Services Digital Network (ISDN) Primary Rate Interface (PRI multi). Each such link can handle 30 simultaneous calls. The domestic traffic is switched to other ISPs direct or indirect through transit with direct access or via public nodes at Netnod-IX (D-GIX), NorrNod, and SOLIX. The international traffic goes via powerful transit using leading ISPs. The domestic core network has at least 99.99% availability with an access capacity up to 10 Gbit/s. MPLS is used for routing. Different service levels are available: 99.7 %, 99.8 %, and 99.9 % guaranteed availability with troubleshooting started within one hour and support or service on customers' site within four hours. Different QoS levels are also offered: expedited forwarding, assured forward, and best effort giving different levels of priority to traffic. Citylink routes the traffic to/from Company X. Company X is using their own VLAN-structure for residential IP telephony sold as fixed telephony. The customer use their ordinary analogue telephone connection. Analogue/digital conversion could in certain cases for example be performed via an Ericsson DRG-22 Residential Gateway in the basement of the residential building. With this solution SYSteam Nät should notify PTS that they would become a teleoperator. One advantage of doing this is that it enables SYSteam Nät to offer similar services as any Public Telephone Operator (PTO), and utilize their own prefix number. A separate application to PTS is required for the allocation and reservation of numbering capacity.

In this context there are also opportunities for a PTO to provide service elements for other ISPs, besides provisioning of high capacity transport services for national backbones. Examples of such services are [75]:

- Integration services (interconnection points and clearinghouses),
- Gateways between the Internet and the PSTN for IP telephony,
- Shared points of presence facilities,
- Billing and collection services, and
- Network maintenance.

I believe that it is likely that since TeliaSonera announced its move to VoIP to replace all AXEs within five years (see Section 3.7.5) that in due course they will provide several of above mentioned service elements for other ISPs. The same would apply to other PTOs like Tele2.

4.2.4 Resale

With a resale business model, another service provider, who only pays a commission back to the ISP, owns the customers. In this case the third party handles all technical matters related to IP telephony.

Together with SYSteam Nät I have participated in a meeting with a teleoperator, ”Company Y”, interested in a resale agreement. Company Y offers to residential customers a fixed telephone subscription via their broadband IP network not using the Internet. This means that the customer needs to have an ordinary analogue telephone and a connection kit, but not necessarily an Internet subscription. To be able to offer low prices Company Y does not offer connection to pre-select operators. SYSteam Nät's technical responsibility will in this case be their own network, including giving priority to voice traffic (QoS), between the residential customer and the connection point with Company Y's network in Uppsala. A residential gateway (ATA) for customers could either be supplied by Company Y or SYSteam Nät. Company Y terminates all calls in Uppsala.

4.2.5 Recommending an IP telephony provider

In this case the ISP simply helps its customers to choose an IP telephony provider by recommending suitable providers. The provider then handles all technical matters related to IP telephony.

4.3 Conclusions

From a technology point of view my conclusion is that SYSteam Nät is a too small player in a rather embryonic market to pursue a build-it-yourself strategy. Some of the technical barriers for this solution are [77]:

- Risk of purchasing a switch and billing system that does not work or deliver the required functionality and services,
- Lack of knowledge and experience, and
- Experienced technical employees are required to maintain equipment and applications.

The substantial list of requirements for a build-it-yourself strategy as outlined in Section 4.2.2 underscores that SYSteam Nät should not try to “reinvent the wheel” in terms of both applications development and network operations and infrastructure. ISPs will be better served and exposed to less risk by devoting their own resources to areas in which they have a clear core competency to leverage [74]. In the case of System Nät, at least for the time being, this does not include IP telephony. On top of all this you also have financial risks to overcome with a build-it-yourself strategy. These aspects are examined in more detail in next chapter.

Chapter 5

Economics

5.1 IP telephony solutions

5.1.1 Build-it-yourself

The build-it-yourself model, according to a report from Stratecast Partners in January 2005 [74], would force broadband ISPs to invest significant upfront capital in the form of softswitches, trunks, media gateways and port cards, session border controllers, test equipment, facility upgrades, and the back office systems required to provision and manage services. Additionally, collocation and backhaul expenses will be incurred. It would also require the ISP to build or expand their organisational and human resources. These costs must be incurred in *advance* of significant revenue generation and does not include the costs of developing and deploying new service capabilities. Costs will be higher in smaller/new markets, where economies of scale cannot be realised.

Financial Business Case for SYSteam Nät

I have with input from discussions with SYSteam Nät, e-mail correspondence with Cisco Sweden, and literature search made a rough indicative financial analysis for a build-it-yourself IP telephony solution for SYSteam Nät. Three alternatives with 100, 1,000 respectively 3,000 subscribers are analysed using Leida's VoIP model [104]. The model has a mixed top-down and bottom-up approach. That is, the model used an ISP's present cost (top-down approach) as an IP network costs and quantified the impact of IP telephony on these costs (bottom-up approach). The model used five types of subscribers, which are residential dial-in, business dial-in, 128 kb dial-in ISDN, 56 kb leased-line, and T1 leased-line subscribers. Hence, the ISP's overall cost distribution varied substantially according to the mix of subscribers. I have for my analysis used the T1 leased-line subscriber case although SYSteam Nät offers 10 Mb/s Ethernet to their dark fibre customers. The model considers five cost categories of an ISP: capital equipment, transport, customer service, operations, and other expenses.
These categories contain following cost elements:

Capital equipment	Gateways, softswitches, OSS, BSS, etc.
Transport	Leased lines, costs for the ISP to Network Access Point (NAP) etc.
Customer Service	Staff and facilities for supporting the customers, e.g. technical support to the subscribers etc.
Operations	Billing, facilities maintenance, operations personnel, etc.
Other Expenses	Sales/marketing, general/administrative expenses, etc.

The cost distribution for a T1 subscriber is according to Leida [105]:

- Capital equipment 8 %
- Transport 50 %
- Operations 6 %
- Customer service 8 %
- Other expenses 28 %

I have used following input parameters:

	100 Subscribers	1,000 Subscribers	3,000 Subscribers
Total yearly revenue, kSEK (calculated using 265 SEK as the monthly revenue per subscriber, see section 3.2.3)	**318**	**3.180**	**9.540**
Capital equipment investment, kSEK (according to Cisco Sweden email correspondence)			
-Gateways	**150**	**600**	**1.500**
-Softswitches	**800**	**800**	**800**
-OSS, BSS	**200**	**200**	**200**
Total capital equipment investment	**1.150**	**1.600**	**2.500**
Yearly depreciation on capital equipment investment, kSEK (using a 5-year straight line depreciation)	**230**	**320**	**500**

Table 5.1. Input parameters to Financial Business Case

If the capital equipment costs are set to 8% of total costs according to Leida's model, then we will have following *Financial Business Case* for one "normal" year:

	100 Subscribers	1,000 Subscribers	3,000 Subscribers
Total revenue, kSEK	**318**	**3.180**	**9.540**
Costs, kSEK			
-Capital Equipment	**230**	**320**	**500**
-Transport	**1.438**	**2.000**	**3.125**
-Operations	**173**	**240**	**375**
-Customer Service	**230**	**320**	**500**
-Other Expenses	**805**	**1.120**	**1.750**
Total costs	**2.876**	**4.000**	**6.250**
Contribution, kSEK	**- 2.558**	**- 820**	**+ 3.290**
Contribution margin	**Neg.**	**Neg.**	**34.5 %**

Table 5.2. Financial Business Case for one "normal" year

The financial contribution from IP telephony business will with above assumptions become positive from approximately 1,500 subscribers, which is ca. 14 % of the present accessible market for SYSteam Nät or 4 % of the present maximum market potential according to section 3.2.3. Another way to look at it is that it represents 50 % of SYSteam Nät's present dark fibre accessible customers. A weakness using Leida's cost model from 1997 is that it provides a snapshot in time. "Conclusions (and the models on which conclusions may be drawn) must be reassessed, in real time, as technologies, industries, and regulating environments evolve" [105]. There have been a small number of cost studies of VoIP in the academic literature [104], and I have not been able to find a more updated study than Leida's.

5.1.2 Outsourcing

Outsourcing allows an ISP to take advantage of the partner's existing facilities and (co-carrier) interconnection agreements with other network operators. It also allows successive scaling of capacity as service penetration grows. This helps reduce initial capital expenses requirements and limits operating costs. Partnering also provides other benefits, such as the ability to accelerate market entry, reduced demand on human resources, avoiding upgrade costs over time, and achieving faster return on investments in their total network investment [74].

The core offering of wholesale providers is the support for transport and termination of wholesale minutes [78]. Wholesale providers generally collect fees for successfully transporting and terminating minutes. Broadband ISPs, such as SYSteam Nät, should choose the carrier that provides the lowest cost, highest completion rate, and the best QoS. ISPs should look for wholesale terminating carriers that have broad network coverage. ISPs want to have one point of contact in order to not have to create agreements with several terminating carriers. Typically, wholesale terminating carriers and ISPs enter into contracts with an agreed price per minute for a specific time period.

5.1.3 Resale

In a resale arrangement with Company Y SYSteam Nät receives a fixed percentage of their invoiced value to the end customer. SYSteam Nät is responsible for the marketing of the telephony service to its existing broadband customers. Co-marketing arrangements are possible to new residential properties.

The customer pays a connection fee of 225 SEK. Calls to other subscribers within Company Y's network are free of charge. Four different subscription packages are offered with monthly fees between 0-50 SEK and the price for call minutes within Sweden is between 0-0.16 SEK depending on which day (Monday-Friday or weekends) and at what time (08-18 or 18-08) the calls are placed. There is also a connection fee of 0.40 SEK for every call except for calls within Company Y's own network. International calls are priced according to a separate price list. A call to a fixed or a mobile phone in the US costs 0.31 SEK per minute plus a connection fee of 0.40 SEK.

5.1.4 Recommending an IP telephony provider

It is likely that this activity could lead to a finder's fee paid by the provider to the ISP. I have not during my work found precise figures, but I would estimate the amount to 500 SEK per referred subscriber, which is equivalent to approximately 2 months revenue for an ISP (see section 3.2.3).

5.2 Conclusions

From a financial point of view my conclusion is that SYSteam Nät is a too small player in a rather embryonic market to choose a build-it-yourself solution. This conclusion is supported by the success of Skype and its introduction of new customer offers, that will threaten established tele companies and their strategies. Skype's CEO said in an interview in April, 2005: "We have around 34 million active users, of which 1,2 million are paying users. As we do not have any investments in networks or marketing, we don't need to earn money on all our customers. It's part of our business model just like Yahoo and Google.Within ten years there will be no subscription fees or call fees. The tele companies that have networks will probably survive as they can earn money on broadband and Internet access instead, but those who only sell call minutes need to reassess their business idea." [96].

As Stratecast Partners conclude [74]:"Pursuit of profit and cash flow requires not only identification and pursuit of new revenue opportunities, but also the ability to control capital expenditure, reduce recurring operating expenses, and optimise network operations". These results for SYSteam Nät can, with least risks, be achieved by partnering with a third party.

Chapter 6

In Practice

.

6.1 Tests performed together with SYSteam Nät AB

Test calls were performed with softphones in SYSteam Nät's dark fibre network in order to get a hands-on experience of measuring and evaluating the quality of calls and to see if SYSteam Nät could provide IP telephony in their access network. No tests were done in Up Unet-S due to an upgrade and maintenance project in progress.

6.1.1Test software

The following software, downloaded from the Internet, was used to perform the test calls:

- Xten's X-Lite SIP Softphone [106].
- Acterna PVA-1000 Capture Agent [107].
- Acterna PVA-1000 VoIP Analysis Software [107].

X-lite SIP Softphone is a free version of Xten Networks softphone X-PRO available for evaluation purposes.

Acterna PVA-1000 Capture Agent is used for advanced VoIP analysis and troubleshooting. It automates network packet capture on the end users desktop PC. Capture files can then be automatically forwarded to a server for evaluation and analysis by support personnel.

Acterna PVA-1000 VoIP Analysis Software performs complete analysis of capture files created with the PVA-1000 Capture Agent. The analysis is designed to identify problems and emulate the user experience with problem VoIP calls.

6.1.2 A Call

The software was installed on one laptop and one desktop PC, both connected to SYSteam Nät's dark fibre network. Test calls were executed between two X-Lite SIP Softphones through direct IP dialling using G.711 μ-law as codec. The laptop was connected to the network at a tenant-owner society building and the desktop was connected at SYSteam Nät's office. A radio program was transmitted as voice input. The capture agent then captured the calls and the analysis software searched for every potential RTP conversation in the capture file and presented a detailed VoIP analysis report of the call quality.

6.1.3 Analysis report

The VoIP analysis report presents call quality and call statistics.

Call Quality

The call quality is represented with Mean Opinion Score (MOS) and R Factor.
Call degradation factors reducing the R factor are also presented.
Three different MOS values are presented:

- Conversational Quality MOS (**CQ**) – a MOS score representing conversational quality considering delay and recency effects. Recency means that if an impairment comes near the end of a call, it lowers the overall call quality much more than if it occurred earlier in the call.
- Listening Quality MOS (**LQ**) – a MOS score representing listening quality without considering delay or recency effects. This value is similar to ITU P.862 Listening Quality implementations
- ITU P.862 Perceptual Evaluation of Speech Quality (PESQ or **PQ**) – a MOS score that is normalized to the P.862 raw quality score. PQ is an objective measurement that predicts the results of subjective listening tests. The resulting quality score is analogous to the subjective MOS.

Voice quality is represented with a MOS value between 1 and 5, where 5 is excellent, 4 is good, 3 is fair, 2 is poor, and 1 is bad.

Thee different R-factors are presented:
- The Conversation Quality R-factor (**R-Factor CQ**) is a voice quality metric that measures voice quality based on echo transmission delay, burst packet loss, and burst loss recency.
- The Listening Quality R-factor (**R-Factor LQ**) is a voice quality metric that does not take into account the effect of delay and recency.
- **G.107** is the R-factor calculated according to the ITU G.107 specification.

An R-factor of 93 is the accepted high-quality level for toll-quality voice transmission systems.

The reported call degradation factors are recency, echo level, silence period noise level, voiced segment signal level, one-way delay, voice codec selection, jitter buffer packet discards, and packet loss. The factors are presented in points that reduce an R-factor of 93. The sum of all degradation points is equal to the difference between 93 and the actual R Factor (CQ).

Call Statistics

The following statistics are listed for each channel of the currently selected call:
- **Max Jitter** shows the highest absolute jitter value, in milliseconds,
- **Avg. Jitter** shows the average jitter value, in milliseconds,
- **Max Gap** is the largest seen interframe gap,
- **Max Pkt Loss** shows the highest observed value for consecutive packet loss,
- **Avg. Pkt Loss** shows the average value for consecutive packet loss, and
- **Total Pkt Loss** shows the total value for packet loss.

6.1.4 Test results

Call Quality		Measure 1	Measure 2	Measure 3	Measure 4
Mos Value	**CQ**	4,00	4,00	4,00	4,00
	LQ	4,00	4,00	4,00	4,00
	PQ	4,00	4,00	4,00	4,00
R factor	**CQ**	91	91	91	91
	LQ	93	93	93	93
	G.107	91	91	91	91
Call degradation factors					
	Echo Level	1,0	1,0	1,0	1,0

Table 6.1. Call quality test results

Call statistics	Measure 1	Measure 2	Measure 3	Measure 4
Max Jitter (ms)	23,00	13,00	21,00	13,00
Avg. Jitter (ms)	3,00	3,00	2,00	4,00
Max Gap	260,00	87,00	330,00	93,00
Max Pkt Loss	0	0	1	0
Avg. Pkt Loss	0,00	0,00	4,00	0,00
Total Pkt Loss	0	0	1	0

Table 6.2. Call statistics test results

A VoIP analysis report for measure 2 is shown in Appendix B.

6.1.5 Conclusions

MOS values are 4,00, which is indicating good call quality. R-factors are close to 93, which is also indicating good call quality. A slight degradation of the call quality is due to echo, when transmitting voice from radio. Almost no packet loss was detected and the jitter was also reasonable. When playing back some of the captured calls, some packets were discarded regularly perhaps due to a full jitter buffer, resulting in the call sounding chopped.

The results indicate that SYSteam Nät's dark fibre network is adequate for IP telephony, but more tests are necessary to get a complete picture.

6.2 An automated test approach

An approach to verify an IP telephony implementation and ensure its ongoing operation is to manually test all of the phones. This approach to testing the functionality and voice quality of IP telephony is time-consuming, costly, and prone to errors. Moreover, this method does not scale to support large deployments of thousands of phones across many physical locations [93].

There is Network Management Software (NMS) that monitors metrics like packet loss and jitter. "Conventional NMS metrics such as Central Processing Unit (CPU) utilisation, network latency or packet loss does not easily translate to an end user's experience. In IP telephony, testing is not just a matter of "pinging" a signal to each phone to make sure it is connected to the network. Users are counting on accessing a broad array of phone functions and being able to clearly hear and communicate with whomever they talk to. Any attempt to implement and maintain IP telephony using NMS alone must be supplemented with significant manual effort" [93].

Today more sophisticated tools are available such as Clarus Systems [93] providing automated testing solutions to certify the implementation and assuring the ongoing operation of IP telephony. It uses software to automate manual processes [93].
Another example of this is the Acterna software that was used in the tests described in section 6.1.

Chapter 7

Conclusions and Future Work

This chapter is intended to summarise the conclusions of the previous chapters and also suggest areas where SYSteam Nät AB is recommended to do more work before they make a final decision regarding a possible introduction of IP telephony.

7.1 Market and Business Model

Broadband subscriptions are commoditized, which means that ISPs like SYSteam Nät need to raise the end-user value of their broadband subscription. One way of doing this is initially to introduce IP telephony followed by added next-generation services such as Click-to-dial, Caller ID, and Video Calling. Broadband competitors to SYSteam Nät either have or are in the process of developing IP telephony services and even triple play.

There are four models for offering VoIP services to end customers, in following falling order of revenue, complexity and undertakings:

- Build-it-yourself,
- Outsource,
- Resale, and
- Recommending a suitable VoIP provider.

Build-it-yourself is a serious undertaking, technically challenging especially without a skilled staff. Resale is an easier method, but also dangerous from a long-term view as customers could skip the ISP and go direct to the VoIP provider. The ISP "owns" the customer in an outsourcing arrangement, thus avoiding customer confusion over branding and support.

Pricing residential VoIP normally includes an initial setup or service activation fee. To this adds a monthly or a yearly service fee and traffic fees for telephone calls (particularly for calls terminated in the PSTN). Finally there is a pricelist for value added features. Calls to other subscribers of respective ISP and to other SIP-users "on line" are free of charge.

The maximum market potential for SYSteam Nät today, with the chosen Uppsala-strategy to target residential customers in tenant-owner societies and students living in student housing, is approximately 40,300 subscribers generating a revenue of around MSEK per year. The accessible market potential is presently around 11,000 subscribers, (3,000 dark fibre customers and 8,000 students via Up Unet-S), without regard for technical limitations. This represents a revenue of around 35 M SEK per year.

7.2 Technology

SYSteam Nät offers broadband Internet access to residential customers with up to 10 Mb/s full duplex over Ethernet, with a dynamic public IP address. Their own core network is a Gigabit Ethernet network connecting customers through dark fibre links. This means that they have full control of QoS on this network and can offer guaranteed QoS. Around 8,000 students are accessed via Up Unet-S, where Uppsala University is the ISP, while SYSteam Nät is responsible for operating the network and supplying support. Results of tests indicate that if SYSteam Nät wants to introduce IP telephony to these students, Uppsala University would have to upgrade their equipment.

It is important to follow best practises when designing an IP telephony network, otherwise severe QoS problems will occur.

A complete end-to-end IP telephony solution would consist of CPE, gateways, softswitches, OSS, and BSS. SIP should be used as protocol. ISPs that choose to build and operate their own VoIP infrastructure must deploy the above equipment and establish themselves as teleoperators with quite a few rather demanding obligations stipulated by The Swedish Electronic Communications Act. ISPs can optionally work with a third party vendor for some or all of their VoIP services, i.e. follow an outsourcing strategy. As VoIP technology continues to undergo rapid and continuous change, this means technology risk avoidance. With a resale business model, another service provider or teleoperator owns the customers and pays a commission back to the ISP. The third party handles in this case all technical matters related to IP telephony. An option is also to recommend an IP telephony provider who then handles all matters, including technology, related to IP telephony.

From a technology point of view my conclusion is that SYSteam Nät is too small a player in a rather embryonic market to pursue a build-it-yourself strategy, with its quite high technical barriers and risks. The company should not try to “reinvent the wheel” in terms of both application development and network operations and infrastructure. This is outside SYSteam Nät’s current core competence.

7.3 Economics

The build-it-yourself model would force broadband ISPs to invest significant upfront capital in the form of equipment, facility upgrades, and back office systems. Costs (i.e. collocation, backhaul expenses, expanding organisational, and human resources) will be incurred in advance of significant revenue generation. Cost will be higher in smaller and new markets due to economies of scale reasons. A rough Financial Business Case for SYSteam Nät indicates that approximately 1,500 IP telephony subscribers are needed for a positive financial contribution. This number represents 50 % of the present dark fibre accessible customers and will most likely take some years to reach.

Outsourcing helps reduce initial capital expenses and limits operating costs. It allows successive scaling up of capacity. Partnering also provides the ability to accelerate market entry and avoiding upgrade costs. In a resale agreement SYSteam Nät would receive a fixed percentage of the invoiced value to the end customers, probably approximately 10 % of revenue. Recommending an IP telephony provider could lead to a finder’s fee paid by the provider to the ISP. My estimation is that the amount could be around 500 SEK per referred subscriber.

Thus from a financial point of view my conclusion is that SYSteam Nät is also too small a player in a rather embryonic market to choose a build-it-yourself solution. Partnering with a third party would reduce the financial and operating risks considerably.

7.4 Final Conclusions and Future Work

My final over all conclusion is that SYSteam Nät AB *should* include IP telephony in their offerings to their existing dark fibre broadband customers among tenant-owner societies and customers living in student housing in Uppsala. That should be done either by an outsourcing or a resale solution. SYSteam Nät should do further work in order to evaluate these two alternative strategies including financial details. In order to do that the company needs to soon start in depth discussions with selected providers, leading to submitting of quotations, negotiations, and final choice of provider(s). SYSteam Nät should choose the terminating carrier(s) that provides the lowest cost, highest completion rate, the best QoS, and the broadest network coverage.

After a certain period, if the experience gained is positive, SYSteam Nät should also offer IP telephony to existing enterprise customers and to new dark fibre customers. I also recommend SYSteam Nät soon start discussions with Uppsala University in order to find out what needs to be done to upgrade their Up Unet-S network in order to accommodate IP telephony. The active parts of this network infrastructure owned by different real estate owners also need to be looked at. A feasibility study should be done before final decisions are taken regarding possible introduction of an IP telephony solution for students using Up Unet-S. As SYSteam Nät currently is responsible for operating this network and supplying support, it would probably also be beneficial for Uppsala University and the real estate owners to have the same company as an IP telephony provider. In the medium term SYSteam Nät should also evaluate introduction of broadband TV in order to offer triple play.

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Appendix A

A.1 Acronyms and abbreviations

3GPP	3G (third generation) Partnership Project
AAA	Authentication, Authorisation, and Accounting
ADSL	Asymmetric Digital Subscriber Line
AH	Authentication Header
ALG	Application Level Gateway
AOR	Address of Record
ARPA	Advanced Research Projects Agency
ATA	Analogue Telephone Adapter
BSS	Business Support System
CODEC	Coder/DeCoder
CoS	Class of Service
CPE	Customer Premises Equipment
CPU	Central Processing Unit
DiffServ	Differentiated Services
DNS	Domain Name System
ENUM	Electronic NUmber Mapping
ESP	Encapsulating Security Payload
ETSI	European Telecommunications Standards Institute
FQDN	Fully Qualified Domain Name
FX	Foreign Exchange
H.225	Protocol defined in ITU-recommendation H.225
H.323	Protocol defined in ITU-recommendation H.323
HTTP	Hyper Text Transfer Protocol
IETF	Internet Engineering Task Force
IGD	Internet Gateway Device
IGP	Interior Gateway Protocol
IKE	Internet Key Exchange
iLBC	Internet Low Bitrate Codec
IntServ	Integrated Services
IP	Internet Protocol
IPSec	IP Security protocol
ISDN	Integrated Services Digital Network
ISP	Internet Service Provider
ITU	International Telecommunication Union
KTH	Kungliga Tekniska Högskolan (Royal Institute of Technology)
LAN	Local Area Network
LX	Local Exchange
LSP	Label-Switched Path
MAN	Metropolitan Area Network
MGCP	Media Gateway Protocol (H.248, Megaco)
MIKEY	Multimedia Internet KEYing protocol
MIME	Multipurpose Internet Mail Extensions
MMUSIC	Multiparty Multimedia Session Control

MOS	Mean Opinion Score
MPEG	The Moving Picture Experts Group
MPLS	Multi-Protocol Label Switching
NAP	Network Access Point
NAT	Network Address Translation
NGN	Next Generation Networks
NMS	Network Management Software
OSI	Open Systems Interconnection
OSS	Operations Support System
P2P	Peer-to-peer
PDA	Personal Digital Assistant
PRI	Primary Rate Interface (ISDN)
PSTN	Public Switched Telephone Network
PTO	Public Telecommunications Operator
PTS	Post- och Telestyrelsen (The Swedish National Post and Telecom Agency)
QoS	Quality of Service
RFC	Request For Comments
RSVP	Resource Reservation Protocol
RTCP	Real Time Control Protocol
RTP	Real-time Transport Protocol
RTSP	Real Time Streaming Protocol
RTT	Round Trip Time
SDP	Session Description Protocol
SDPng	SDP Next Generation
SIP	Session Initiation Protocol
SLA	Service Level Agreement
SMTP	Simple Mail Transfer Protocol
SNPAC	Swedish Number Portability Administrative Centre
SRTP	Secure Real-time Transport Protocol
SRV	Service Location
SS7	Signalling System 7
STUN	Simple Traversal of UDP through NATs
SUNET	Swedish University Network
TA	Transit Access
TCP	Transmission Control Protocol
TLS	Transport Layer Security
ToS	Type of Service
UA	User Agent
UAC	User Agent Client
UAS	User Agent Server
UDP	User Datagram Protocol
UPnP	Universal Plug and Play
UpUnet-S	Uppsala University Computer Network for Students
URI	Uniform Resource Identifier
URL	Uniform Resource Locator
VLAN	Virtual Local Area Network
VoIP	Voice over IP
VPN	Virtual Private Network
WAN	Wide Area Network
WLAN	Wireless Local Area Network

Appendix B

B.1 VoIP Analysis Report

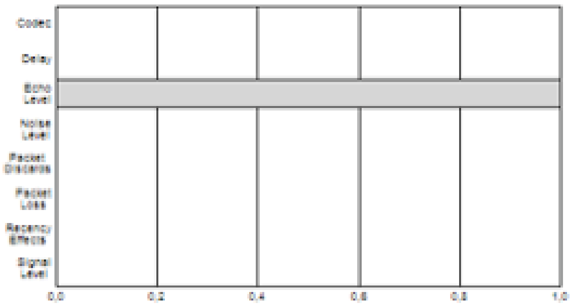
Call #1: 82.196.162.254:8000 (G.711 u-law) - 82.196.163.210:8000 (G.711 u-law)

Call Quality

	MOS Value			R Factor		
Source	**CQ**	**LQ**	**PQ**	**CQ**	**LQ**	**G.107**
82.196.162.254:8000 (G.711 u-law)	4,00	4,00	4,00	91	93	91
82.196.163.210:8000 (G.711 u-law)	5,00	5,00	5,00	100	100	100

Voice Quality	5	4	3	2	1
	Excellent	*Good*	*Fair*	*Poor*	*Bad*

Call Degradation Factors



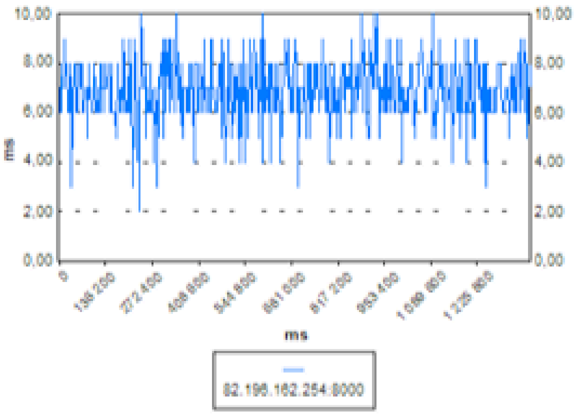
82.196.162.254:8000 (G.711 u-law)

82.196.163.210:8000 (G.711 u-law)

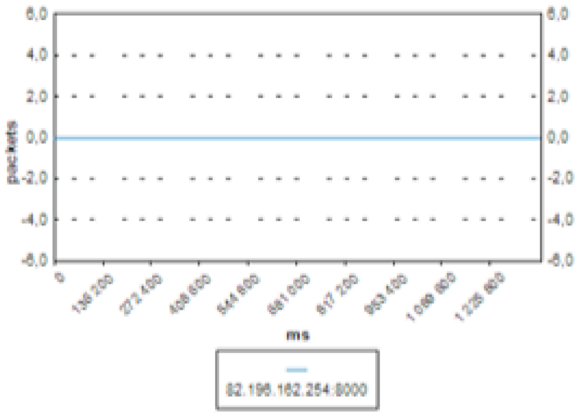
Call Statistics

	Jitter			Packet Loss		
Source	**Max**	**Avg**	**Max Gap**	**Max**	**Avg**	**Total**
82.196.162.254:8000 (G.711 u-law)	13,00	3,00	87,00	0	0,00	0
82.196.163.210:8000 (G.711 u-law)	0,00	0,00	0,00	0	0,00	0

Jitter



Packet Loss



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