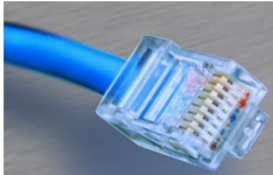




VoIP signalling standards

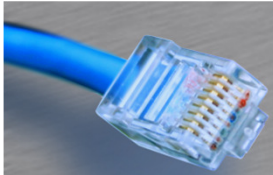
VoIP – Voice over IP





VoIP signalling standards

- A variety of IP based signalling standards to VoIP defined – and used
 - SIP
 - H.323
 - MGCP
 -
- SIP and H.323 included in this course
- The purpose of VoIP signalling
 - Establish, maintain and close voice calls
 - Eventually collect information for taxation



signalling standards

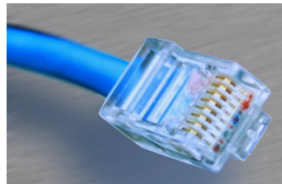
- Two kinds of information necessary to phone
 - Call control (signalling phone to/from exchange)
 - Voice transfer (Between participating phones)
- In traditional telephony
 - Call Control (For example POTS or ISDN)
 - Voice transfer (Circuit switched channel)
- In VoIP – Voice over IP
 - Call control (For example SIP or H.323)
 - Voice transfer (Packet switched)



SIP and H.323

- H.323
 - Defined by ITU in 1996
 - International Telecommunication Union
 - Uses many technologies from PSTN
 - A suite of different protocols incl. voice/video
- SIP – Session Initiation Protocol
 - Defined by IETF in 1996
 - Internet Engineering Task Force
 - Uses many technologies from the Internet
 - Is a signalling protocol for voice/video





SIP SIGNALLING



Session initiation protocol



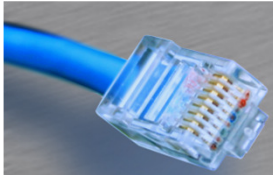
SIP

Session Initiation Protocol



- Defined by IETF in 1996
 - Internet Engineering Task Force
- Uses many technologies from the Internet
- Is a signalling protocol for voice/video
- SIP are widely used as VoIP signalling

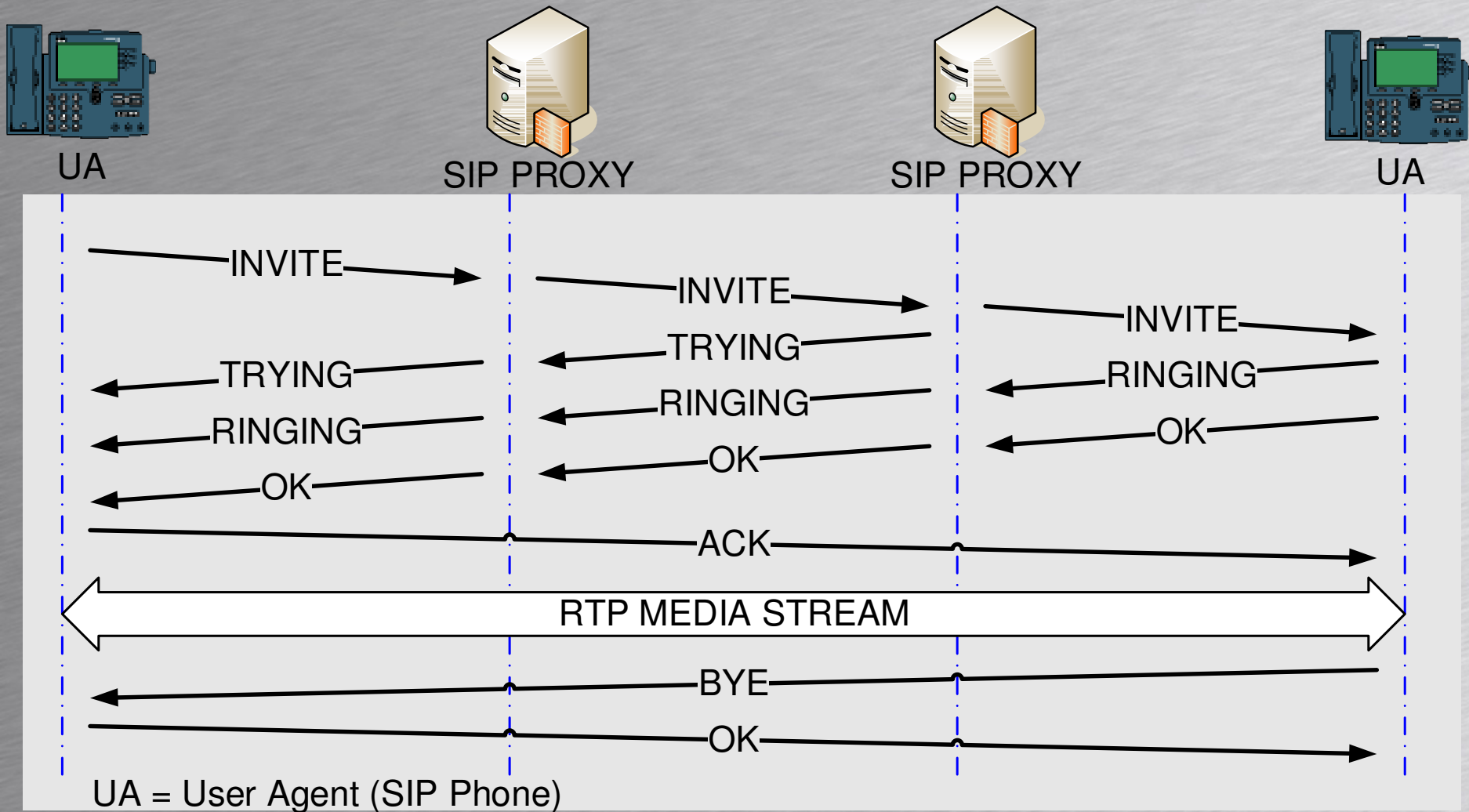


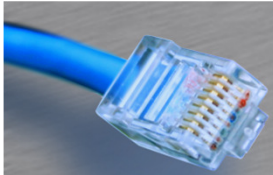


SIP Design

- Expandable
 - Easy integration of new facilities
- (Not) a substitute for PSTN
 - SIP are used by many service providers
- Diversity of end-devices
 - Intelligent end-devices
- Multimedia
 - Voice, video.....

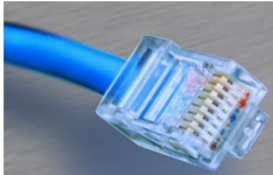
SIP call flow example





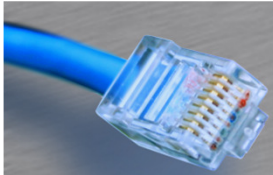
SIP

- Is a application layer protocol
- Is a text based signalling protocol
 - Reminding of the HTTP and SMTP protocols
- Are used to establish, maintain and terminate multimedia sessions
- Support unicast and multicast sessions
- Location independent
 - Suitable for mobile users



SIP components

- There are two SIP components
- User Agents (UA) – SIP End-points (IP Phones) consisting of
 - User Agent Client (UAC): Initiates sessions
 - User Agent Server (UAS): Respond sessions
- SIP servers
 - Proxy server – Like a SIP IP-PBX
 - Registrar server – User Agent register
 - Redirect server – Redirect sessions



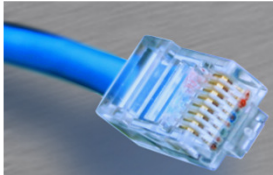
SIP addressing

- SIP addresses are identified by SIP URI's
 - Uniform Resource Identifier
- Example
 - `sip:4103@mercantec.dk;transport=UDP`
 - sip – The URI service
 - 4103 – The URI user part (extension/phone)
 - Mercantec.dk – The URI host part
 - ;transport=UDP – A URI parameter



SIP addressing

- Different formats of URI's
 - FQDN's: sip:heth@mercantec.dk
 - E.164: sip:30539361@mercantec.dk
 - E.164: tel:30539361
 - Mixed: sip:heth@194.123.12.23
- E.164 numbers in DNS



SIP messaging

- Messages contain header describing the communication in details
- Uses a text based syntax and header like HTTP
- Transport protocols TCP, UDP or SCTP
 - SCTP: Stream Control Transport Protocol
- Basically there are two kinds of messages
 - Requests: Sent by Clients
 - Response: Sent by servers



SIP messaging

- Request message header

Method	Request URI	SIP version
--------	-------------	-------------

- Method: INVITE, ACK, BYE
- Request URI: The receiver of the request
- SIP version: Used SIP version (2.0 in use)

- Response message header

SIP version	Status code	Reason phrase
-------------	-------------	---------------

- SIP version: Used SIP version (2.0 in use)
- Status code: An integer describing the answer
- Reason Phrase: A text description of status code



SIP requests methods

Command	Function
INVITE	Used to establish a media session between user agents. Starting a call
ACK	Confirms reliable message exchanges
BYE	Terminates a established session between agents
CANCEL	Terminates a pending request
OPTIONS	Requests information about the capabilities of a caller, without setting up a call
REGISTER	Used by a user agent to register to registrar. The user agent informs the registrar of its IP address and URI.

NOTE: There are defined more SIP request methods than indicated in the above table. The shown methods are from [RFC 3261](#)



SIP response codes

With examples

Respons class	Status code	Reason phrase
Info	1xx	Provisional responses
	100	Trying
	180	Ringning
Succes	2xx	Successful responses
	200	OK
Redirect	3xx	Redirection responses
	302	Moved temporarily
Client error	4xx	Client failure responses
	401	Unauthorized
Server error	5xx	Server failure responses
	503	Service unavailable
Global failure	6xx	Global failure responses
	603	Decline



SIP User Agent – UA

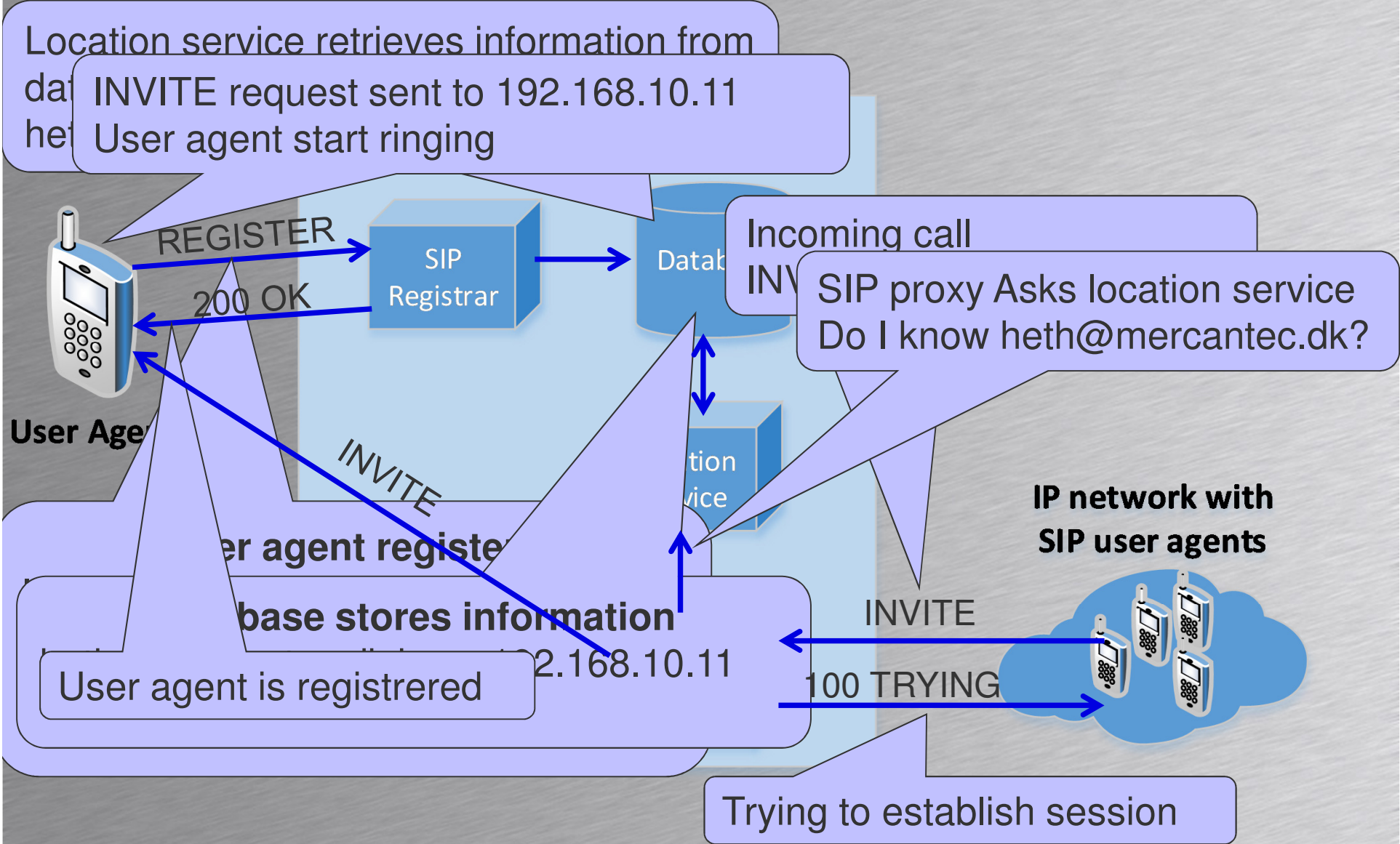
- A User Agent is a software component acting on behalf of a user
- The SIP phone soft or hard is a SIP UA
- User Agent Client – UAC
 - Initiates SIP sessions when calling
 - Send SIP requests
- User Agent Server – UAS
 - Accepts or rejects SIP session requests
 - Receives SIP requests and return SIP response



SIP servers

- Proxy server
 - Receive and sends SIP request and responses on behalf of the User Agents. (UA)
- Registrar server
 - User Agents register to the registrar server
 - Keeps track on User Agents URI and IP addresses and acts as a location service
- Redirection Server
 - Returns “contact this address” responses
- The servers often live on the same “box”

SIP servers



Ascom SIP proxy - sip.ascom.se – receives the invite to Lasses SIP phone and forwards the invite to Lasses SIP phones IP address

heth@mercantec.dk

sip.mercantec.com

sip.ascom.se

lasse@ascom.se



UA

SIP PROXY

SIP PROXY

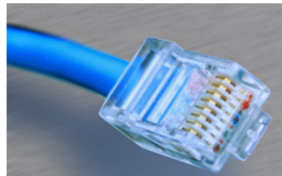
UA

INVITE lasse@ascom.se

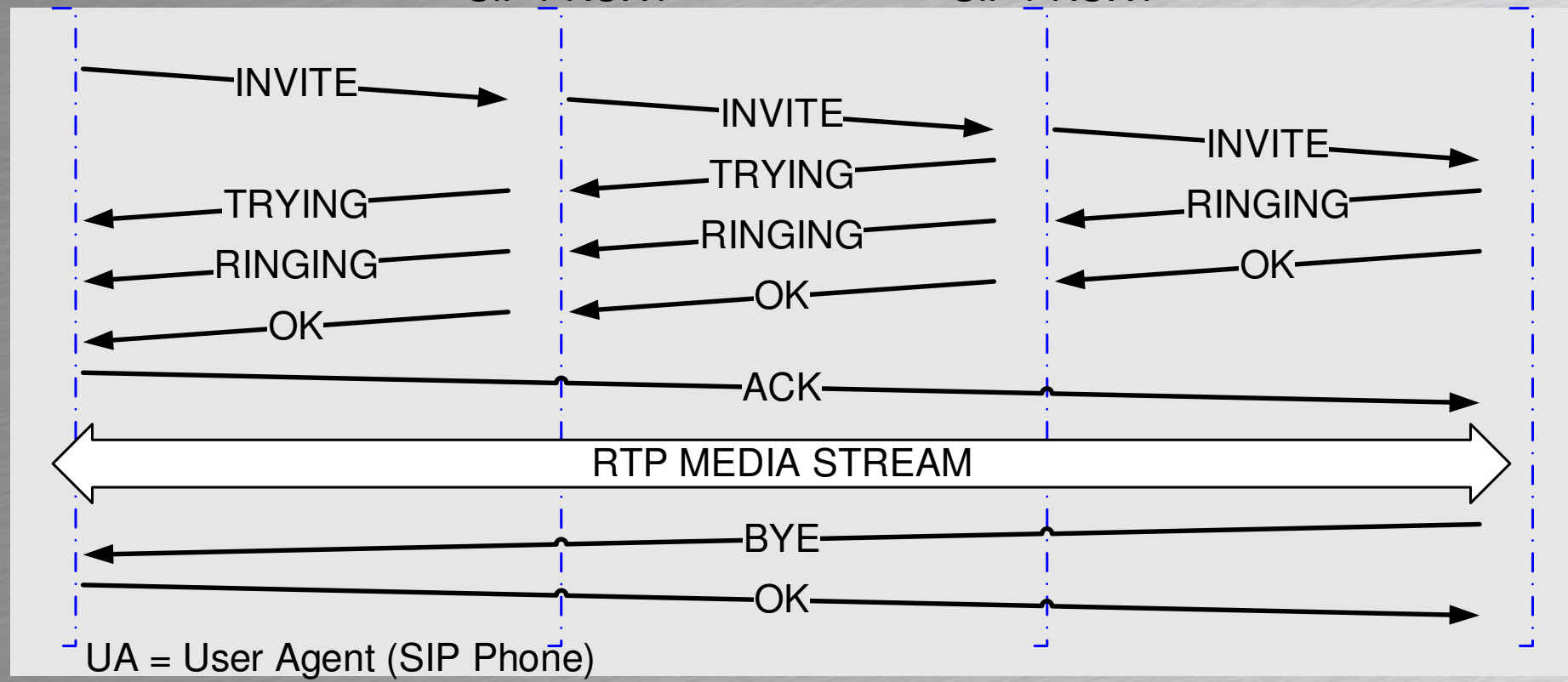
INVITE lasse@ascom.se

INVITE lasse@ascom.se

UA = User Agent (SIP Phone)



SIP Proxy Server



SIP Redirect

heth@mercantec.dk



UA

sip.mercantec.com



SIP PROXY

sip.ascom.se



SIP PROXY

sip.ascom.no



SIP PROXY

per@ascom.no



UA

