KUNGL TEKNISKA HÖGSKOLAN
Institutionen för teleinformatik
CCSlab

2G1305 Internetworking/Internetteknik Winter 2002, Period 3

Module 5: Network Management and VoIP

Lecture notes of G. Q. Maguire Jr.

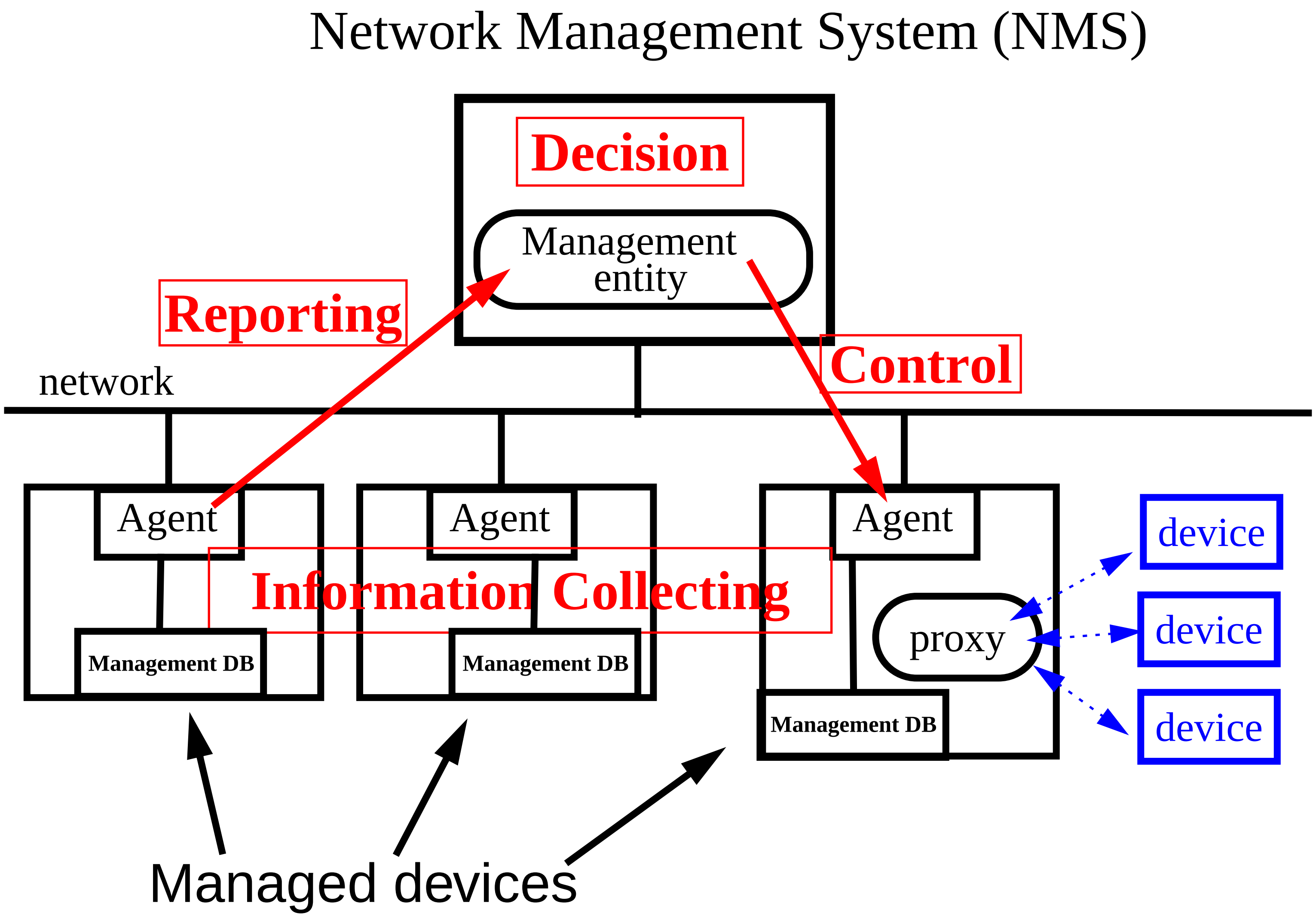
Lecture 5: Outline

- Network Management
- SNMP
- VoIP

ISO FCAPS Network Management Model

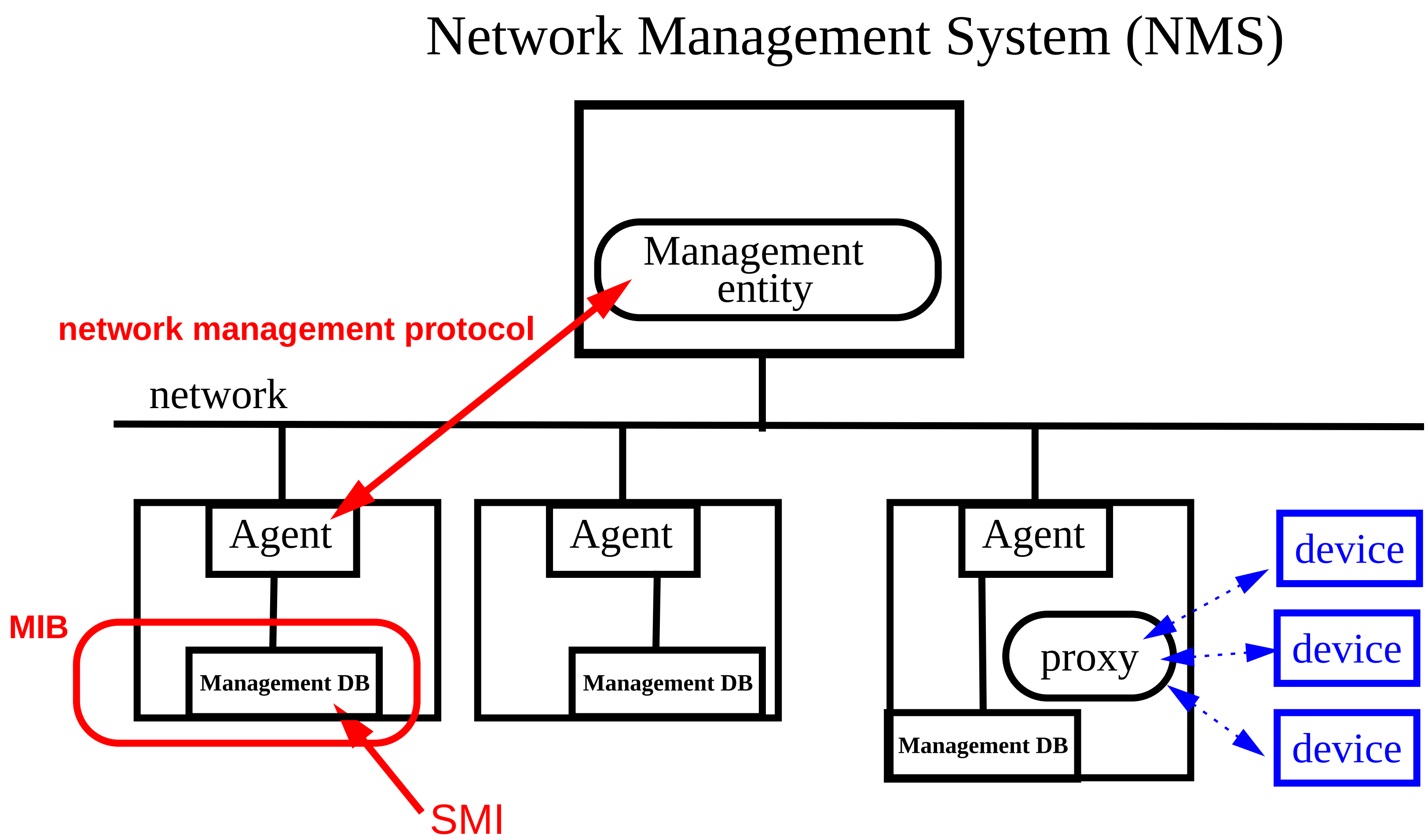
- Fault management
- Configuration management
- Accounting management
- Performance management
- Security management

Network Management Process



Network Management Process

Simple Network Management Protocol (SNMP), Management Information Base (MIB), Sturecture of Management Information (SMI)



SNMP

Version 1

Version 2 - in 1992-1993, the SNMPv2 Working Group developed a security model based on parties to an SNMP transaction - this was known as SNMPv2p. But the working group decided that a user-based security model was much simpler - and hence more likely to be deployed.

December 1995, the SNMPv2 Working Group was deactivated, but two prominent approaches emerged from independent groups:

SNMPv2u	early standardization of the security features and a minimal specification - to encourage rapid deployment of simple agents; deferred standardization of features for managing large networks
SNMPv2*	concurrent standardization of **security** and **scalability** features to ensure that the security design addressed issues of: proxy, trap destinations, discovery, and remote configuration of security Focus was effective management of medium and large networks.

August 1996 a team was formed to recommend a single approach.

SNMPv3

March 1997, the *SNMPv3 Working group* was chartered to define a standard for SNMP security and administration. Target: April 1998 - all SNMPv3 specifications submitted to IESG for consideration as Proposed Standards.

Based on “An Architecture for Describing SNMP Management Frameworks” (*RFC 2271*)

Composed of multiple subsystems:

1. a message processing and control subsystem - Message Processing and Dispatching for SNMP (*RFC 2272*)
2. a security subsystem - based on a User-based Security Model (USM) (*RFC 2274*), provides SNMP message level security (Keyed-MD5 as the authentication protocol and the use of CBC-DES as the privacy protocol - but with support for others) defines a MIB for remotely monitoring/managing the configuration parameters for this Security model
3. a local processing subsystem - responsibile for processing the SNMP PDUs that operate on local instrumentation, applies access control [View-based Access Control Model (VACM) (*RFC 2275*)] and invokes method routines to access management information, and prepares a response to the received SNMP request.
4. SNMPv3 Applications (*RFC 2273*) - includes Proxy Forwarder Applications, which can forward SNMP requests to other SNMP entities, to translate SNMP requests of one version into SNMP requests of another version or into operations of some non-SNMP management protocol; and support aggregated managed objects where the value of one managed object depends upon the values of multiple (remote) items.

SNMP

- SNMPv1
  - only 5 commands: get-request, get-next request, set-request, response
  - Clear-text password
- SNMPv2: 1992-1996
  - get-bulk-request
  - inform-request (for proxy)
  - trap
  - v2 MIB and M2M MIB
  - Authentication
- SNMPv3: 1997-
  - more security enhancement
  - View-based access control - so different managers can see different subset of the information
  - remote configuration

Management Information Base: MIB

MIB is the database of information maintained by the agent that the manager can query or set.

It specifies the data items a managed device must keep, the operations allowed on each item.

See RFC 1213 “Management Information Base for Network Management of TCP/IP-based internets: MIB-II” *http://www.ietf.org/rfc/rfc1213.txt*

See also:

- RFC 2011: SNMPv2 Management Information Base for the Internet Protocol using SMIv2.
- RFC 2012: SNMPv2 Management Information Base for the Transmission Control Protocol using SMIv2.
- RFC 2013: SNMPv2 Management Information Base for the User Datagram Protocol using SMIv2.

Case Diagram

To understand the relationship between counters and to make sure that all the data paths for a packet are accounted for.

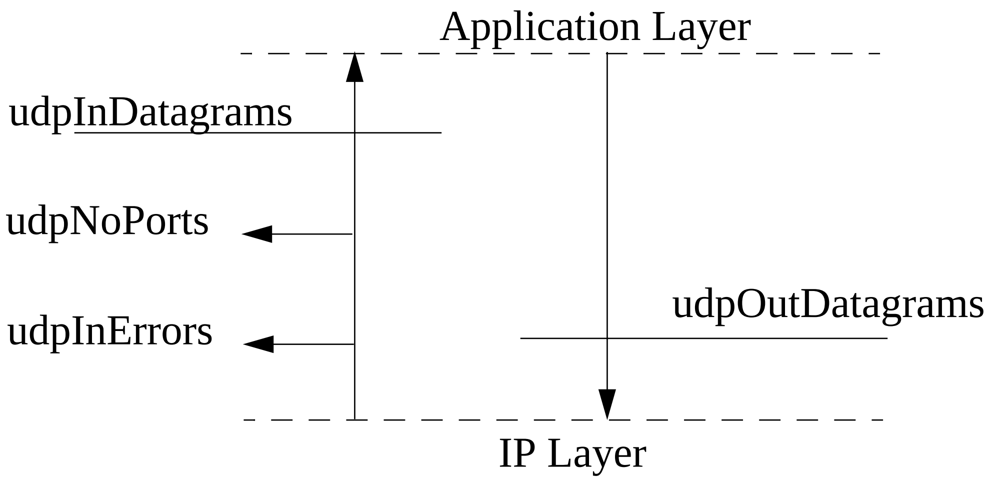


Figure 61: Case diagram of UDP group (W. R. Stevens, TCP/IP Illustrated, V.1, pg. 367)

SNMP Traps

Agent sends a trap to manager to indicate that something has happened.

Following trap types are defined:

0: coldStart

1: warmStart

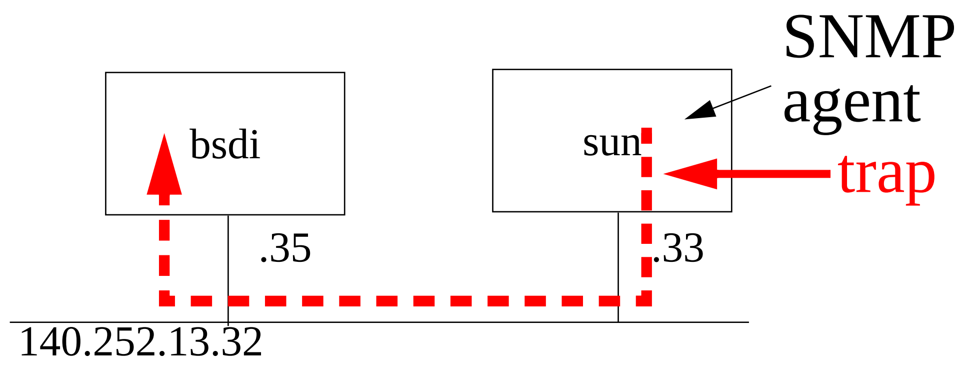
2: linkDown

3: linkUP

4: authenticationFailure

5: egpNeighborLoss

6: enterpriseSpecific



Example: start the SNMP agent on sun and send traps to bsdi; tcpdump output:

trap type
PDU type (length)
Port 162 Port 161

1 0.0 sun.snmp > bsdi.snmp-trap: C=traps Trap(28)
E:unix.1.2.5 [140.252.13.33] coldstart 20

timestamp

2 18.86 (18.86) sun.snmp > bsdi.snmp-trap: C=traps Trap(29)
E:unix.1.2.5 [140.252.13.33] authenticationFailure 1907

Enterprise: sysObjectID
IP address of agent

Remote MONitoring (RMON)

[*RMON MIB 1 (RFC1757)*, *RMON MIB 2 (RFC 2021)*, *RMON MIB Protocol Identifiers (RFC 2074)*, *MIB II (RFC1213)*]

Standard way for users to proactively manage multiple LANs from a central site.

RMON 1

- Notify manager of errors
- provide alerts for network problems
- collects statistical baseline data (i.e., what is “normal” on this LAN), and
- acts as a remote network analyser.

RMON 2

- access higher level protocol information,
- Point-to-point traffic statistics broken down by higher layer protocols,
- eases trouble-shooting, and
- enables network capacity planning [and to solve problems before they become problems].

RMON Probes or Monitors

Network monitoring devices (monitor or probes) are instruments that exist for the purpose of managing a network. Essentially a LAN analyzer - which is always connected to the segment.

- A physical device which is attached to a segment of the network (it will promisciously listen to traffic - to collect statistics and if requested packets)
- Generally a microprocessor based system with 8^{+}MBytes of memory.
- Fairly powerful processors so that events and alarms are not missed.
- In-band or out-of-band communication
  - In-band - you communicate via the probe via the segment it is monitoring
  - Out-of-band - you communicate with it via another path, e.g., a PPP/SLIP/serial connection
- Probes can operate off-line, i.e., they operate even though they may not be in contact with the network managment system.

Probes are sold by lots of vendors.

RMON1 Statistics

Information collected by examing MAC layer

		Group	Description
Tables	1	Statistics	Statistics for the segment to which the RMON probe is attached
	2	History	History (Baselines) of the segment
	4	Host	Per host statistics for each individual transmitting and receiving device.
	5	Host Top N	Top N reports on base statistics.
	6	Matrix	Statistics on all conversations (i.e., who talks to whom)
Packet Capture	7	Filter	Match on any part of a frame, including errors (CRC, overruns, etc.)
	8	Capture	Collect packets, based on filters, for later retrieval (as if you were a network analyzer
SNMP traps	3	Alarm	Alarms to monitor for user-defined events.
	9	Event	Log file for use in conjunction with the Alarm or Filter Group.
Token rings		Token Ring	Ring Station Order, Ring Configuration and Source Routing Information.

Defined by RFC 1757.

Ethernet Statistics Group

"These statistics take the form of free running counters that start from zero when a valid entry is created. Each etherStatsEntry contains statistics for one Ethernet interface. The probe must create one etherStats entry for each monitored Ethernet interface on the device." - from RFC1757

etherStatsTable OBJECT-TYPE

SYNTAX SEQUENCE OF EtherStatsEntry ... ::= { statistics 1 }

etherStatsEntry OBJECT-TYPE

DESCRIPTION

"A collection of statistics kept for a particular Ethernet interface. As an example, an instance of the etherStatsPkts object might be named etherStatsPkts.1"

INDEX { etherStatsIndex } ::= { etherStatsTable 1 }

EtherStatsEntry

```
EtherStatsEntry ::= SEQUENCE {
    etherStatsIndex                 INTEGER (1..65535),
etherStatsDataSource                OBJECT IDENTIFIER,
etherStatsDropEvents                Counter,
etherStatsOctets                    Counter,
etherStatsPkts                      Counter,
etherStatsBroadcastPkts             Counter,
etherStatsMulticastPkts             Counter,
etherStatsCRCAlignErrors            Counter,
etherStatsUndersizePkts             Counter,
etherStatsOversizePkts              Counter,
etherStatsFragments                 Counter,
etherStatsJabbers                   Counter,
etherStatsCollisions                Counter,
etherStatsPkts64Octets              Counter,
etherStatsPkts65to127Octets         Counter,
etherStatsPkts128to255Octets        Counter,
etherStatsPkts256to511Octets        Counter,
etherStatsPkts512to1023Octets       Counter,
etherStatsPkts1024to1518Octets      Counter,
etherStatsOwner                     OwnerString,
etherStatsStatus                    EntryStatus
}
```

EtherHistoryEntry

```
EtherHistoryEntry::= SEQUENCE {
    etherHistoryIndex              INTEGER (1..65535),
    etherHistorySampleIndex        INTEGER (1..2147483647),
    etherHistoryIntervalStart      TimeTicks,
    etherHistoryDropEvents         Counter,
    etherHistoryOctets             Counter,
    etherHistoryPkts               Counter,
    etherHistoryBroadcastPkts      Counter,
    etherHistoryMulticastPkts      Counter,
    etherHistoryCRCAlignErrors     Counter,
    etherHistoryUndersizePkts      Counter,
    etherHistoryOversizePkts       Counter,
    etherHistoryFragments          Counter,
    etherHistoryJabbers            Counter,
    etherHistoryCollisions         Counter,
    etherHistoryUtilization        INTEGER (0..10000)
}
```

HostEntry

```
HostEntry ::= SEQUENCE {
    hostAddress            OCTET STRING,
    hostCreationOrder      INTEGER (1..65535),
    hostIndex              INTEGER (1..65535),
    hostInPkts             Counter,
    hostOutPkts            Counter,
    hostInOctets           Counter,
    hostOutOctets          Counter,
    hostOutErrors          Counter,
    hostOutBroadcastPkts   Counter,
    hostOutMulticastPkts   Counter
    }
```

Host Top N group

Used to prepare reports that describe the hosts that top a list **ordered** by one of their statistics.

hostTopNControlTable is used to initiate the generation of such a report, the management station selects the parameters, such as:

- which interface,
- which statistic,
- how many hosts, and
- the start and stop times of the sampling.

The Matrix Group

Matrix group consists of the matrixControlTable, matrixSDTable and the matrixDSTable.

These tables store statistics for a particular conversation between two addresses.

The maxtrixSDTable - contains a entries indexed by source and destination.

```
MatrixSDEntry ::= SEQUENCE {
    matrixSDSourceAddress   OCTET STRING,
    matrixSDDestAddress     OCTET STRING,
    matrixSDIndex           INTEGER (1..65535),
    matrixSDPkts            Counter,
    matrixSDOctets          Counter,
    matrixSDErrors          Counter
}
```

The maxtrixSDTable - a similar set of statistics (MatrixDSEntry) indexed by destination and source.

RMON2

Information collected from network and higher layer ("application") headers (defined by *RFC2021*)

		Group	Description
Protocols	11	Protocol Directory	List of protocol types the probe is capable of monitoring
	12	Protocol Distribution	Number of packets and octets by protocols on a network segment
Network layer	13	Address Mapping	MAC addresses and corresponding network addresses
	14	Network Layer Host	Amount of traffic sent to and from each network address
	15	Network Layer Matrix	Amount of traffic between each pair of network addresses
		Network Layer Matrix Top N	Top N conversations over a user-defined period (packet or octet counts)
Higher layers	16	Application Layer Host	Amount of traffic, by protocol
	17	Application Layer Matrix	Amount of traffic, by Protocol, between each pair of network addresses.
		Application Layer Matrix Top N	Top N conversations over a user-defined period (packet or octet counts)

Information collected from network and higher layer ("application") headers (defined by *RFC2021*)

		Group	Description
	18	User History	Users created custom History Tables based on supported OID's.
Probe itself	19	Probe Configuration	Configuration of various operating parameters of the probe
	20	RMON Conformance	Lists which groups and instances of a group a probe supports

Proprietary MIBs to extend RMON functions

ION Network, Inc. adds:

Group	Description
FDDI	FDDI MAC level and User Data Statistics for FDDI networks
Protocol	Bandwidth utilization by protocols
SolCom Host	Tracks MAC to IP address mappings; including when a host was first and last seen, when a new host appears on the segment
Traffic Generation	Generate traffic using user-defined packets (including packet with errors)
Response Time Monitoring	Works out response times and helps to pin-point WAN failures using ICMP echo-requests initiated from the central site.

Network Management Systems

- HP OpenView -- *http://www.managementsoftware.hp.com/marketsegments/enterprises/index.asp*
  - Derived from OpenView: IBM NetView, Digital Polycenter NetView, and NCR OneVision
- SunSoft Solstice: Site Manager, Solstice Domain Manager, and Enterprise Manager -- *http://www.sun.com/solstice/index.html*
- Aprisma Management Technologies' Spectrum *http://www.aprisma.com/*

WEB based Management

Using the Web as an interface

- Web based Reporting/Statistics
  - Netscount *http://www.netscout.com/*
  - HP*Netmetrix* WebReporter
  - Network Statistics Collection And Reporting Faclility (Netscarf *http://www.merit.edu/internet/net-research/netscarf/index.html*), their Scion package consists of five components:
    - scollect - collects network data from a set of routers
    - scook - preprocesses network data into a more convenient (condensed) form
    - scserver - delivers the network data in response to client requests
    - sclient - requests network data from the scserver on behalf of a reporting or graphing application
    - Real-Time Data (rtdata) tree - a flat-file database: stores the data collected by scollect
  - Merit Internet Performance and Analysis Project (*IPMA*), tools: NetNow, AS Explorer, Route Flap, Routing Table Statistics Generator, …
    - See also pointers to tools developed by *others*.
- Web based Interfaced Management Platforms
  - *OpenView World Wide Web Interface*
  - *DR-Web Manager* and Agent
  - *SiteScope* v2.2 - a Java-based Web Site Monitoring and Administration Software

- Web based Interfaced Management Tools
  - Cisco *ClickStart* - for configuring a Router with a Web Browser
  - Axis Communications AB's *Thin Server*
- Management of Web Services
  - Harrie Hazewinkel, Carl W. Kalbfleisch, Juergen Schoenwaelder, "Definitions of Managed Objects for WWW Services", November 11, 1997 *http://ietf.org/ids.by.wg/applmib.html*
    - Service Information Group
    - Protocol Statistics Group
    - Document Statistics Group

Web Based Enterprise Management Initiative (WBEM)

see *http://www.dmtf.org/wbem*

Goal: to consolidate and unify the data provided by **existing** management technologies - in order to solve enterprise problems; i.e., from the application layer problem report down to the interface card - even if the card is in a remote branch office.

Builds on:Intel's Wired for Management (WfM) effort ==> *Distributed Management Task Force* (formerly Desktop Management Task Force) and Desktop Management Interface (now DMI 2.0)

The DMI was designed to be:

- "independent of a specific computer or operating system
- independent of a specific management protocol
- easy for vendors to adopt
- usable locally -- no network required
- usable remotely using DCE/RPC, ONC/RPC, or TI/RPC
- mappable to existing management protocols (e.g., CMIP, SNMP)
- The DMI procedural interfaces are specifically designed to be remotely accessible through the use of Remote Procedure Calls. The RPCs supported by the DMI include: DCE/RPC, ONC/RPC, and TI/RPC." -- DMI 2.0 Introduction

DMI 2.0 has three groups:

- ComponentID group - required for all DMI components, includes information such as the six named attributes: "Manufacturer", "Product", “Version”, "Serial Number", "Installation", and "Verify" [asking for this last group causes the device to check itself].
- Event Groups
  - includes a template group used to describe the format of event data for standard events
  - Event State group is defined to hold the current state of state-based events
  - Events can be of different severity levels: Monitor, Information, OK, Non-Critical, Critical, and Non-Recoverable.
- DMI Service Provider Groups - provides the means for those interested in specific events to subscript to just the events that they want; subscribers can say how they want to be notified (DCE RPC, TI RPC, ONC RPC), what transport protocol should be used (TCP/IP, IPX, …), when the no longer want to be notified (Subscription Expiration DateStamp), …

Four Elements of DMI

- a format for describing management information - Management Information Format (MIF)
  - a language for describing each component;
  - each component has a MIF file to describe its manageable characteristics; and
  - When a component is initially installed into the system, the MIF is added to the (implementation-dependent) MIF database.
- a service provider entity
- two sets of APIs, one set for service providers and management applications to interact (Service Provider API for Components), and the other for service providers and components to interact (Component Provider API), and
- set of services for facilitating remote communication.

Common Information Model (CIM)

- DMTF Common Information Model (CIM) *http://www.dmtf.org/standards/standard_wbem.php* based on object-oriented technologies for use in Web-based management
- XML Mapping Specification v2.0.0
- XML Document Type Definition v2.0.0
- CIM Operations over HTTP, V1.0

Java and Management

Java Management API (JMAPI)- Set of extensible objects and methods, defines an application programming interfaces (API) which includes:

- JavaManagement API User Interface Style Guide
- Admin View Module (AVM)
- Base Object Interfaces
- Managed Container Interfaces
- Managed Notification Interfaces
- Managed Data Interfaces
- Managed Protocol Interfaces
- SNMP Interfaces
- Applet Integration Interfaces

Java Dynamic Management Kit - A Java agent toolkit for rapid development of autonomous Java agents for system, application, or network devices.

Inter-domain Management task force (XoJIDM)

Sponsored by X/Open and the Network Management Forum (NMF), see

Inter-Domain Management, Open Group Technical Standard, C802 ISBN 1-85912-256-6 January 2000 524 pages.

They have specificed such things as SNMP MIBS to CORBA-IDL conversion, CORBA-IDL to GDMO/ASN.1 conversion, CORBA/SNMP Gateway, … .

Policy Based Management

See the recent exjobb report “Implementing policy-based network management” by Yavor Adel Al-Shaikhly

Check Point’s <u>*http://www.checkpoint.com/products/management/index.html*</u> policy management framework

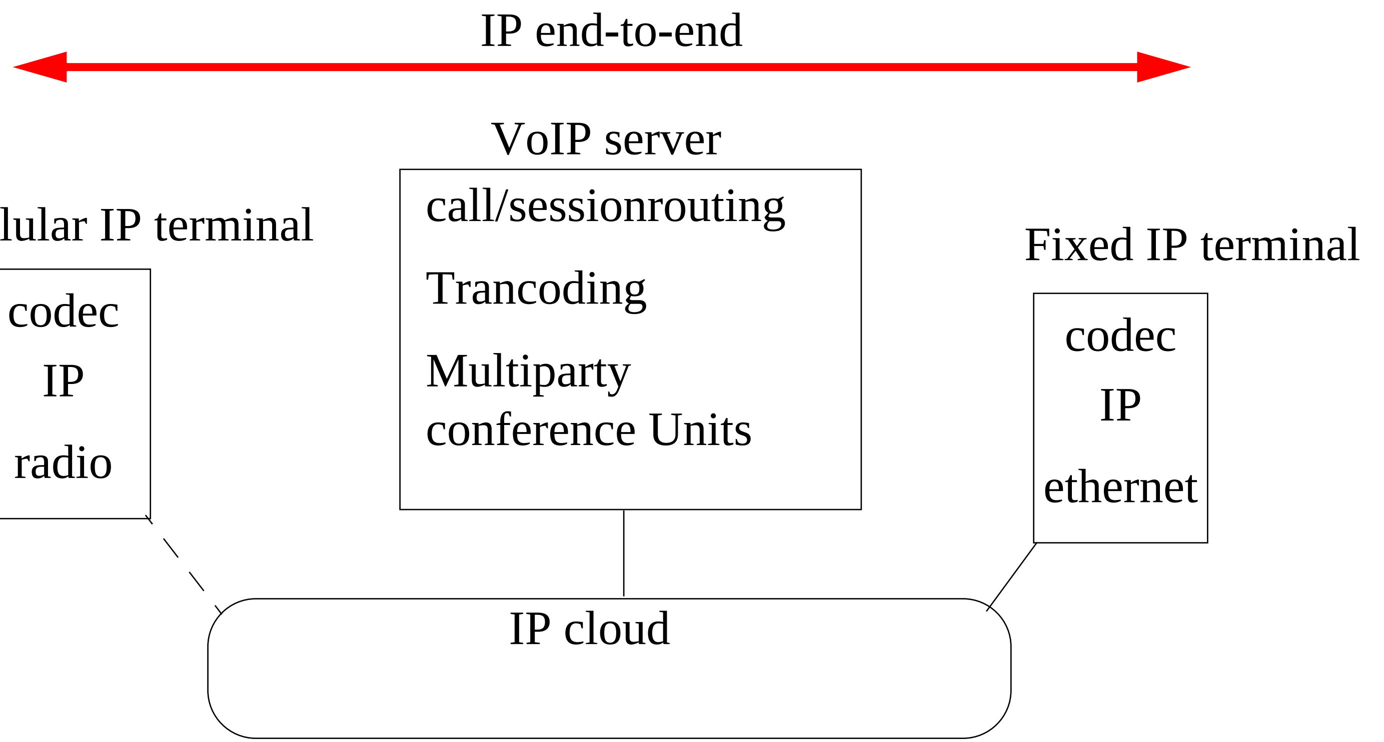
“<u>*Policy Agents: Licensed to Manage/Policy Based Management of Distributed Systems*</u>” by Morris Sloman. Department of Computing, Imperial College, London, U.K.

Cisco’s “<u>*CiscoAssure Policy Networking: Enabling Business Applications through Intelligent Networking*</u>”

Voice over IP (VoIP)

First we will set the context and then we will examine the technical details.

VoIP End-to-End Architecture



Deregulation ⇒ New regulations

- US Telecommunications Act of 1996[1]
  - “The goal of this new law is to let anyone enter any communications business -- to let any communications business compete in any market against any other.”[2]
  - updated the Communications Act of 1934
- New interconnection points
  - perhaps there is something that LECs can do with all the empty space in their central exchanges [which appeared due to the shrinking size of their own switching equipment]
- Number portability - even local numbers
  - every call results in ~10 DB lookups
- “Universal Service”
  - from a myth to a legal requirement
  - an evolving service level - not a fixed service or service level!
  - special subsidies for schools, health care, libraries, etc.
- February 1997 World Trade Organization (WTO) agreement[3]

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1. The official citation for the new Act is: Telecommunications Act of 1996, Pub. LA. No. 104-104, 110 Stat. 56 (1996).

2. http://www.fcc.gov:80/telecom.html

3. For informal background see “WTO negotiations on basic Telecommunications” - http://www.wto.org/wto/services/tel.htm

Deregulation ⇒ New operators

Lots of new actors as operators:

- `WorldCom` - from a local player to a global player in ~5 years
- WinStar - wireless bypass to >70% of the US propulation
- Qwest - Internet telephony
- Level3 - Internet telephony
- Delta Three-Internet telephony, subsidiary of RSL Communications Ltd.
- Net2Phone - internet telephony
- ITXC and GRIC - concentrating on interconnecting ISP and selling minutes of voice
- …

Deregulation ⇒ New Suppliers

Lots of new actors as equipment suppliers:

- Cisco, 3Com, Nortel Networks, …

Traditional telecom equipment vendors buying datacom vendors:

- Lucent buys Prominet,
- Ericsson buys ACC,
- Alcatel buys DSC Communications and Packet Engines,
- Nortel + Bay Networks becomes Nortel Networks: unified networks

Lots of mergers and acquisitions among datacom vendors.

Latency

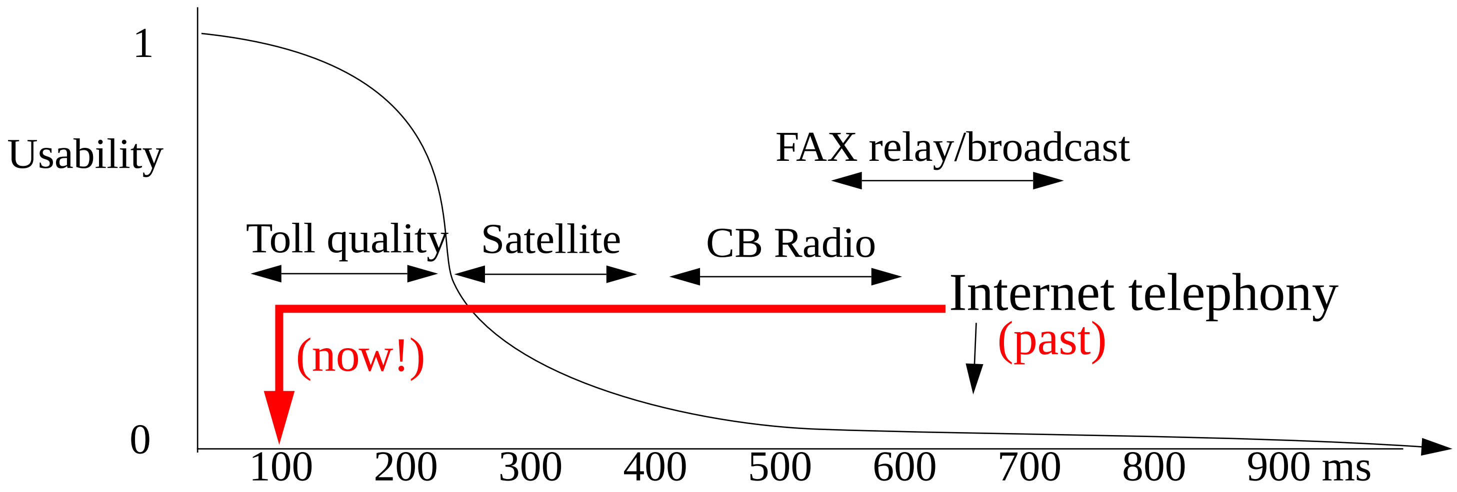


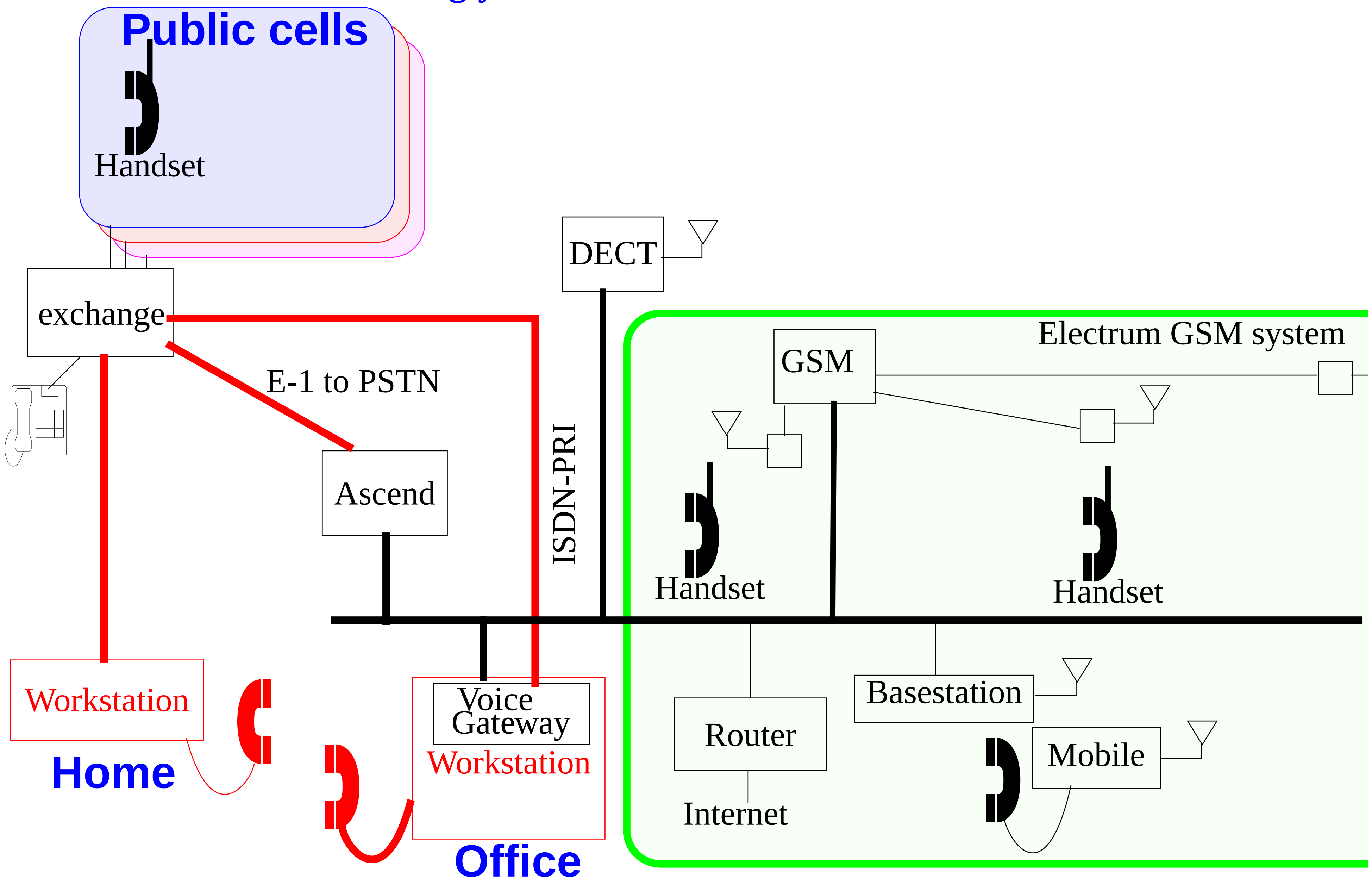
Figure 62: Usability of a voice circuit as a function of end-to-end delay (adapted from a drawing by Cisco)[a]

a. http://www.packeteer.com/solutions/voip/sld006.htm

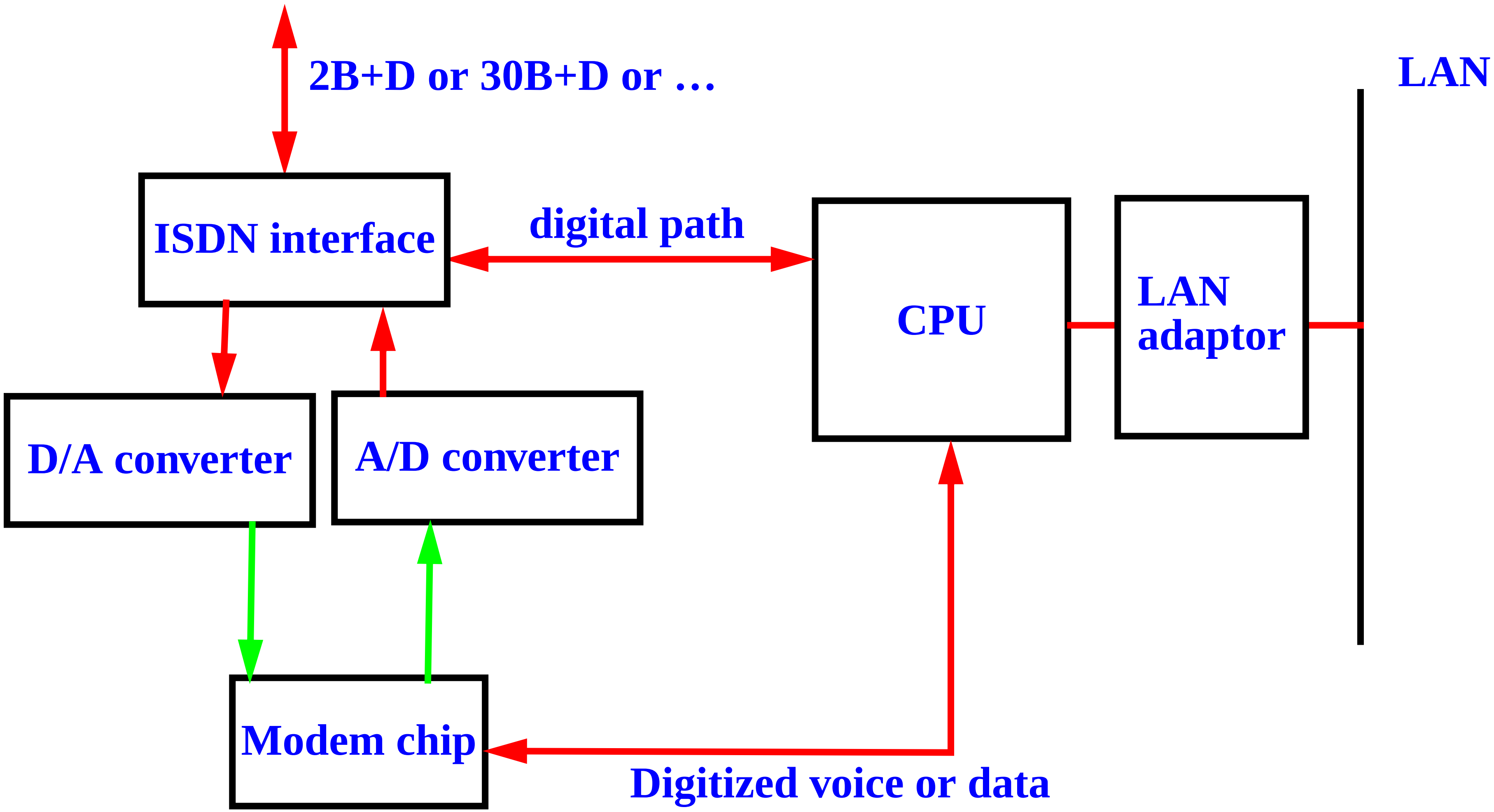
However:

Round-trip	min (ms)	avg (ms)	max (ms)	hops
Local LAN	1	1	3	0
to northern Sweden (basil.cdt.luth.se)	21	25	41	8
to Austria (freebee.tu-graz.ac.at)	73	109	353	18
To server in US network	131	306	526	19
To my machine in the US (~30 ms is the ISDN link)	175	328	600	21
To KTH's subnet at Stanford University in the US (ssvl.stanford.edu)	166	170	217	20

Increasingly IP based data+voice infrastructure



Voice Gateway



Use access servers such as Ascend Communications MAX, filled with digital modems (currently used for current analog modem pools) as voice gateways [see Ascend's MultiVoice Application for the MAX]

Voice over IP (VOIP)

Gateways not only provide basic telephony and fax services but can also will enable lots of value-added services, e.g., call-centers, integrated messaging, least-cost routing, … .

Such gateways provide three basic functions:

- Interface between the PSTN network and the Internet

  Terminate incoming synchronous voice calls, compress the voice, encapsulate it into packets, and send it as IP packets. Incoming IP voice packets are unpacked, decompressed, buffered, and then sent out as synchronous voice to the PSTN connection.

- Global directory mapping

  Translate between the names and IP addresses of the Internet world and the E.164 telephone numbering scheme of the PSTN network.

- Authentication and billing

Voice representation

ITU G.723.1 algorithm for voice encoding/decoding or G.729 (CS-ACELP voice compression).

Signaling

Based on the H.323 standard on the LAN and conventional signaling will be used on telephone networks.

Fax Support

Both store-and-forward and real-time fax modes - with store-and-forward the system records the entire FAX before transmission.

Management

Full SNMP management capabilities via MIBs (Management Information Base) will be provided to control all functions of the Gateway. Extensive statistical data will be collected on dropped calls, lost/resent packets, and network delays.

Compatibility

De jure standards:

- ITU G 723.1/G.729 and H.323
- VoIP Forum IA 1.0

De facto standards:

- Netscape’s Cooltalk
- Microsoft’s NetMeeting

A protocol to keep you eyes on: Session Initiation Protocol (SIP) [RFC 2543], much simpler than H.323

VOIP Modes of Operation

- PC to PC
- PC-to-Telephone calls
- Telephone-to-PC calls
- Telephone-to-Telephone calls via the Internet
- Premises to Premises
  - use IP to tunnel from one PBX/Exchange to another
- Premises to Network
  - use IP to tunnel from one PBX/Exchange to a gateway of an operator

Commercial VOIP systems

Ericsson's Internet Telephony site

- IP telephony
- Phone Doubler, Voice Gateway, Gatekeeper, …

VocalTec Communications Ltd.

http://www.vocaltec.com/

- Surf&Call: a Web browser plug-in enables online customers to connect directly from a website to a live sales or support agent on a regular telephone.

NMS Communications (formerly ViaDSP, Inc.)

- http://www.viadsp.com/
- PacketTel Gateway - a carrier class gateway with real-time voice support, ITU G.723.1, G.729a; Hybrid echo cancellation, Silence suppression

3Com

- licensed an H.323 toolkit from DataBeam Corp + Total Control™ HiPer™ Access System remote access device ⇒ voice gateway.
- Note that they simply download different software in the DSPs which are normally acting like a 56Kbps modem.
- joins with Siemens to form a joint venture company (US$100M) to do internet and LAN telephony

Packeteer's Packet Shaper

- *http://www.packeteer.com/*
- prioritizes traffic and shapes its delay distribution

Lucent Technologies

- *PathStar* - integrated Voice and IP switch
- *MultiVoice*

Cisco Voice Over IP

Enables Cisco 3600 series routers to carry live voice traffic (e.g., telephone calls and faxes) over an IP network.

They state that this could be used for:

- "Toll bypass
- Remote PBX presence over WANs
- Unified voice/data trunking
- POTS-Internet telephony gateways"

Uses Real-Time Transport Protocol (RTP) for carrying packetized audio and video traffic over an IP network.

Cisco 3600 supports a selection of CODECs:

- G.711 A-Law 64,000 bits per second (bps)
- G.711 u-Law 64,000 bps
- G.729 8000 bps

Cisco 3800 supports even more CODECs:

- ITU G.726 standard, 32k rate
- ITU G.726 standard, 24k rate
- ITU G.726 standard, 16k rate
- ITU G.728 standard, 16k rate (default)
- ITU G.729 standard, 8k rate

By using Voice Activity Detection (VAD) - you only need to send traffic if there is something to send.

An interesting aspect is that user's worry when they hear absolute silence, so to help make them comfortable it is useful to play noise when there is nothing to output. Cisco provide a "**comfort-noise** command to generate background noise to fill silent gaps during calls if VAD is activated".

Cisco 3600 series router can be used as the voice gateway with software such as Microsoft NetMeeting.

Cisco 3800 also supports "fax-relay" - at various rates either current voice rate or

2,400/4,800/7,200/9,600/14,400 bps bps fax rates.

For futher information see

http://www.cisco.com/univercd/cc/td/doc/product/software/ios113ed/113t/113t_1/voip/config.htm

Intranet Telephone System

On January 19, 1998, *Symbol Technologies* and Cisco Systems announced that they had combined the Symbol Technologies' NetVision™ wireless LAN handset and Cisco 3600 to provide a complete wireless local area network telephone system based on Voice-Over-IP technology. (*White Paper*)

The handset use wireless LAN (IEEE 802.11) infrastructure and a voice gateway via Cisco 3600 voice/ fax modules. The system conforms to H.323.

> "I believe that this is the first wireless local area network telephone based on this technology" -- Jeff Pulver

Seamless roaming via Symbol's pre-emptive roaming algorithm with load balancing.

Claim each cell can accommodate ~25 simultaneous, full-duplex phone calls.

Current Ericsson is a partner with Symbol, using Ericsson's *WebSwitch2000*

Wireless LANs

“The wireless workplace will soon be upon us[1]

Telia has strengthened its position within the area of radio-based data solutions through the acquisition of Global Cast Internetworking. The company will primarily enhance Telia Mobile’s offering in wireless LANs and develop solutions that will lead to the introduction of the wireless office. A number of different alternatives to fixed data connections are currently under development and, *later wireless IP telephony will also be introduced.*

…

The acquisition means that Telia Mobile has secured the resources it needs to maintain its continued expansion and product development within the field of radio-based LAN solutions. *Radio LANs are particularly suitable for use by small and medium-sized companies as well as by operators of public buildings such as airports and railway stations.*

Today’s radio-LAN technology is based on *inexpensive products that do not require frequency certification*. They are *easy to install* and are often used to replace cabled data networks in, for example, large buildings.

…”

[*emphasis* added by Maguire]

1. Telia press annoucement: 1999-01-25

Telia’s HomeRun

http://www.homerun.telia.com/

A subscription based service to link you to your corporate network from airports, train stations, ferry terminals, hotels, conference centers, etc.

Look for Telia’s HomeRun logo:

Ericsson’s "GSM on the Net"

- Provide communication services over an integrated GSM- IP (Internet Protocol) network
- support local and global mobility
- support multimedia capabilities and IP-based applications
- uses small radio base stations to add local-area GSM coverage to office LANs
- provides computer-telephony integration: applications include web-initiated telephony, directory-assisted dialing, unified messaging and advanced conferencing and application-sharing using voice datacoms and video.

Carriers offering VOIP

"Equant, a network services provider, will announce tomorrow that it is introducing voice-over-frame relay service in 40 countries, ...

The company says customers can save 20% to 40% or more by sending voice traffic over its frame relay network. "This is the nearest you're going to get to free voice," says Laurence Huntley, executive VP of marketing for Equant Network Service.

The Equant service uses the Cisco Systems 3810 router, which takes the customer's voice and data traffic and integrates them before putting the traffic on the Equant network. Equant is also working with Cisco to introduce a voice-over-IP service. …
Equant isn't alone in its pursuit to send voice traffic over data networks. Most of the major carriers are testing services that would send voice over data networks. ... ."[1]

AT&T VoIP phone: *http://www.telephones.att.com/new_prod.html*

Deutsche Telekom running a pilot Internet telephony service using networking products from Ascend Communications and VocalTec.

1. Mary E. Thyfault, Equant To Roll Out Voice-Over-Frame Relay Service, InformationWeek Daily, 10/21/98.

Gatekeeper

To control an H.323 VOIP network Ericsson has introduced a produce called H.323 Gatekeeper. It provides for control of:

- How much traffic is allocated to voice, video, and data;
- Do network bandwidth management;
- Handle routing when there are multiple H.323 Gateways;
- Manage Network Subscriber Access;
- Provides for Charging/Billing Systems;
- Add new Services & Applications;
- Support Network Security and Subscriber Authentication

Gatekeeper uses RAS (Registration, Admission, and Status) for call signalling and its communication.

VOIP vs. traditional telephony

In "*Telcos Hear New Voices*" by Margrit Sessions, Phillips Tarifica Ltd., she predicts that by 2001, Internet telephony could squeeze nearly US$1.2 billion in revenue out of 16 international service providers, while losses due to e-mail (US$463 million) and Internet fax (US$170 million) will be much less.

Expected loss of international call revenue due to: Internet phone, fax, and e-mail, by operator:

Company	**Expected Losses (millions of US Dollars)**	**Loss as a percentage of revenue**
AT&T	~350	3.6%
Kokusai Denshin Denwa (KDD) Co. Ltd. (Japan)	~307	10.4%
Deutsche Telekom	~175	4.2%
Telstra Corp. (Australia)	~168	9%
Embratel (Brazil)	~28	11.5%
Bezeq (Israel)	~30	10.7%

Economics

"Can Carriers Make Money On IP Telephony? by Bart Stuck and Michael Weingarten, Business Communication Review, Volume 28, Number 8,August 1998, pp. 39-44 - `http://www.bcr.com/bcrmag/08/98p39.htm`

"What is the reality in the battle over packet-versus-circuit telephony, and what is hype?

Looking at the potential savings by cost element, it is clear that in 1998, access arbitrage is the major economic driver behind VOIP. By 2003, we anticipate that switched-access arbitrage will diminish in importance, as the ESP exemption disappears and/or access rates drop to true underlying cost.

However, we believe that the convergence between voice and data via packetized networks will offset the disappearance of a gap in switched access costs. As a result, VOIP will continue to enjoy a substantial advantage over circuit-switched voice. Indeed, as voice/data convergence occurs, we see standalone circuit-switched voice becoming economically nonviable."

VoIP Handsets

In addition to the WLAN VoIP handset, there are now starting to appear USB attached VoIP handsets:

- TigerJet Network *http://www.tjnet.com/solutions/usb_handset.htm*

Coming are VoIP cellular handsets

Conferences

VON (Voice on the Net)

Patents

Mixing voice and data in the LAN goes back to at least this patent:

4581735 : Local area network packet protocol for combined voice and data transmission

INVENTORS:

Flamm; Lois E., Chatham Township, Morris County, NJ

Limb; John O., Berkeley Heights, NJ

ASSIGNEES: AT&T Bell Laboratories, Murray Hill, NJ

ISSUED: Apr. 8 , 1986

FILED: May 31, 1983

ABSTRACT: In order to control the transfer of packets of information among a plurality of stations, the instant communications system, station and protocol contemplate first and second oppositely directed signal paths. At least two stations are coupled to both the first and the second signal paths. A station reads one signal from a path and writes another signal on the path. The one signal is read by an arrangement which electrically precedes the arrangement for writing the other signal. Packets are transmitted in a regular, cyclic sequence. A head station on a forward path writes a start cycle code for enabling each station to transmit one or more packets. If a station has a packet to transmit, it can read the bus field of a packet on the forward path. Responsive thereto, a logical interpretation may be made as to whether the forward path is busy or is not busy. If the path is not busy, the packet may be written on the path by overwriting any signal thereon including the busy field. If the path is busy, the station may defer the writing until the path is detected as not busy. In order to accommodate different types of traffic, the head station may write different start cycle codes. For example, a start-of-voice code may enable stations to transmit voice packets; a start-of-data code may enable stations to transmit data packets, etc. for the different types of traffic. Further, the start cycle codes may be written in a regular, e.g., periodic, fashion to mitigate deleterious effects, such as speech clipping. Still further, the last station on the forward path may write end cycle codes in packets on a reverse path for communicating control information to the head station. Responsive to the control information, the head station may modify the cycle to permit the respective stations to, for example, transmit more than one packet per cycle or to vary the number of packet time slots, which are allocated to each of the different types of traffic.

Deregulation ⇒ Trends

- replacing multiplexors with Routers/Switches/… << 1/10 circuit swi. cost
- Standard telco interfaces being replaced by datacom interfaces
- New Alliances:
  - HP/AT&T Alliance - a specific application: electronic commerce
  - 3Com/Siemens, Bay/Ericcson, Cabletron/Nortel, Alcatel integrating Cisco IOS software technology, Ericsson Radio Systems & Cisco Systems collaborate wireless Internet services
- future developments building on VOIP
  - ♦ Fax broadcast, Improved quality of service, Multipoint audio bridging, Text-to-speech conversion and Speech-to-Text conversion, Voice response systems, …
  - ♦ Replacing the wireless voice network's infrastructure with IP:
    U. C. Berkeley's ICEBERG: Internet-based core for CEllular networks BEyond the thiRd Generation

See the Univ. of California at Berkeley ICEBERG project report:

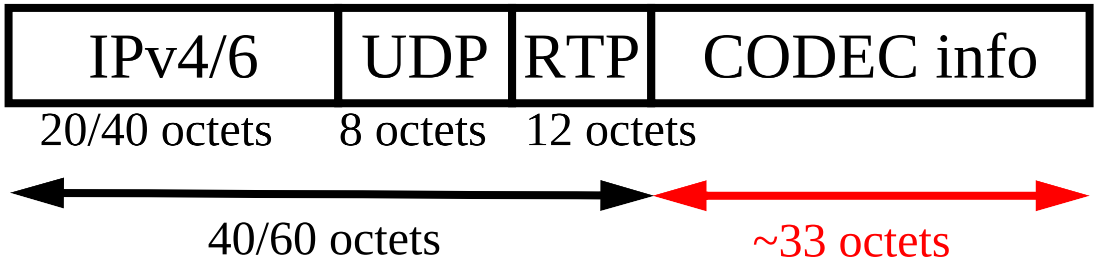
http://iceberg.cs.berkeley.edu/release/

⇒Telecom (only) operators have no future

⇒Telecom (only) companies have no future

VoIP details

Carry the speech frame inside an RTP packet



Typical packetisation time of 10-20ms per audio frame.

See *http://www.ietf.org/ids.by.wg/avt.html*

RTP: Real-Time Transport Protocol

Designed to carry out variety of real-time data: audio and video.

Provides two key facilities:

- Sequence number for order of delivery
- Timestamp for control playback

Provides no mechanisms to ensure timely delivery.

0 1	2	3	4	8	9	16
VER	P	X	CC	M	PTYPE	Sequence number
Timestamp						
Synchronization source identifier						
Contributing source ID …						

- P - whether zero padding follows the payload.
- X - whether extension or not.
- M - marker for beginning of each frame.
- PTYPE - Type of payload.

RTP and H.323 for IP Telephony

<table>
<tr><td colspan="2">audio/video applications</td><td colspan="4">signaling and control</td><td>data applications</td></tr>
<tr><td>video code</td><td>audio codec</td><td rowspan="2">RTCP</td><td rowspan="2">H.225 registration</td><td rowspan="2">H.225 Signaling</td><td rowspan="2">H.245 Control</td><td rowspan="2">T.120</td></tr>
<tr><td colspan="2">RTP</td></tr>
<tr><td colspan="4">UDP</td><td colspan="3">TCP</td></tr>
<tr><td colspan="7">IP</td></tr>
</table>

- H.323 is the framework of a group protocols for IP telephony (from ITU)
- H.225 - Signaling used to establish a call
- H.245 - Control and feedback during the call
- T.120 - Exchange of data associated with a call
- RTP - Real-time data transfer
- RTCP - Real-time Control Protocol

SIP: Session Initiation Protocol

SIP is an alternative to H.323 proposed by IETF. Only covers signaling (parts of H.323). Does not use RTP (but sessions can use RTP)

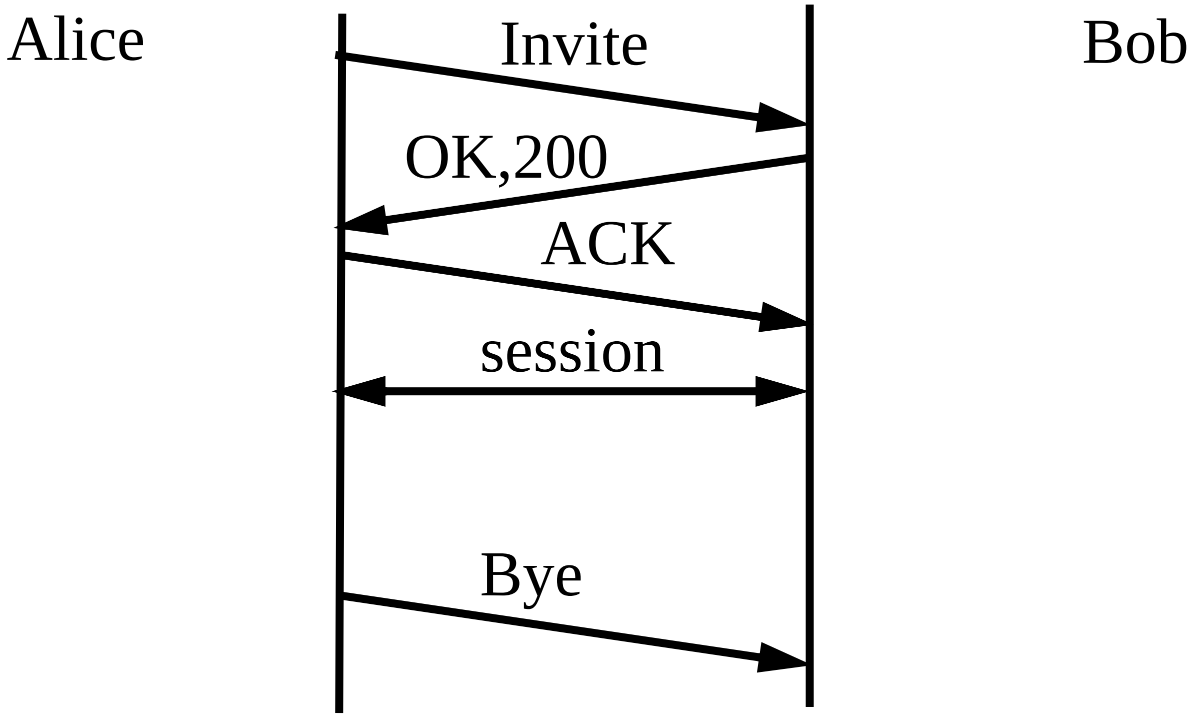
Several types of servers defined:

- User agent server runs on a SIP terminal = a client element, User Agent Client (UAC) + server element, User Agent Server (UAS)
- SIP proxy - interprets, and, if necessary, rewrites specific parts of a request message before forwarding it to a server closer to the destination:
  - SIP stateful proxy server - remembers its queries and answer; can also forward several queries in parallel.
  - SIP stateless proxy server
- SIP redirect server - directes the client to contact an alternate URI
- Location server - knows the current binding (from REGISTER msgs)

SIP uses SDP (Session Description Protocol) to get information about a call, such as, the media encoding, protocol port number, multicast addresses, etc.

SIP timeline

Alice invites Bob to a SIP session:



SIP Invite[1]

```
INVITE sip:bob@biloxi.com SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bK776asdhds
To: Bob <sip:bob@biloxi.com>
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE
Contact: <sip:alice@pc33.atlanta.com>
Content-Type: application/sdp
Content-Length: 142

(Alices SDP not shown)
```

SIP is a text-based protocol and uses ISO 10646 character set in UTF-8 encoding (RFC 2279). The message body uses MIME and *can* use S/MIME for security.

The generic form of a message is:

```
generic-message = start-line
                  message-header*
                  CRLF
                  [ message-body ]
```

1. Example from draft-ietf-sip-rfc2543bis-06.ps

Bob's response[1]

```
SIP/2.0 200 OK Via: SIP/2.0/UDP
pc33.atlanta.com;branch=z9hG4bK776asdhds
Via: SIP/2.0/UDP
bigbox3.site3.atlanta.com;branch=z9hG4bK77ef4c2312983.1
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE
Contact: <sip:bob@192.0.2.8>
Content-Type: application/sdp
Content-Length: 131

(Bobs SDP not shown)
```

1. Example from draft-ietf-sip-rfc2543bis-06.ps

SIP Methods

Method	Purpose
Invite	Invites a user to join a call.
Bye	Terminates the call between two of the users on a call.
Options	Requests information on the capabilities of a server.
Ack	Confirms that a client has received a final response to an INVITE.
Register	Provides the map for address resolution, this lets a server know the location of a user.
Cancel	Ends a pending request, but does not end the call.

SIP Status codes

SIP status codes are patterned on and simular to HTTP's status codes:

1xx	Provisional request received, continuing to process the request
2xx	Success - the action was successfully received, understood, and accepted
3xx	Redirection - further action needs to be taken in order to complete the request
4xx	Client Error - the request contains bad syntax or cannot be fulfilled at this server
5xx	Server Error - the server failed to fulfill an apparently valid request
6xx	Global Failure - the request cannot be fulfilled at any server

ENUM

IETF’s E.164 Number Mapping standard uses Domain Name Server (DNS) to map standard International Telecommunication Union (ITU-T) international public telecommunications numbering plan (E.164) telephone numbers to a list of Universal Resource Locators (URL). SIP then uses those URL’s to initiate sessions.

For example, ENUM DNS converts a telephone number in E.164 format, e.g. +46812345, and returns e.g., a Universal Resource Identifier (URI) SIP:olle.svenson@telia.se

Then a SIP client can make a connection to the SIP gateway telia.se passing the local part olle.svenson.

ENUM can return a wide variety of URI types.

Further Reading

IP Telephony (*iptel*)

PSTN and Internet Internetworking (*pint*)

Also important are the measures of delay, delay jitter, throughput, packet loss, etc. IP Performance Metrics (*ippm*) is attempting to specify how to measure and exchange information about measurements of these quantities.

A great set of references compiled by prof. Raj Jain is available at:
http://www.cis.ohio-state.edu/~jain/refs/ref_voip.htm

Summary

This lecture we have discussed:

- Network Management
- SNMP
- VoIP (including RTP)