IP Telephony signalling

Overview

H.323

SIP - session initiation protocol

Megaco (MGCP) - Media gateway control

IP Telephony Signaling alternatives

In Terminals — Intelligence — In Network

SIP
- ascii based
- devil in details
- also NNI coming
- Bakeoffs drive vendor interopera-bility

H.323
- Inherits ISDN
- complex
- still few services
- Vendor Interop has been announ-ced

Megaco/H.248/MGCP
- newest
- seems to be quality spec.
- architecture holds promise
- Interoperability?

SIGTRAN works on ISUP over SCTP over IP
- many (netheads) view this as an interim solution!

H.323 is a key standard for packet based multimedia communication

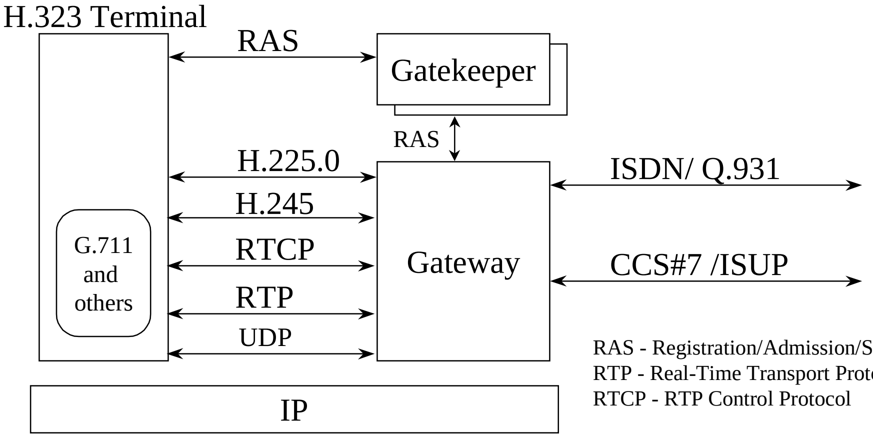
H.323 over: LANs, Enterprise Area Networks, MANs, Intranets, Internets
include dial-up connections and PP-connections over GSTN/ISDN with PPP packet transport.

Example networks:
- Ethernet (IEEE 802.3)
- Fast Ethernet (IEEE 802.3u)
- FDDI
- Token ring (IEEE 802.5)
- ATM

MM includes:
- Audio (mandatory)
- Video (opt)
- Data (opt)

Communication = conference or two party call.

H.323 uses H.225.0, H.245 and RTP



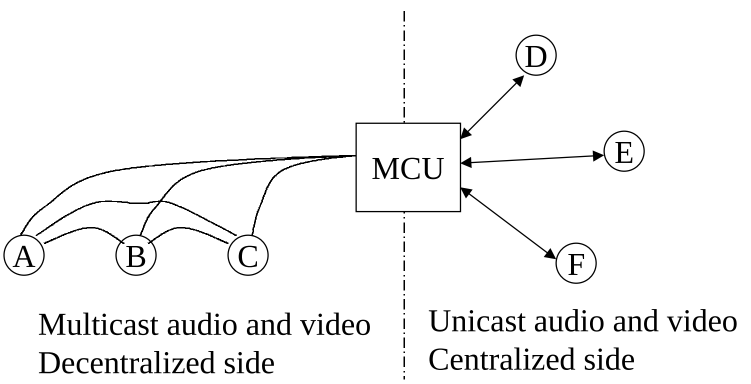
RAS - Registration/Admission/Status
RTP - Real-Time Transport Protocol
RTCP - RTP Control Protocol

Note: this is an example configuration!

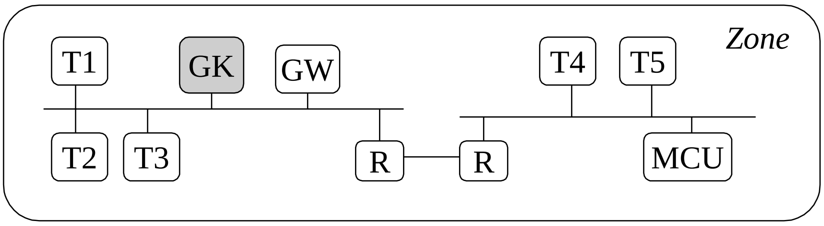
H.323 supports many call modes

- Directly between two H.323 endpoints (no GK)
- Between two H.323 endpoints using a GK
- Many conference types
  - ad hoc multipoint conference (start with 2-party call - expand to conf)
  - broadcast conference (one sender, many receivers)
  - broadcast panel conference (mp conf + bc conf)
  - centralized multipoint conference (trms pp to MCU, MP sends to trms)
  - decentralized multipoint conference (no MCU - all to all coms)
  - hybrid multipoint conference - centralized audio or video
  - mixed multipoint conference (mix of decentralized + centralized modes)

Mixed multipoint conference example

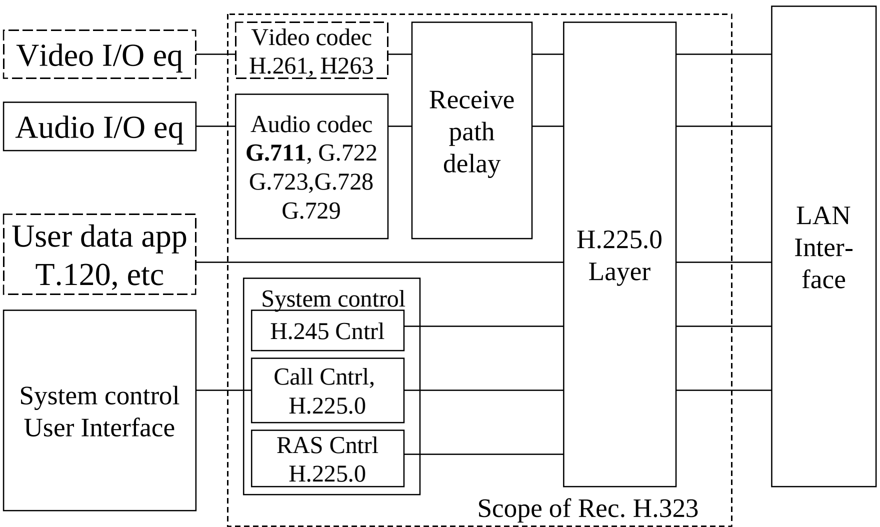


H.323 zone is controlled by a Gatekeeper



- Zone has at least one terminal, MCUs and GWs are optional.
- Zone has one and only one GK.
- Gatekeeper controls access to the network for Ts, GWs and MCUs and provides
  - address translation
  - gateway location
  - bandwidth management

H.323 terminal equipment



H.323 supports many parallel addressing methods

- H.323 entity shall have at least one Network Address (e.g. IP address)
- TSAP identifiers allow multiplexing several channels sharing one Network Address - map to port numbers
- An endpoint may have one or many Alias addresses - may represent the Ep or a Conference that the Ep is hosting. Include: E.164 numbers, H.323 IDs (e.g. John Smith), e-mail addresses. Aliases are unique in a zone.

Gateway translates between transmission formats, communication procedures and media formats

- Example: H.225.0 to and from H.221 (transm.f)
- H.245 to and from H.242 (comm procedure)
- Media format: Audio, video, data
- Represents characteristics of network endpoint to GSTN endpoint and the reverse. May also work as an MCU
- Can also do call set-up and clearing

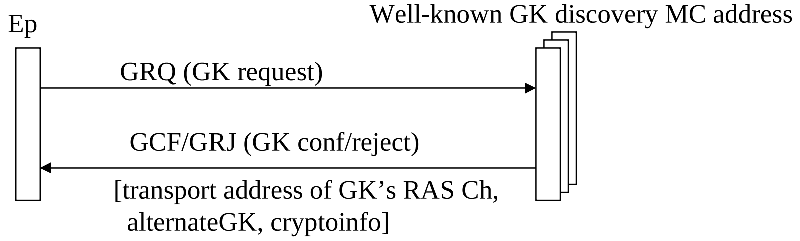
GK provides call control services, when present, shall do:

- Address translation (e.g. alias to transport address using DNS + E.164 to transport address)
  - uses the translation table produced from registration messages
- Admission control: ARQ/ACF/ARJ of H.225.0
  - based on call authorization, bandwidth, other criteria
- Zone management

GK may optionally do

- Call control signalling. May also direct the endpoints to setup call signalling channel between themselves
- Call Authorization using H.225.0 signalling
- Bwidth management controls the number of simultaneous calls in the zone
- Call management - keep list of calls -> busy conditions
- GK management, Directory service etc - FFStudy

Endpoint can discover a Gatekeeper automatically



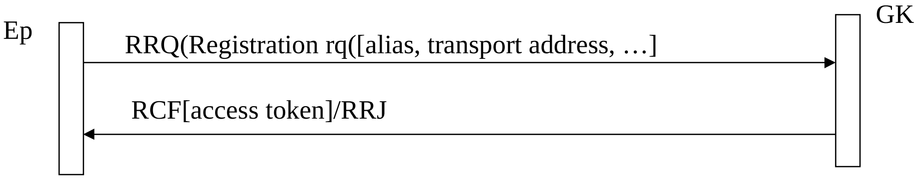
- Automatic discovery eases maintenance of individual terminals
- Terminals may also have the GK id configured

RAS signalling function

Endpoint —— RAS [uses H.225.0 msgs] —— GK

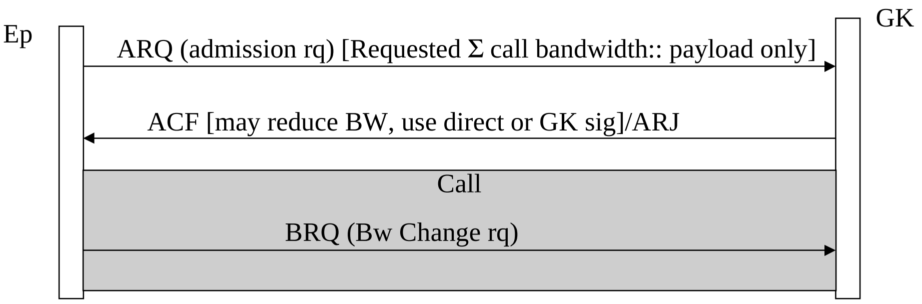
- Performs
  - Registration of endpoints, Admission of calls, Bandwidth changes for calls
  - Status
  - Disengage of endpoints.
- Uses RAS signalling channel =/= call signalling channel and H.245 control channel. GKs have a well def. TSAP id for RAS sig. channel
- Endpoint=H.323 terminal or GW or MCU (is callable)

Endpoints register using GK's RAS Channel Transport Address prior to any calls are made

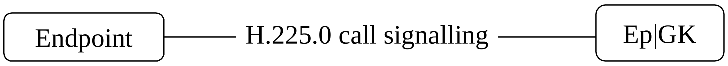


- Security policy may require that registration has time-to-live and has to be repeated from time to time
- Endpoint or GK may unregister using the URQ message
- The GK maintains an alias to Network Address translation table.
- Access token may be used later in call setup

Call Admission sets the upper limit for the aggregate bitrate of the call



Call signalling uses H.225.0



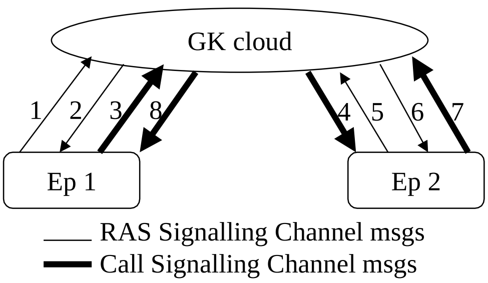
- Call signalling= call setup, request changes in Bw of a call, get status of Ep, disconnect call
- Call signalling is largely inherited from ISDN
- Call Signalling Channel is opened prior to H.245 procedures and prior to any other logical channels between endpoints. Eps have a well known TSAP id for the Call Sig. Channel and a well-known Discovery Multicast address.

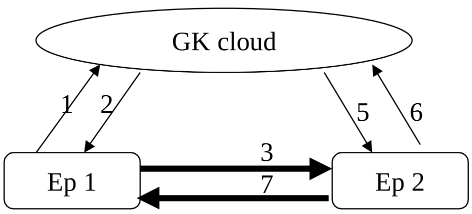
H.323 Call Signalling Channel Routing

- ACF has the Transport Address of the Call Signalling Channel
- The address is either a GK address or an Endpoint address.

Gatekeeper Routed Call Signalling

Direct Endpoint Call Signalling



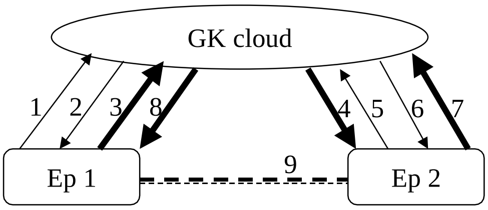


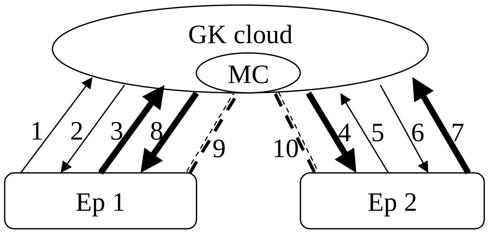
1,5 - ARQ 3,4 - Setup
2,6 - ACF 7,8 - Connect

H.245 Control Channel Routing

The goal of call signalling is the setup/release of H.245 Control Channel!

Direct H.245 Control Channel **GK routed H.245 Control**





RAS Signalling Channel msgs
Call Signalling Channel msgs
H.245 Control Channel

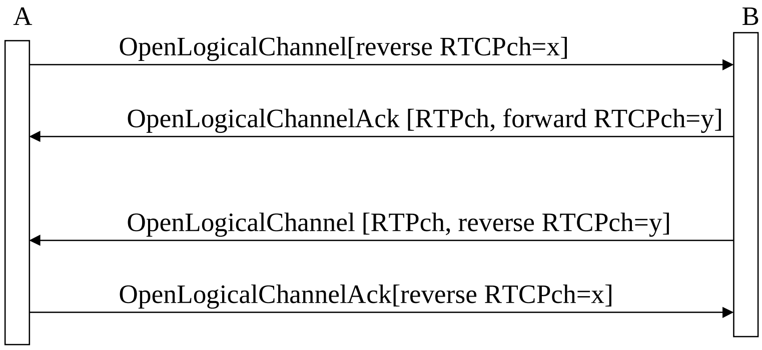
1,5 - ARQ 3,4 - Setup
2,6 - ACF 7,8 - Connect
9,10 - H.245 Channel

H.245 carries end-to-end control messages between H.323 entities

Terminal —— H.245 Control Channel —— Terminal|GW|MC|GK

- Master/slave determination for conflict resolution
- Capability Exchange (e.g. what codecs are supported)
- Logical Channel Signalling (bounds media type, algorithm etc. to ports)
- Bidirectional Logical Channel Signalling
- Close Logical Channel Signalling
- Mode Request (conference modes)
- Round Trip Delay Determination
- Maintenance Loop Signalling
- H.323 also uses flowControlCommand of H.245 to limit bandwidth

Sample H.245 Logical Ch Signalling for two way RTP+RTCP communications setup

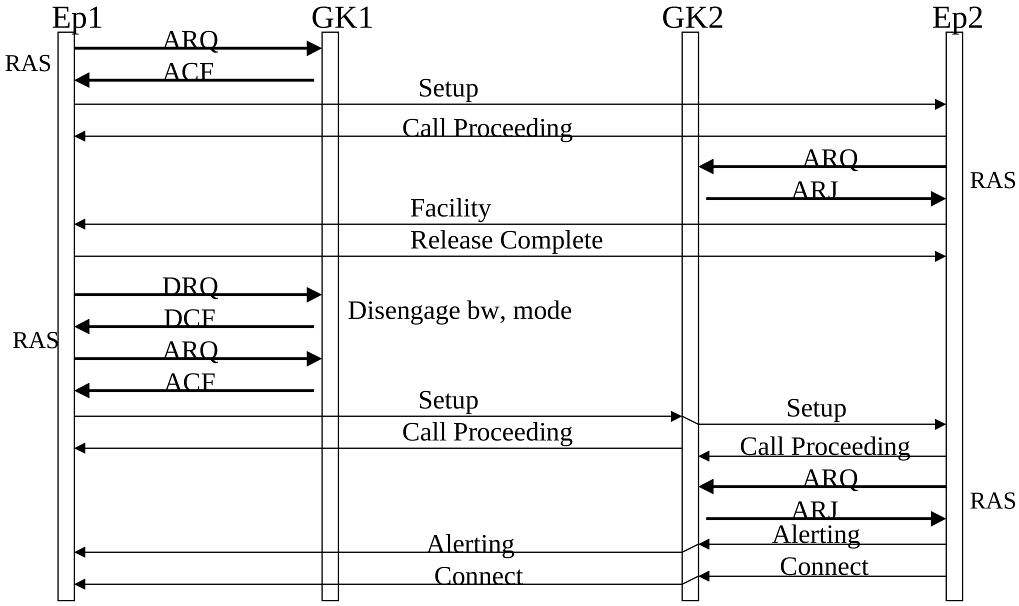


- In IP networks a logical channel corresponds to an IP port number
- Uses H.245 Control Channel

H.323 Call identification uses

- *Call reference value* - between two H.323 entities on a signalling channel (one for call signalling and another for RAS channel
- Call ID - a globally unique non-zero value created by the calling endpoint passed in all H.225 msges
- Conference ID (CID) - in all sub-calls of a conference

Both endpoints registered - Direct/GK-routed call signalling



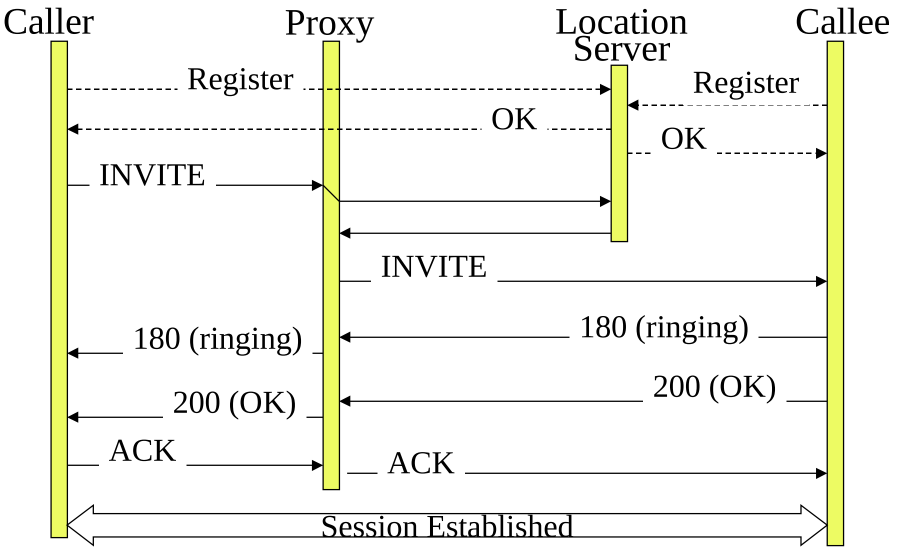
H.323 summary

- H.323 inherits call signalling from ISDN
- H.323 has many conference modes and many signalling and call routing options
- Call setup delay is reduced by using the Fast Connect Procedure: packs all setup info from both H.225.0 and H.245 into fastStart element in *setup* and *connect* (call proceeding, alerting) messages

SIP overview

- Simplicity
  - Ascii based - simple tools for development
  - Lower call setup time than in H.323
- Proxy and redirect servers
- Caller preferences
- Ability to support many media types
- Forking
- Used between both services and call control entities
- Has been adopted as the basis for 3G.IP signalling
- Originally subscriber signalling, proposed also as network to network signalling
- Quality of Specification is not very good! Leaves a lot of decisions to the implementor

A typical SIP call setup

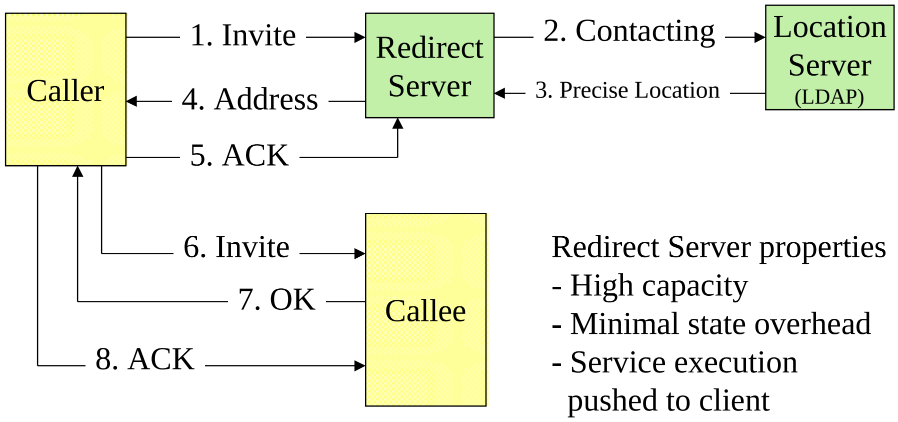


Stateful Proxy vs Stateless Proxy

- Maintains call context
- Replicates UAS/UAC to process requests and responses
- Call state and transaction state can be maintained
- Forking proxies require state
- TCP proxies must be stateful for reliability
- Enhanced services require state
- Can collect charging info

- No call context
- Response is not based on UA replication
- Provides client anonymity
- Restricted gateway access
- High processing capacity
- Easier to replicate than the stateful proxy
- Also semi-stateful is possible

Redirect Server pushes processing to clients

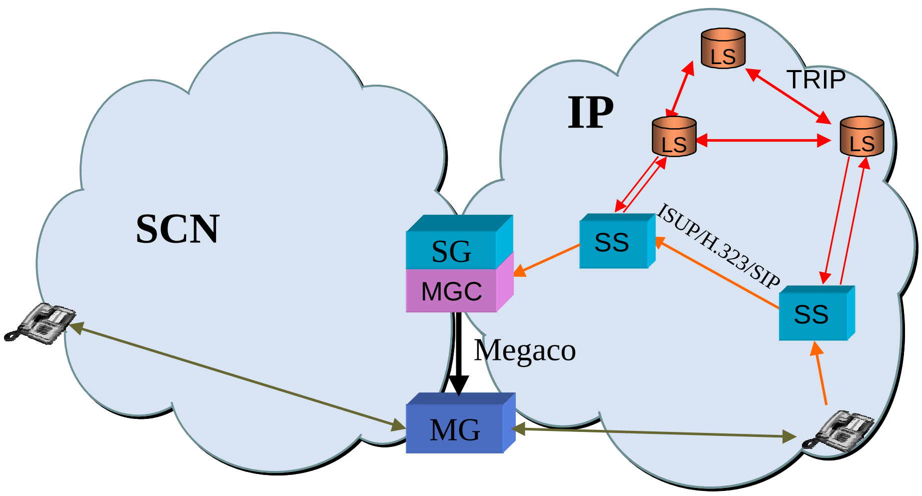


Redirect Server properties
- High capacity
- Minimal state overhead
- Service execution pushed to client

Megaco - Media Gateway Control protocol is the newest entrant

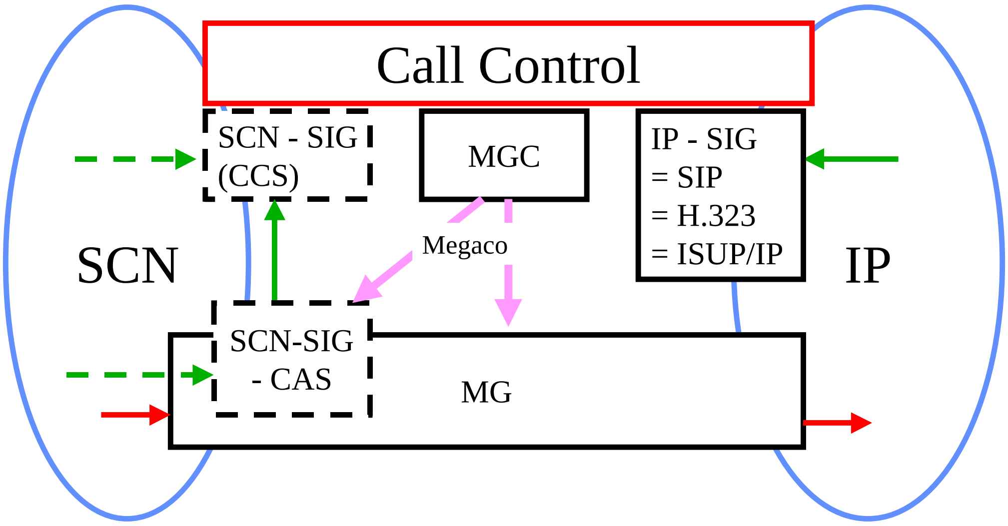
- MGCP has been adopted by Cablelabs = US CATV R&D body as the CATV Telephony standard
- ITU-T is making its own variant called Megaco
- Megaco, MGCP are master-slave protocols by which media gateways can be configured e.g to services - in case of residential media gateway, MGCP becomes a subscriber signalling system

Current Architecture



TRIP = Telephony Routing over IP, SG - Signalling Gateway, MGC - Media Gateway Controller
MG - Media Gateway, SS = Signaling Server, LS = Location Server

Gateway decomposed



Use case of MGCP

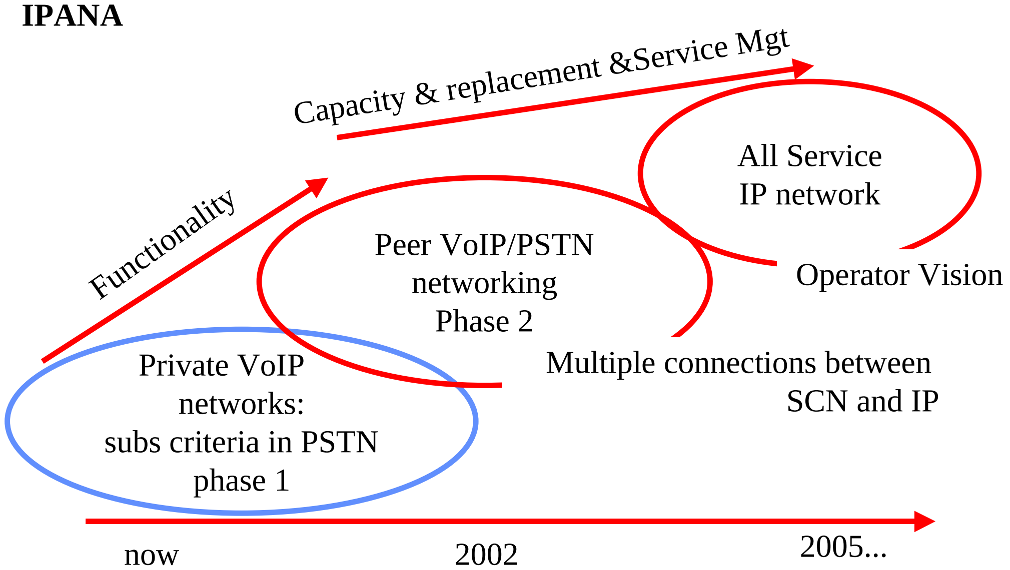
- Media gateway is integrated into the digital CATV set-top box
- MGCP is used to control the residential media gateway - e.g one can download a dialling pattern telling what to do depending on what the user has dialled - this is a step towards *service independent signalling = new services can be deployed without updating GW software*

IP Telephony Research (IPANA) in the Laboratory of Telecom Technology

- Technology evaluation
  - Delay measurements breakdown
  - SIP call waiting
- Numbering and Routing Information Interoperability with ISDN

Roadmap to the Future

IPANA



Interoperability Issues

IPANA

- Signaling and Call control
- Quality of Service

Phase 1
--->

- Telephony Routing and addressing
  - Input Information gathering
  - Alternative routing over IP

Phase 2
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- Service Management in the hybrid network

Phase 3

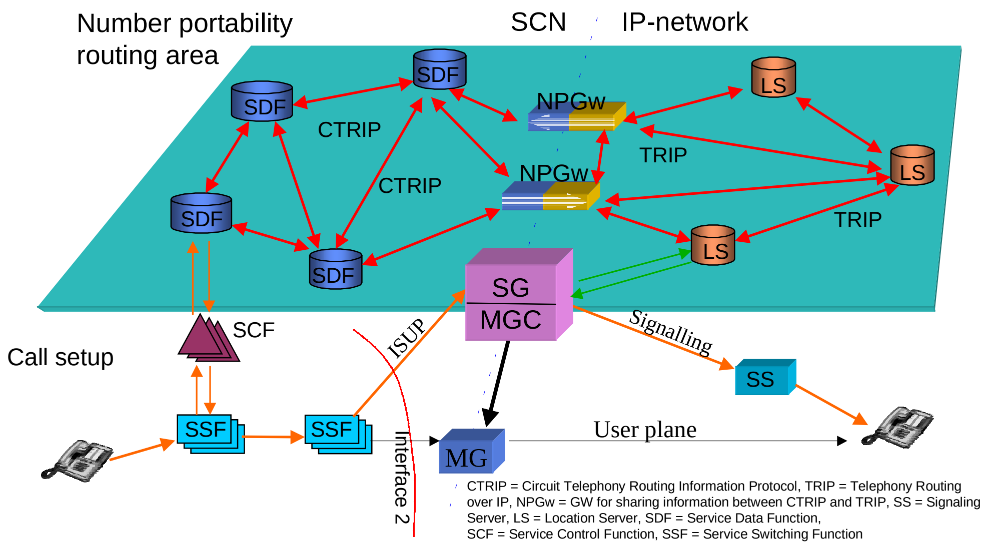
Phase 2 Requirements

IPANA

- Efficient routing and numbering infrastructure across the emerging hybrid network is a necessity
  - Delay and jitter highly depend on call path
- In all call scenarios at all cost we must avoid unnecessary conversion between IP and PSTN.
  - Call Forwarding, Number Portability, Roaming, 800-numbers ...

The solution is CTRIP + Numbering gateway

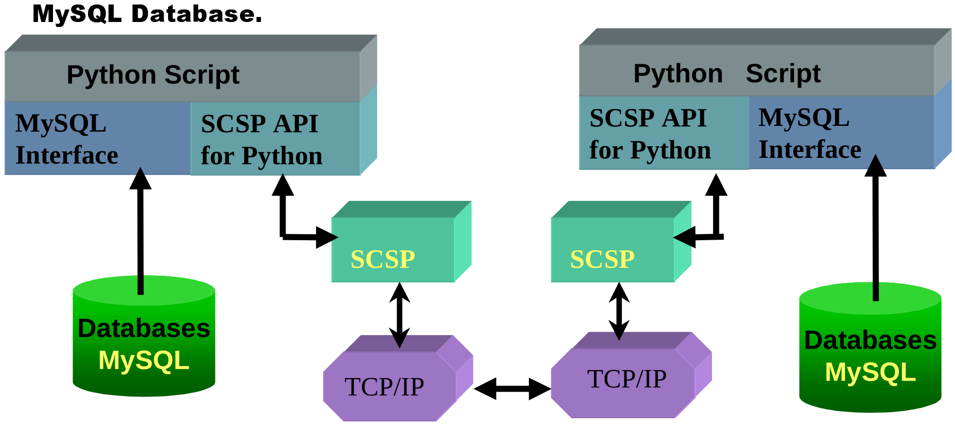
IPANA



TRIP and CTRIP Modules

IPANA

We use Python as Interface to work with SCSP primitives through Python objects and access to the information (CRO) stored in the MySQL Database.



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