VoIP Core Technologies

Aarti Iyengar

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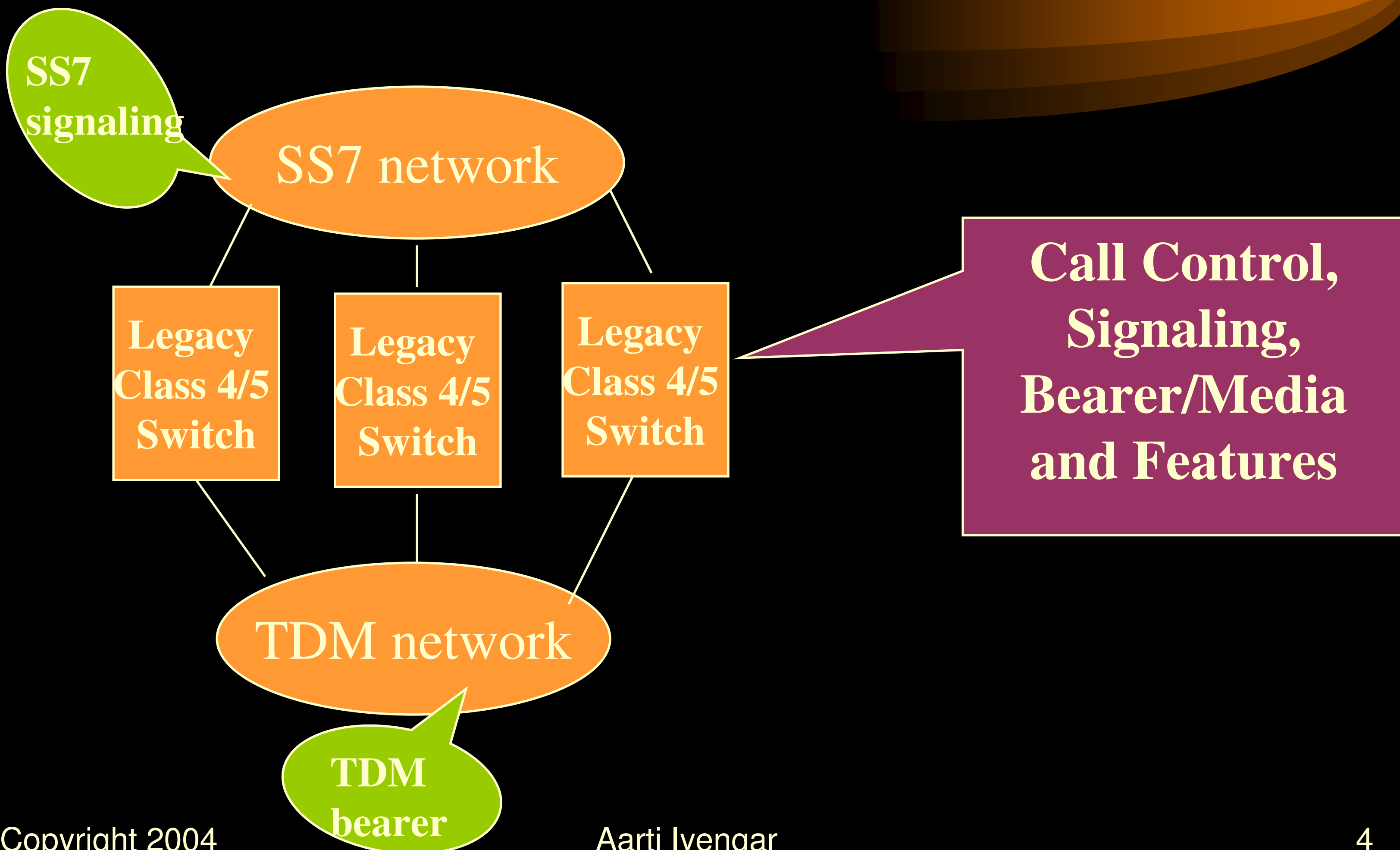
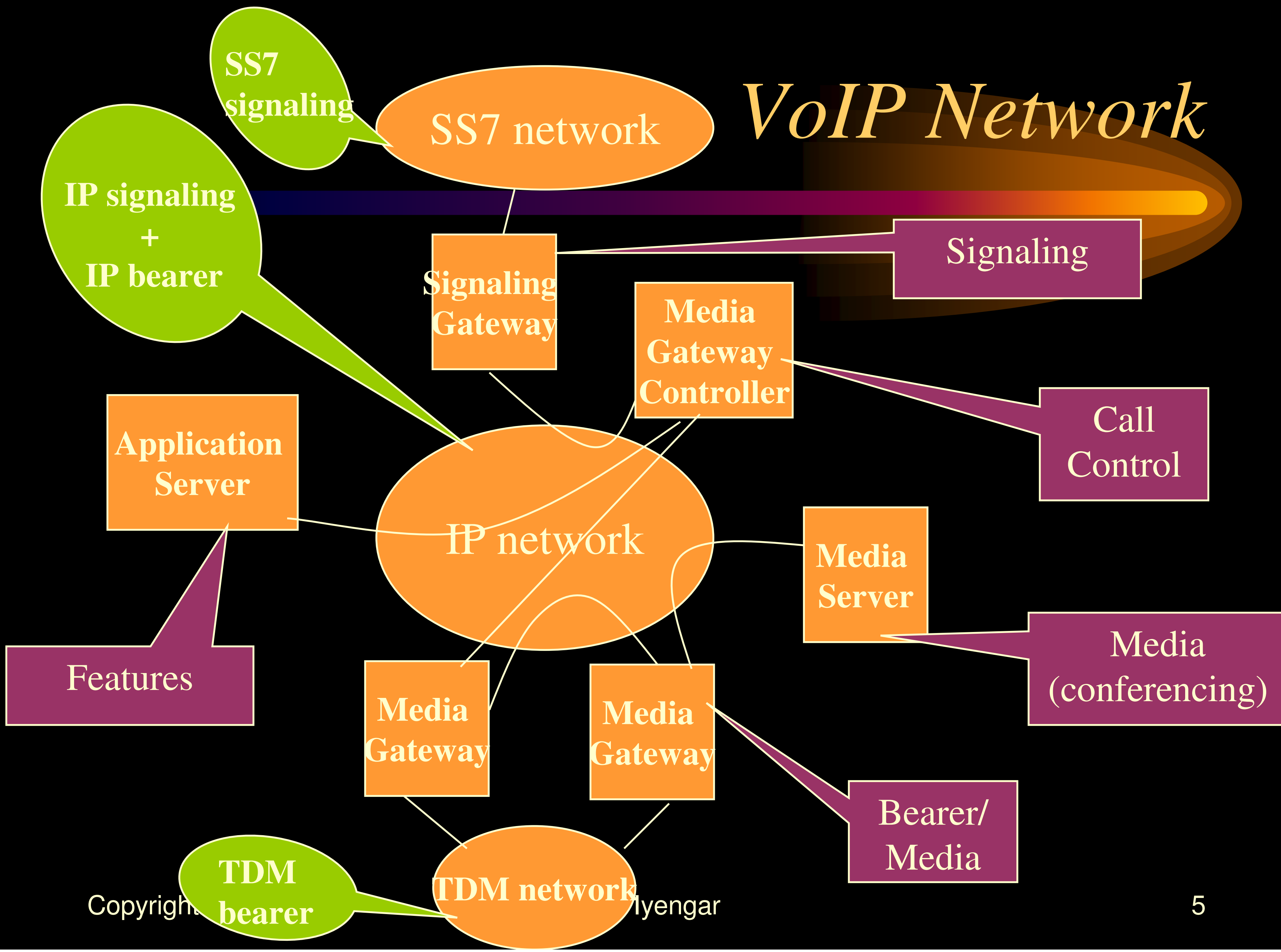
What is VoIP?

- Legacy Telephony
  - TDM/SS7 based infrastructure
  - Traditional Class 5/Class 4 switches
- Voice over IP
  - IP-based packet infrastructure for PSTN voice transport
  - New elements that collectively perform traditional functions and more
- And what is Internet Telephony?

Traditional PSTN Network



VoIP Network



VoIP Network Paradigms

- Centralized a.k.a Master/Slave model
- Distributed a.k.a Peer Model

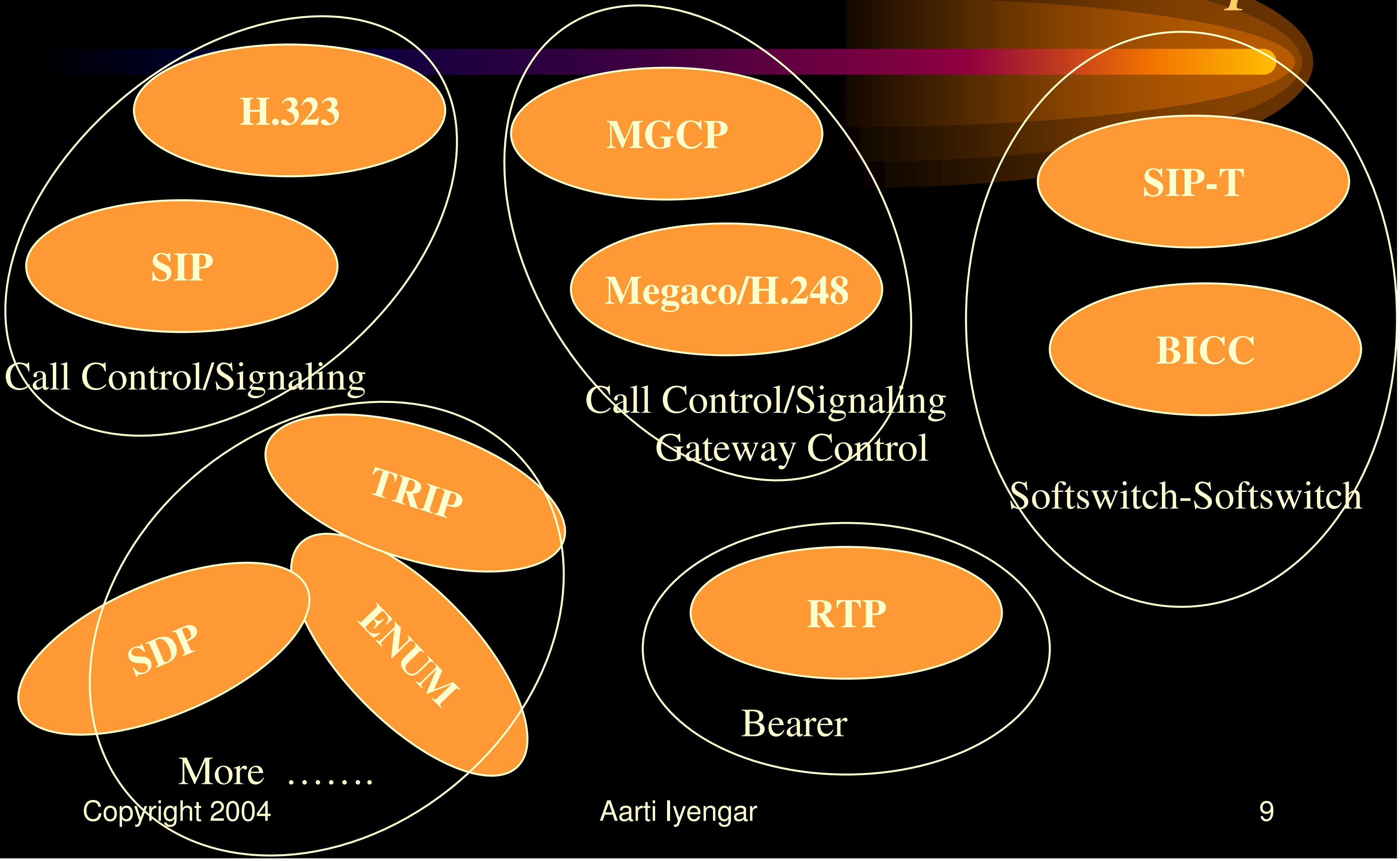
VoIP Network Paradigms (contd.)

- Centralized model
  - Dumb endpoints (media gateways, IADs, phones) and intelligent central entity (call agent or controller)
  - Controller instructs, the endpoints obey
  - More akin to legacy telephony model
  - Well suited to basic telephony features
  - Single intelligent point, hence simpler and easier to manage/maintain

VoIP Network Paradigms (contd.)

- Distributed model
  - Intelligence is distributed throughout the network
  - Client-Server model with intelligent clients (user agents, terminals, IP Phones) and intelligent servers
  - Very well suited for advanced telephony services through service innovation
  - Relatively, more complex because of multiple intelligent points

VoIP Protocol Soup



VoIP Protocols

- Several protocols have been defined by different standards bodies (IETF/ITU)
  - Many perform similar functions but have intrinsic differences
- Next few slides will discuss some of these protocols
  - Each has its own merits/demerits
- *Views expressed here are purely personal*

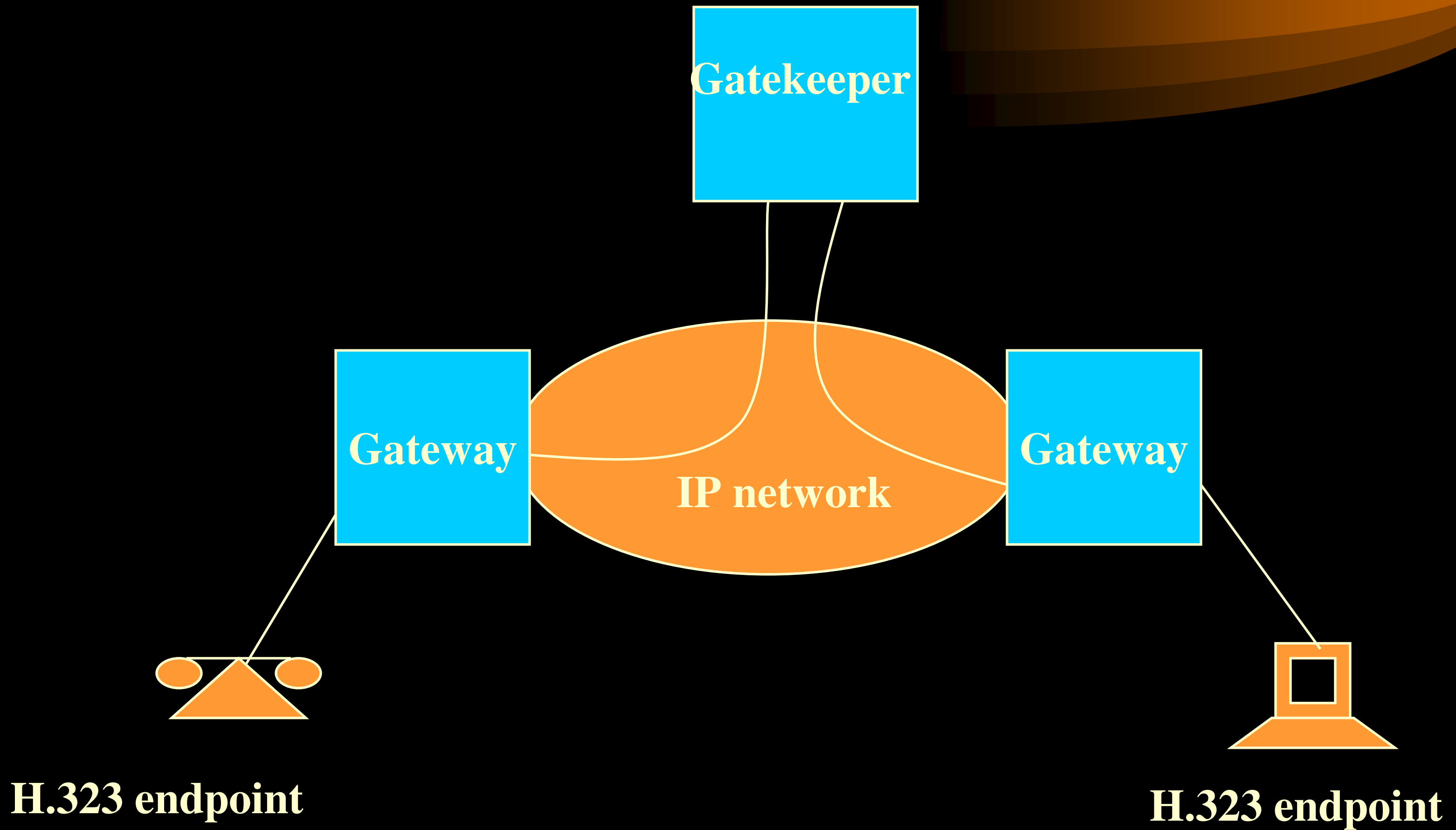
Call Control Signaling Protocols: H.323

- ITU-T defined standard
- Originally developed for ISDN based multimedia services over LAN
- Distributed protocol model
- Consists of
  - Terminals
  - Gatekeepers
  - Gateways
  - Multipoint control units

Call Control Signaling Protocols: H.323

- Umbrella protocol comprising of several other protocols like H.225, H.245, T.120 etc.
- H.323v4 currently implemented everywhere
  - Future H.323v5

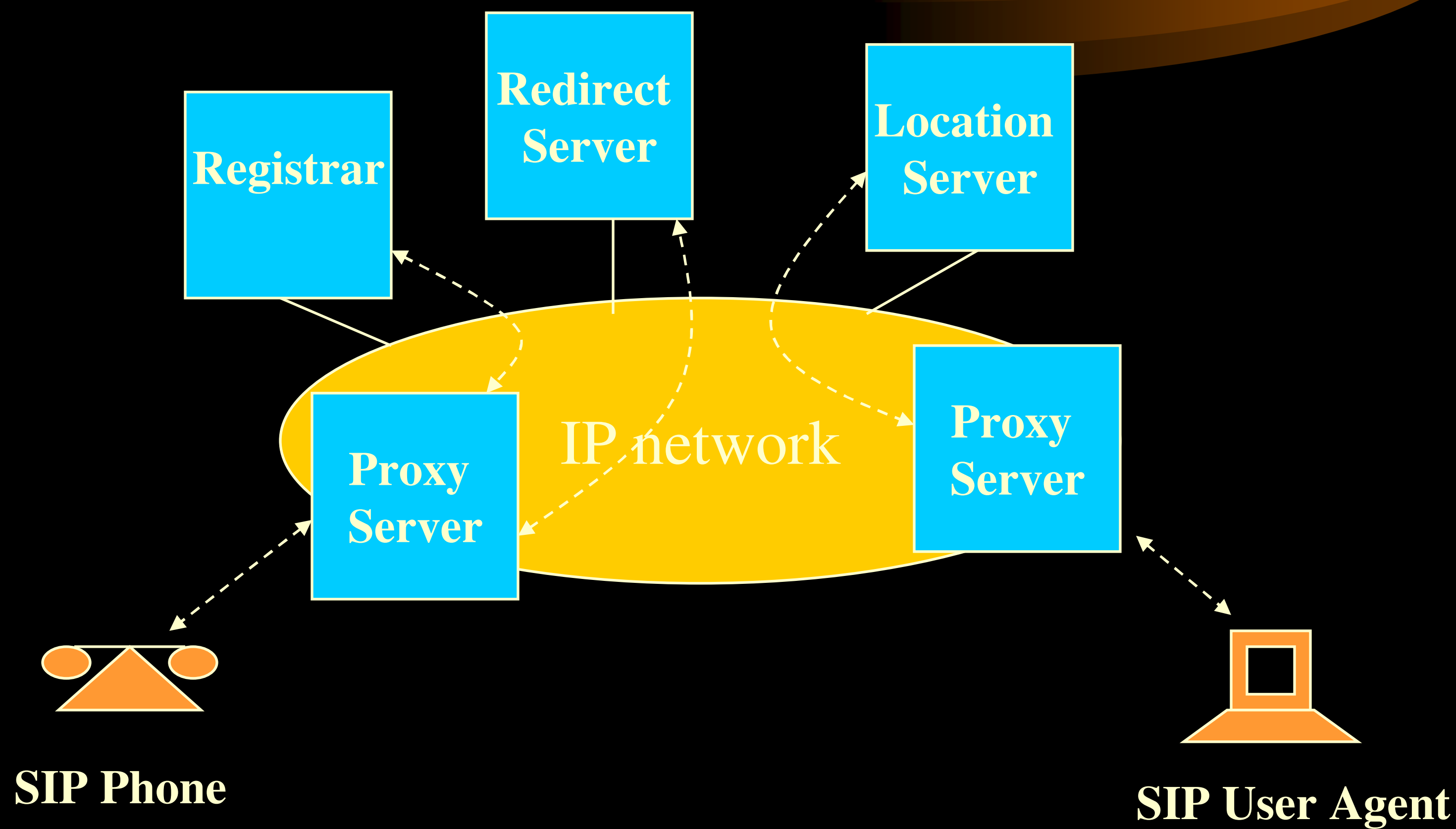
Call Control Signaling Protocols: H.323



Call Control Signaling Protocols: SIP

- IETF RFC 3265 (obsoletes RFC 2543)
- Developed for multimedia services over IP networks
- Designed to employ existing popular Internet protocols like DNS, SDP etc.
- Distributed model consisting of User agents and Servers

Call Control Signaling Protocols: SIP



H.323 versus SIP

General Perceptions

H.323

- Originally developed by the ITU for ISDN based multimedia services
- An umbrella suite of protocols like H.225, H.245, H.235 etc. Monolithic in nature as it defines everything including RAS, Capability negotiation etc.
- Perceived as more complex because of binary ASN.1 encoding, number of protocols, parameter extensibility, stack size

SIP

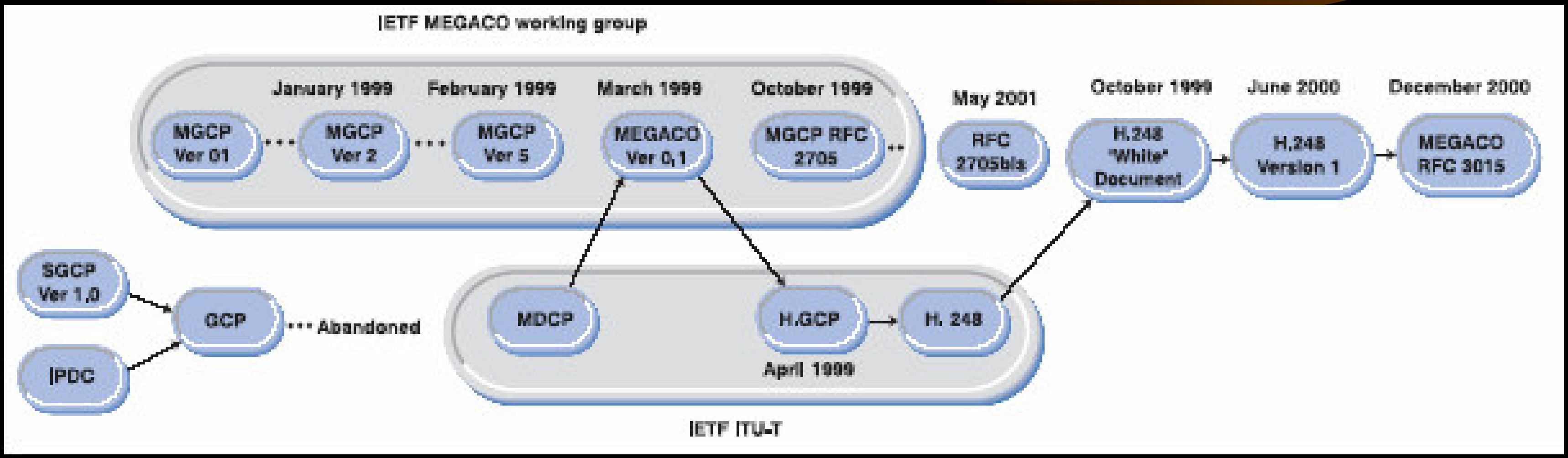
- Developed by the IETF for Internet Telephony and advanced multimedia applications
- Utilizes already developed popular Internet protocols like SDP, DNS etc.
- Text-based implementation is perceived to be simpler, modular, easily adaptable to the www and easier parameter extensibility.

H.323 versus SIP

Reality?

- ITU protocols more tightly defined; IETF looks for looser working code
- H.323 older and more established; SIP relatively newer but fast catching up
- H.323 widely deployed today; SIP is being widely adopted by large players
- Importance of the Internet and web-based applications increasing
- SIP capable of giving service providers greater control of services, extensibility and interoperability with the www; hence, may eventually win the race
- For a long time however, both these protocols need to co-exist
  - Robust standards must be developed to define interoperability to make things easier

Call Control Signaling Protocols: GCP Evolution



Source: Hughes Software Systems

Call Control Signaling Protocols: MGCP

- IETF informational RFC 3661
- Provides call control services in a packet network
- Centralized model
- Consists of media gateways and call agents
  - Call Agents-> centralized intelligent entities handling call control and signaling
  - Media Gateways-> dumb devices handling media
  - Call Agent communicates with Media Gateway via MGCP

Call Control Signaling Protocols: Megaco/H.248

- Enhances MGCP
- Provides call control services in a packet network
- Joint effort by ITU and IETF (IETF nomenclature- Megaco, ITU nomenclature- H.248)
  - IETF RFC 3525
- Centralized model
- Consists of Media gateways and Media gateway controllers
  - MGC-MG communication via Megaco/H.248

MGCP versus Megaco

MGCP

- Early implementation of master-slave protocol for call control signaling in packet networks
- Now a closed effort from a standardization perspective
- Limited support of networks for interfacing with the PSTN
- Deals with endpoints and connections
- MGCP implementations do exist today. MGCP variants NCS/TGCP are adopted by Packetcable.

Megaco/H.248

- Later effort primarily to enhance the capabilities of MGCP
- Standards efforts in the MGC-MG space focused here
- Developed to support ATM, IP networks
- Deals with contexts and terminations; decouples physical terminations from logical (ephemeral) ones and is more suited to handling multimedia
- More complete and robust, hence widely agreed to be the standard of the future allowing for multi-vendor interoperability

Controller - Controller Protocols: SIP-T

- IETF RFC 3372
- Defines a framework to interface SIP with ISUP
  - To maintain feature transparency in the SIP network w.r.t PSTN to support IN services not supported in SIP
  - To deliver SS7 information (in its entirety) to some trusted SIP elements
- Integration methods
  - Encapsulation of ISUP within SIP using MIME
  - Translation of ISUP parameters to SIP header
  - Provision to transmit mid-call ISUP signaling messages through INFO method

Controller - Controller Protocols: SIP-T

- Implemented at SIP-PSTN boundary gateways
  - Carried end to end
- SIP-T is relevant in the following scenarios
  - PSTN origination, IP termination
  - IP origination, PSTN termination
  - PSTN origination, PSTN termination with IP transit
  - IP origination, IP termination : SIP-T is not required

Controller - Controller Protocols: BICC

- Development triggered by a need for a packet-based PSTN replacement
- Functional separation of call and bearer signaling protocols in a broadband network
  - IP/ATM bearers in addition to TDM bearer
  - Uses SS7 signaling (with extensions to ISUP)
  - Binding information allows correlation between call control and bearer

Controller - Controller Protocols

- BICC defines three capability sets
  - CS1: supports ATM-based (AAL1/AAL2) bearer
  - CS2: supports IP-based bearer
  - CS3: still in works to support advanced services and interoperability with SIP

SIP-T versus BICC

SIP-T

- IETF defined
- Defined to maintain feature transparency across SIP networks (deliver ISUP to SIP endpoints)
- Packet-based signaling and bearer
  - SIP signaling
  - IP bearer; ATM supported through RFC 3108 (may be some issues, but defined)
- Provides a transition to pure advanced multimedia services based SIP network

BICC

- ITU defined
- Defined to separate call control from bearer (extends ISUP to handle packet bearers)
- SS7 signaling, Packet bearer
  - Network is SS7 (ISUP) signaled
  - TDM/ATM/IP bearers
- Intended for packet-based next-generation network supporting all existing legacy services.

Bearer Protocol: RTP

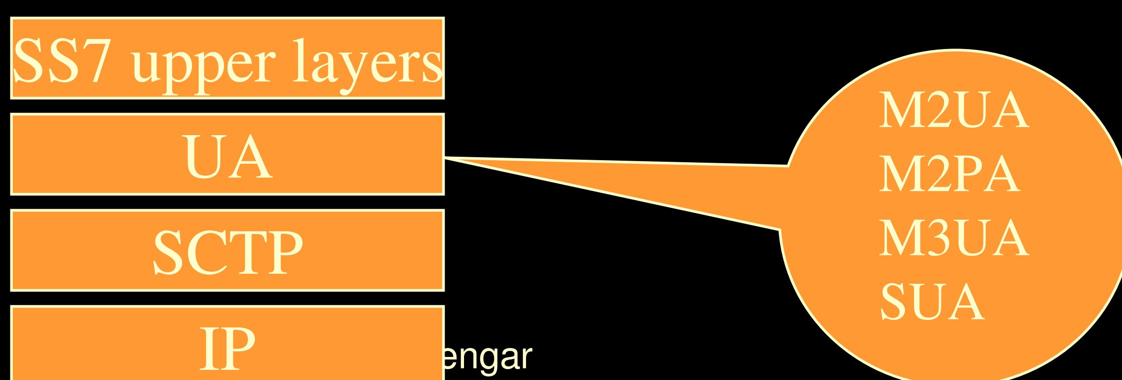
- IETF RFC 3550 (obsoletes RFC 1889)
- End-to-end network transport services for multimedia applications
- Services include payload type identification, sequence numbering, time stamping and delivery monitoring
- Control protocol (RTCP) to monitor data delivery
- Can be used with any transport protocol
  - Depends upon underlying transport layer for QoS
  - Applications typically use RTP over UDP

Others…

- Many other protocols are part of a VoIP network
  - Session Description protocol (SDP) to describe the session
  - Telephony Routing over IP (TRIP), E.164 Numbering (ENUM), Signaling Transport (Sigtran) are still being refined in the standards
  - Dynamic Host Configuration Protocol (DHCP), Domain Name Service (DNS) etc. are intrinsic to any IP network and are needed for its proper functioning
    - These are not exclusive to VoIP and will NOT be discussed in this presentation

Others…SIGTRAN

- To address transport of PSTN signaling over IP networks
- Allows SS7 signaling to be carried between the Signaling Gateway and IP signaling points (MGC or IP SCPs)
- Carriers can maintain existing SS7 infrastructure
- Protocol Suite consists of
  - Transport Protocol (SCTP), User Adaptation Layers (UA)



Others…SDP

- IETF RFC 2327
- Format for describing multimedia sessions
- Conveys session set up information to the participants for session announcement and session invitation
- Information includes session name, session duration, media type, information like address, port, format to receive the media etc.
- SDP consists of one session level description and optionally several media level description

Others…TRIP

- IETF RFC 3219
- Policy driven inter-administrative domain protocol
- Advertises reachability of telephony destinations and route attributes to the destinations
- TRIP is independent of the signaling protocol
- TRIP is modeled after BGP-4
  - used to distribute telephony routing information between administrative domains

Others…ENUM

- IETF RFC 2916
- To facilitate convergence of PSTN and IP networks
- Philosophy – Telephone number in, URL out
- Very familiar Telephone number to be used as universal ID to access Internet Services
- Employs principles of Domain Name Service (DNS)
- Powerful application in VoIP, email service, instant message etc.

Summary

- Several competing protocols in the VoIP arena
  - Overlapping functionality and inherent differences
  - Co-existence and Inter-working of multiple protocols essential for a functional VoIP network (today)
- Centralized and distributed paradigms parallel existence
  - H.323/SIP enabled endpoints for multimedia features
    - SIP-H.323 battle will continue for a while!
  - MGCP/H.248 both suitable for basic telephony applications as gateway control protocols
    - H.248/Megaco final standards derivative
      - Emerging as the future standard for MG-MGC communication for interfacing with the PSTN
    - BUT MGCP will exist for some time
    - Packet Cable NCS and TGCP based on MGCP

Summary

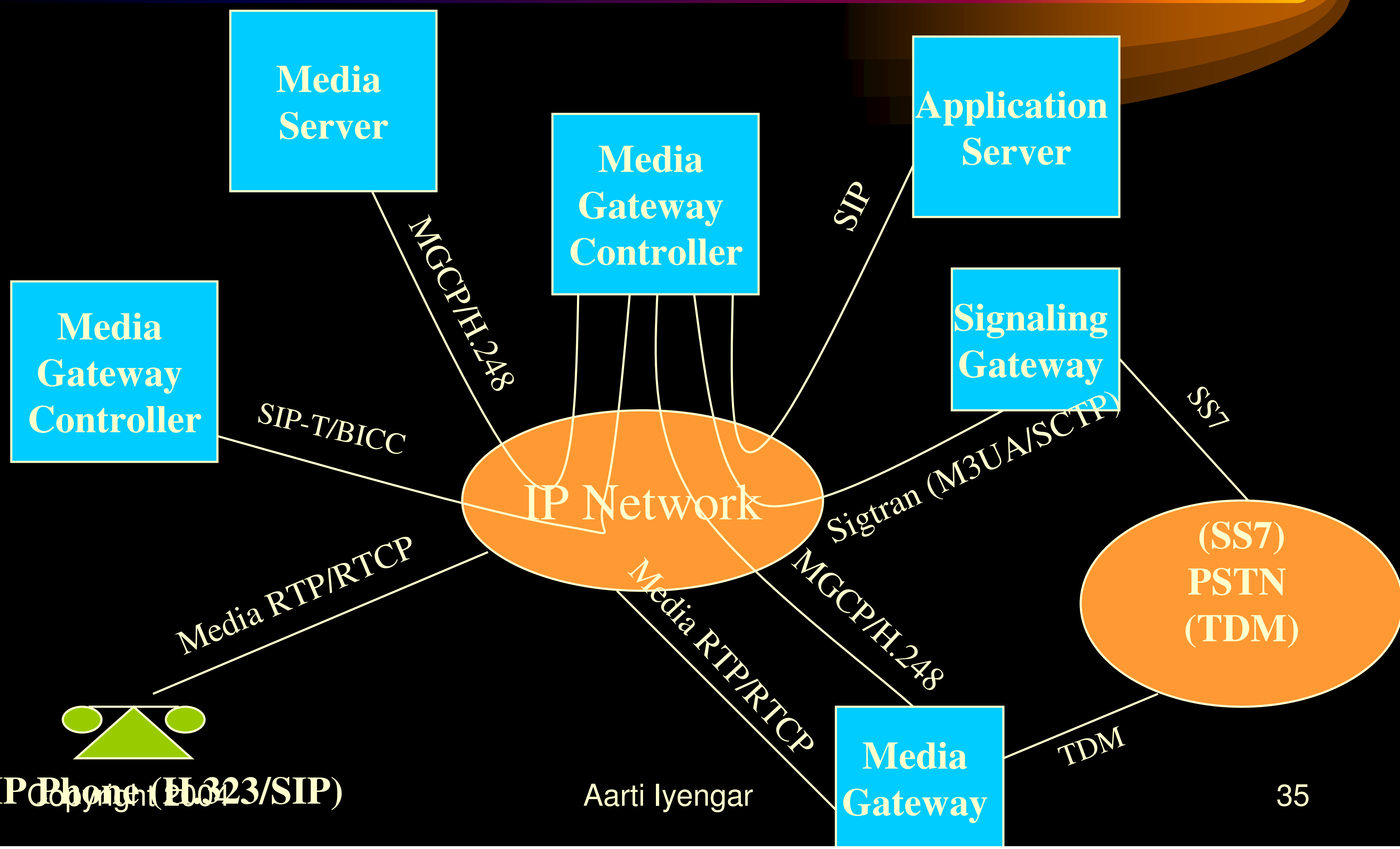
- IP network – SS7 interface
  - SIP-T and BICC are similar in functionality
  - SIP-T may be suitable for operators who want to migrate to SIP-based packet networks offering advanced multimedia services
  - BICC may be suitable for incumbent operators networks emulating legacy PSTN who only want bearer independence from signaling
  - Sigtran defines complete SS7 transport over IP to interact with SS7 elements in a packet network (SG, IP SCP)
    - Eventually, SIP may be able to handle it all??
- All VoIP deployments will use RTP for bearer media

Summary

Key to protocol selection is a strategic decision depending on existing network and future services planned

The winners in the protocol race would be the mighty ones standing test of time, industry acceptance and hype generated.

Ultimately, one winner will make it easy for all !



Glossary

BICC: Bearer Independent Call Control
DHCP: Dynamic Host Configuration Protocol
DNS: Domain Name Service
ENUM: E.164 number
IETF: Internet Engineering Task Force
MG: Media Gateway
MGC: Media Gateway Controller
MGCP: Media Gateway Control Protocol
MIME: Multi-purpose Internet Mail Extensions
NCS: Network Based Call Signaling

Glossary

RTP: Real Time Protocol

RTCP: Real Time Control Protocol

SDP: Session Description Protocol

SG: Signaling Gateway

Sigtran: Signaling Transport

SIP-T: SIP for Telephony

SIP: Session Initiation Protocol

TDM: Time Division Multiplexing

TRIP: Telephony Routing over IP