

# Architecture of a Global H.323/SIP-enabled VoIP Network: An ITSP Perspective

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# Agenda



- **Overview**
  - **Internet Telephony Service Providers (ITSPs)**
  - **Why SIP and H.323 must co-exist**
  - **Comparison of H.323 and SIP protocols**
- **Sample architectures**
  - **Typical wholesale ITSP VoIP networks**
  - **H.323 and SIP inter-networking architectures**
  - **Migration challenges**
- **iBasis SIP Self Certification Laboratory**
- **Summary**
- **Questions and answers**

# Internet Telephony Service Providers



- **What is an ITSP?**

- A service provider that provides real-time voice services over an IP backbone or network

- **Types of ITSPs**

- Retail (Net2Phone, Dialpad, etc.)
- Wholesale (iBasis, ITXC, Genuity, etc.)

- **A word about iBasis**

- Tier 1 US carriers for international termination
- Over 90% of our traffic is transported over the public Internet
- Carefully engineered IP network routes and monitoring
- Roughly 1 Billion minutes transported per year
- 46+ countries and over 110 carriers and customers

# Why SIP and H.323 Must Co-exist



- **Established base of large-scale H.323 networks**
  - ITSPs have been around for several years
  - Equipment vendors only supported H.323 until recently
- **SIP is the new “buzz” word**
  - Gaining industry momentum (embedded SIP clients in future operating systems and PDAs)
  - Protocol based on Internet standards
  - Concept of presence in the network brings with it new and exciting services
- **As a result, existing ITSPs are now beginning to offer SIP services along with their existing H.323 services**

# Comparison of H.323 and SIP

iBasis

## H.323

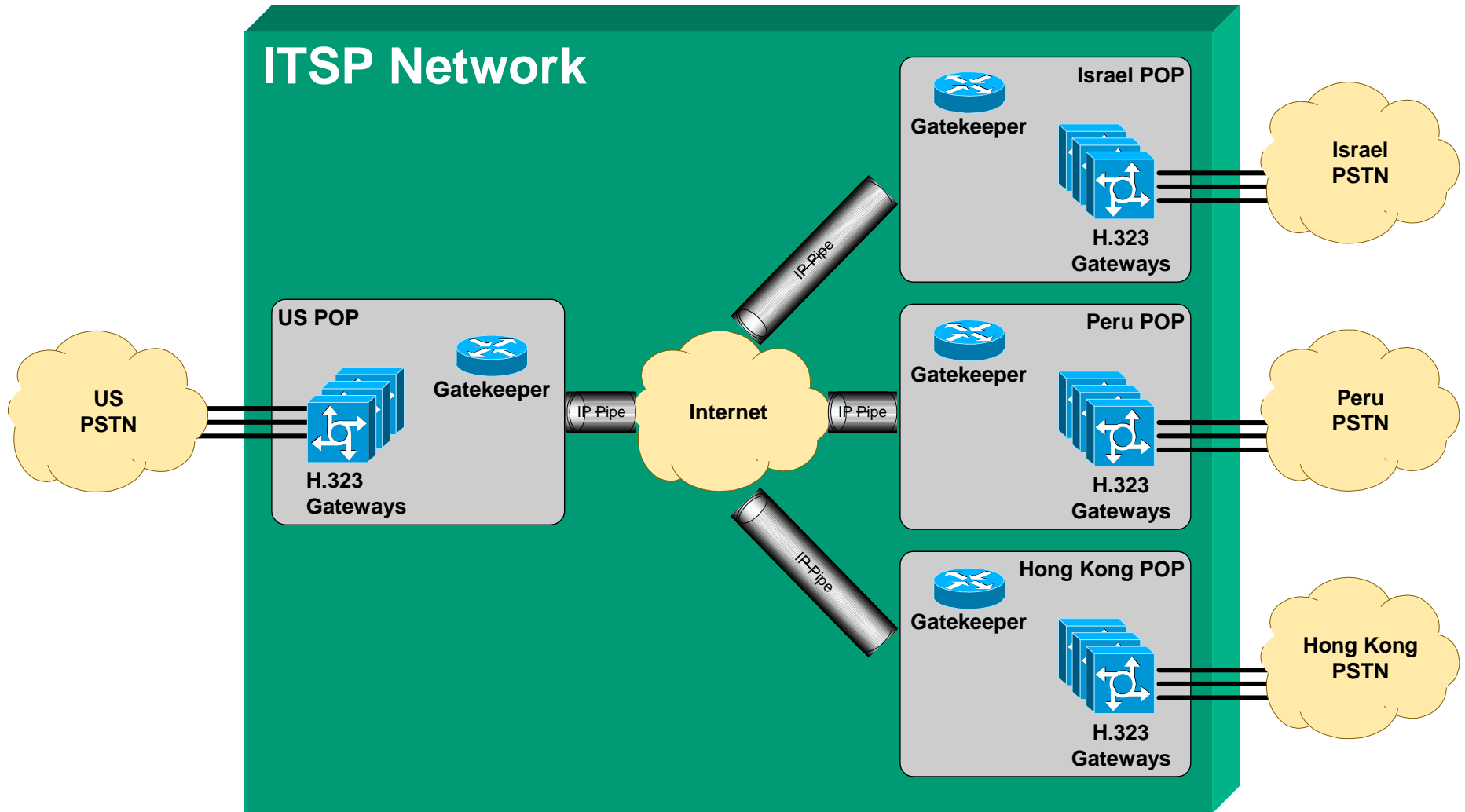
- ITU Standard
- Binary-based ASN.1 encoding
- Utilizes other ITU standards such as ISDN (Q.931) and Q.SIG model
- Defines call signaling, call transfer, codec negotiation
- Network services such as routing are provided by Gatekeepers
- Direct endpoint or Gatekeeper routed call signaling
- Uses RTP for media
- Learning curve typically steep for developers

## SIP

- IETF Standard
- ASCII text-based
- Leverages other Internet standards such as HTTP, DNS, MIME, etc.
- Defines only session initiation and termination (call signaling)
- Network services provided by SIP servers (Proxy, Redirect, and Registrar)
- Endpoint signaling or “record-route” signaling
- Uses RTP for media
- Lots of developers familiar with the Internet standards

# Typical ITSP H.323 Network

iBasis

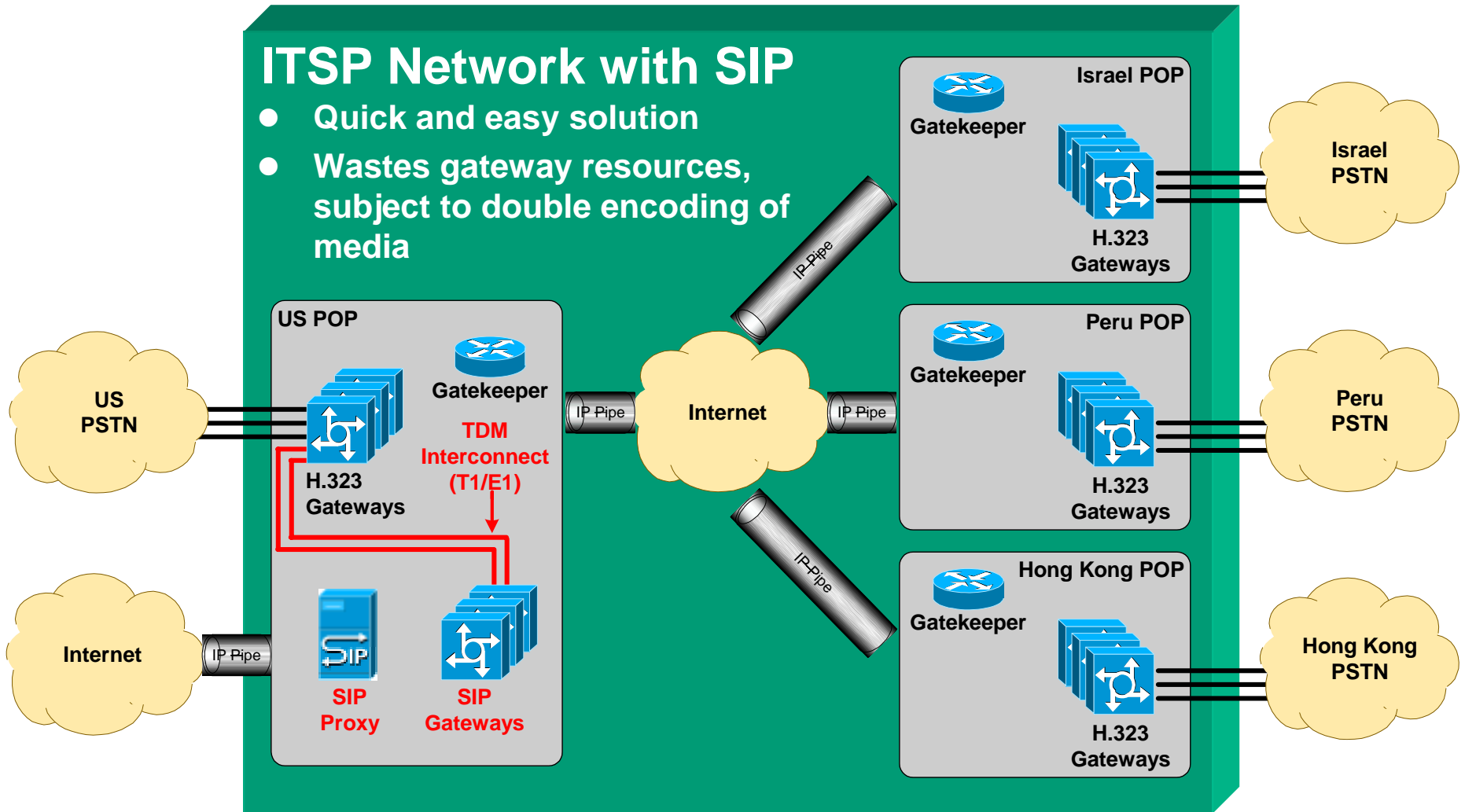


# TDM-based Inter-networking

iBasis

## ITSP Network with SIP

- Quick and easy solution
- Wastes gateway resources, subject to double encoding of media

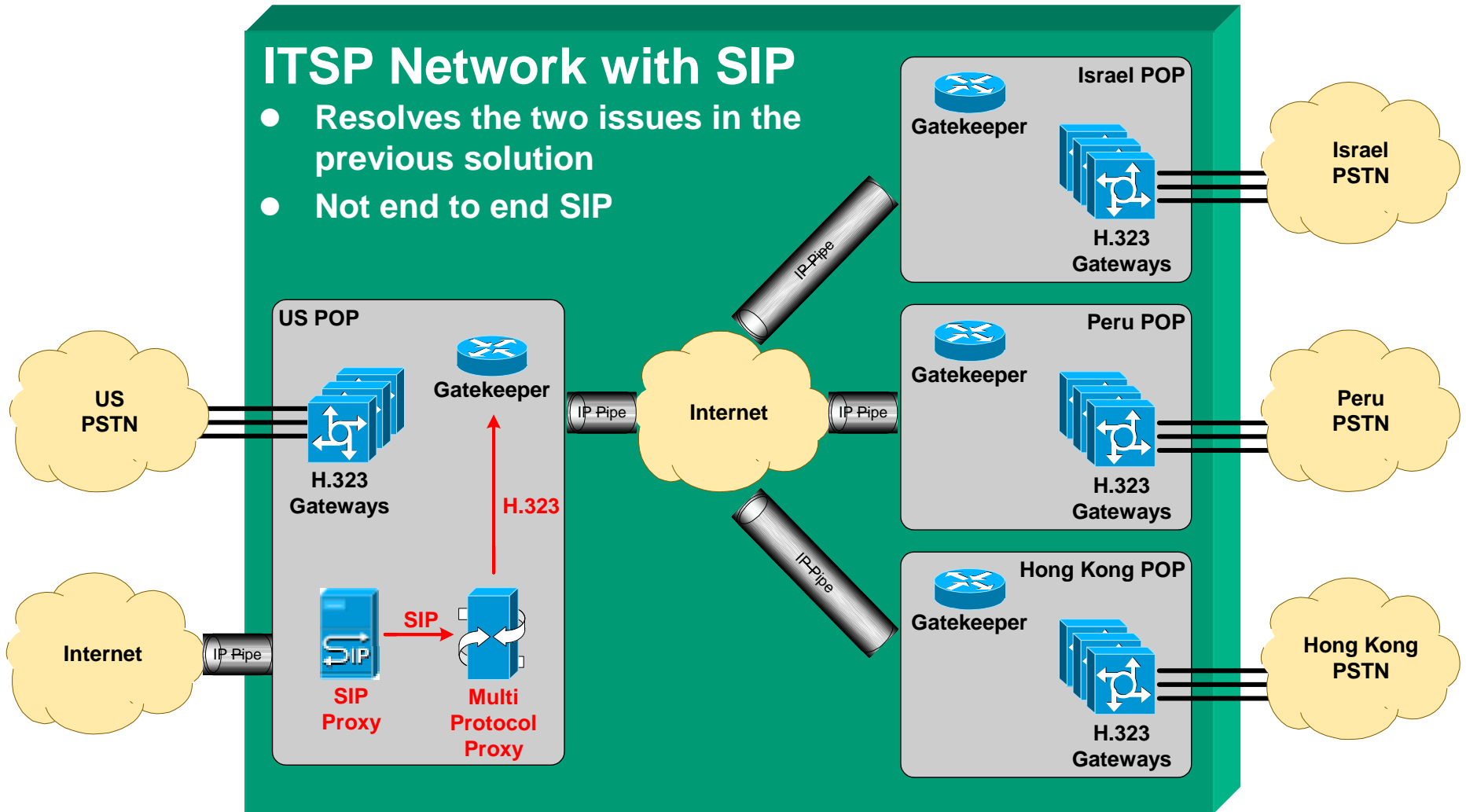


# Multi-protocol Proxy Inter-networking

iBasis

## ITSP Network with SIP

- Resolves the two issues in the previous solution
- Not end to end SIP

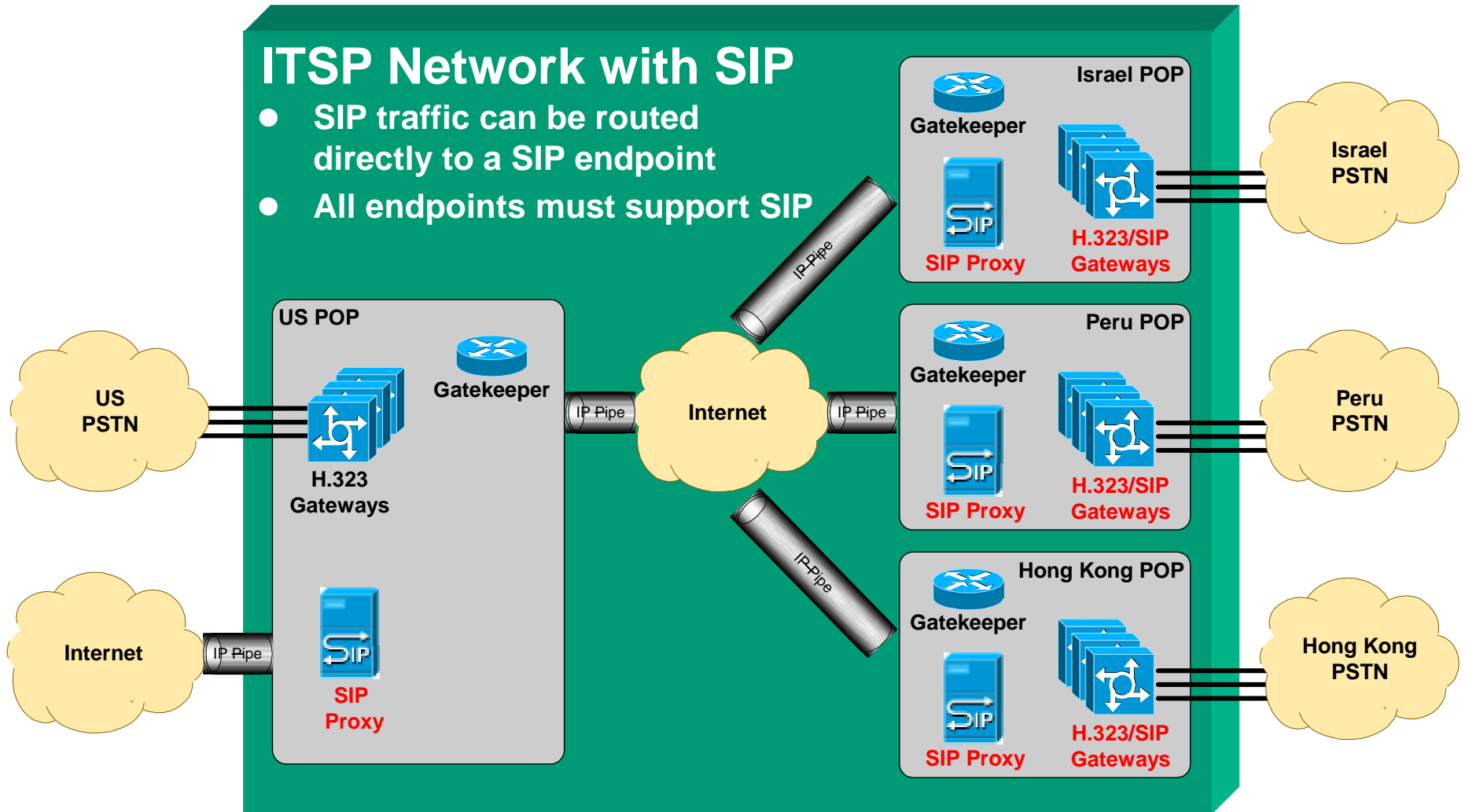


# Dual-stacked Gateway Inter-networking

iBasis

## ITSP Network with SIP

- SIP traffic can be routed directly to a SIP endpoint
- All endpoints must support SIP



# Migration Challenges



- **Operational impact of deploying two protocols**
  - Integration with a centralized routing engine
  - Operability with existing IP-billing systems
  - Training for the support organization
  - Capacity planning
- **Quality of voice and signaling**
  - Post dial delay (PDD)
  - Packet loss, jitter, and latency
- **SIP Proxy servers and Multi-Protocol Proxies need to be robust and high performance**
- **How is the bottom line affected? \$\$\$**
- **Interoperability with other SIP vendors**



# Summary

- **SIP is here**
  - There will be a transition to SIP depending on the new applications that drive the market
  - H.323 is not going to vanish, SIP just enables new applications
- **Several possible H.323/SIP-enabled VoIP architectures**
  - TDM-based internetworking
  - Multi-protocol signaling proxy
  - Dual-stacked gateways
- **Migration will pose several challenges**

# Thank You

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