

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

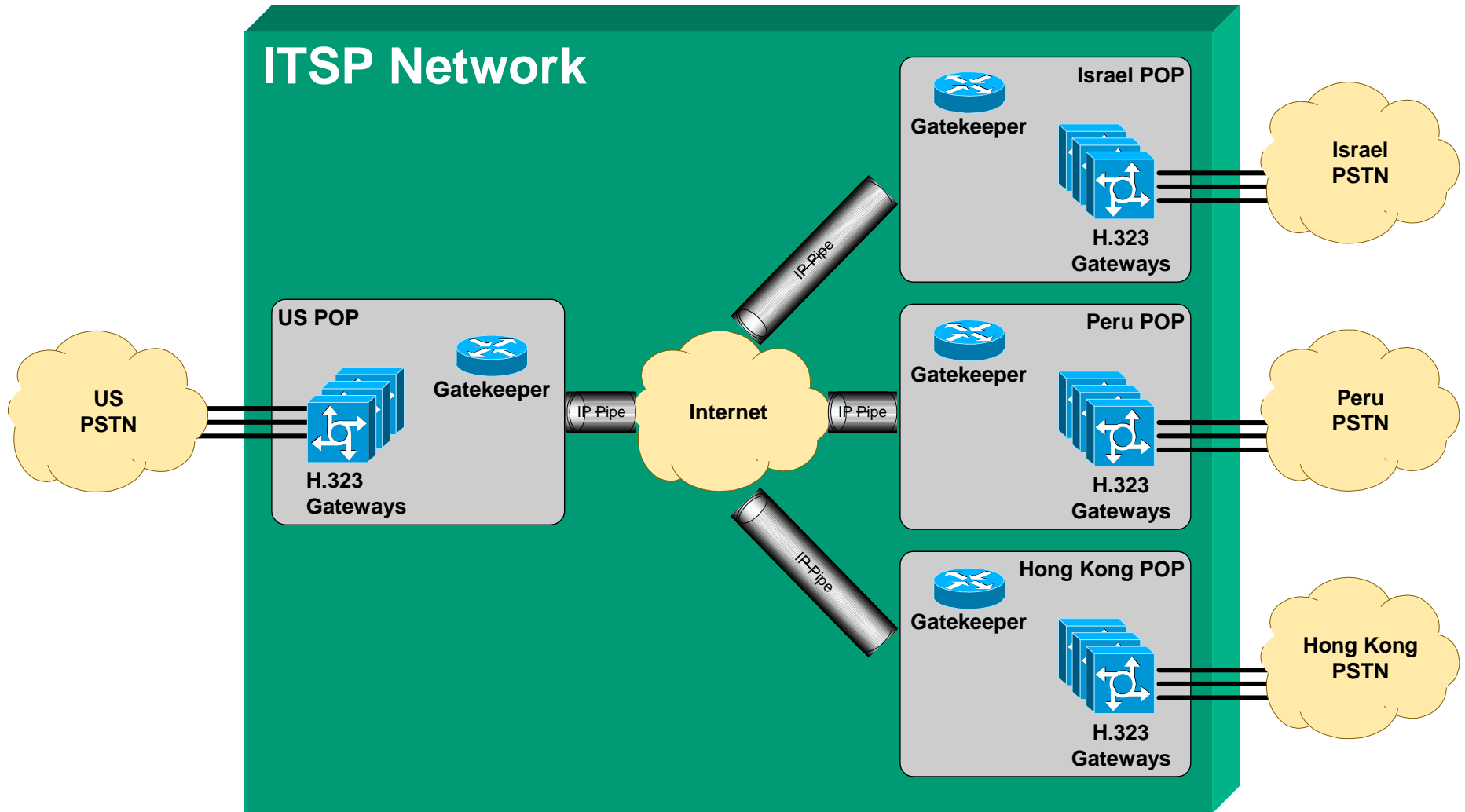
- ITU Standard
- Binary-based ASN.1 encoding
- Utilizes other ITU standards such as ISDN (Q.931) and Q.SIG model
- Defines call signaling, call transfer, codec negotiation
- Network services such as routing are provided by Gatekeepers
- Direct endpoint or Gatekeeper routed call signaling
- Uses RTP for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

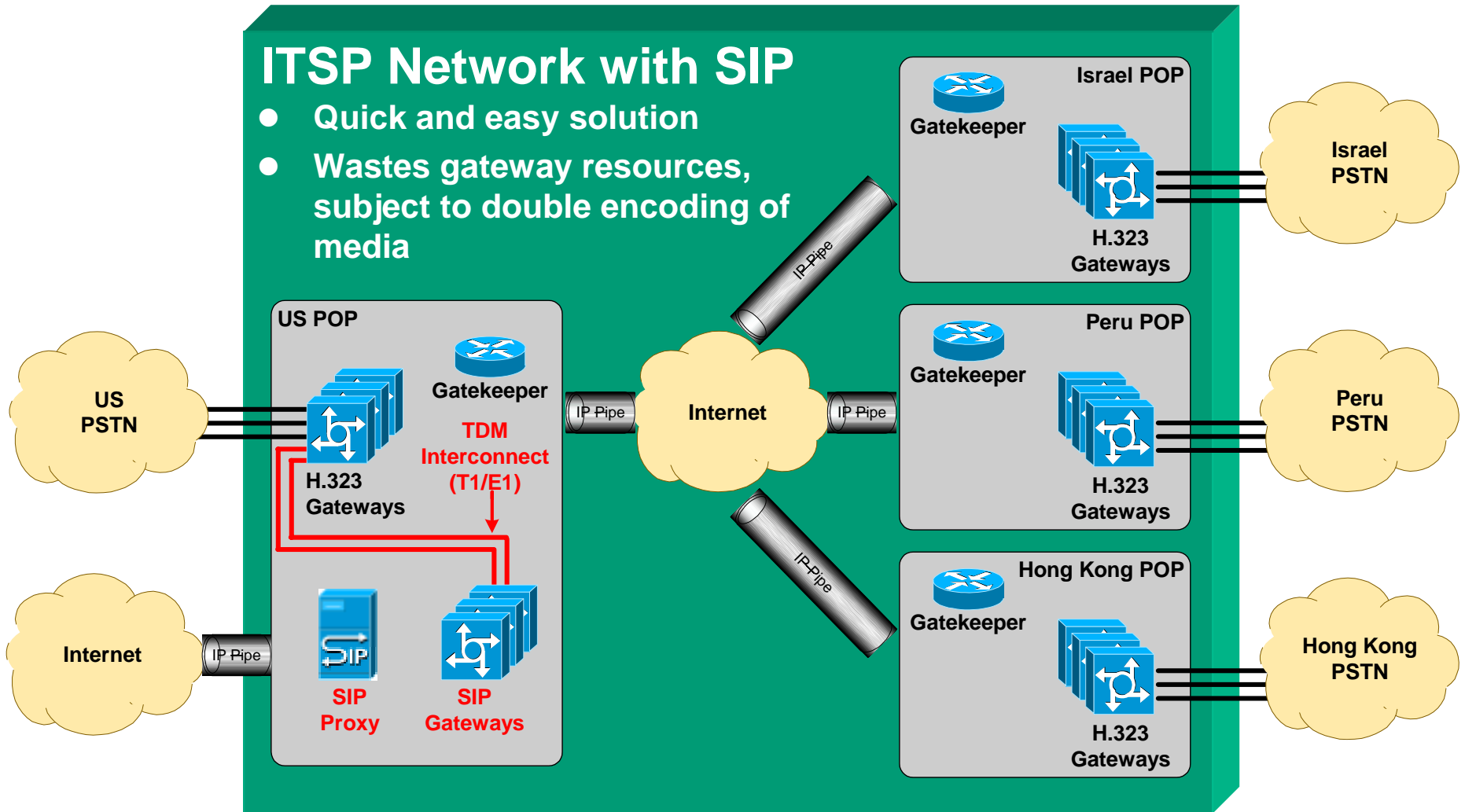


TDM-based Inter-networking

iBasis

ITSP Network with SIP

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

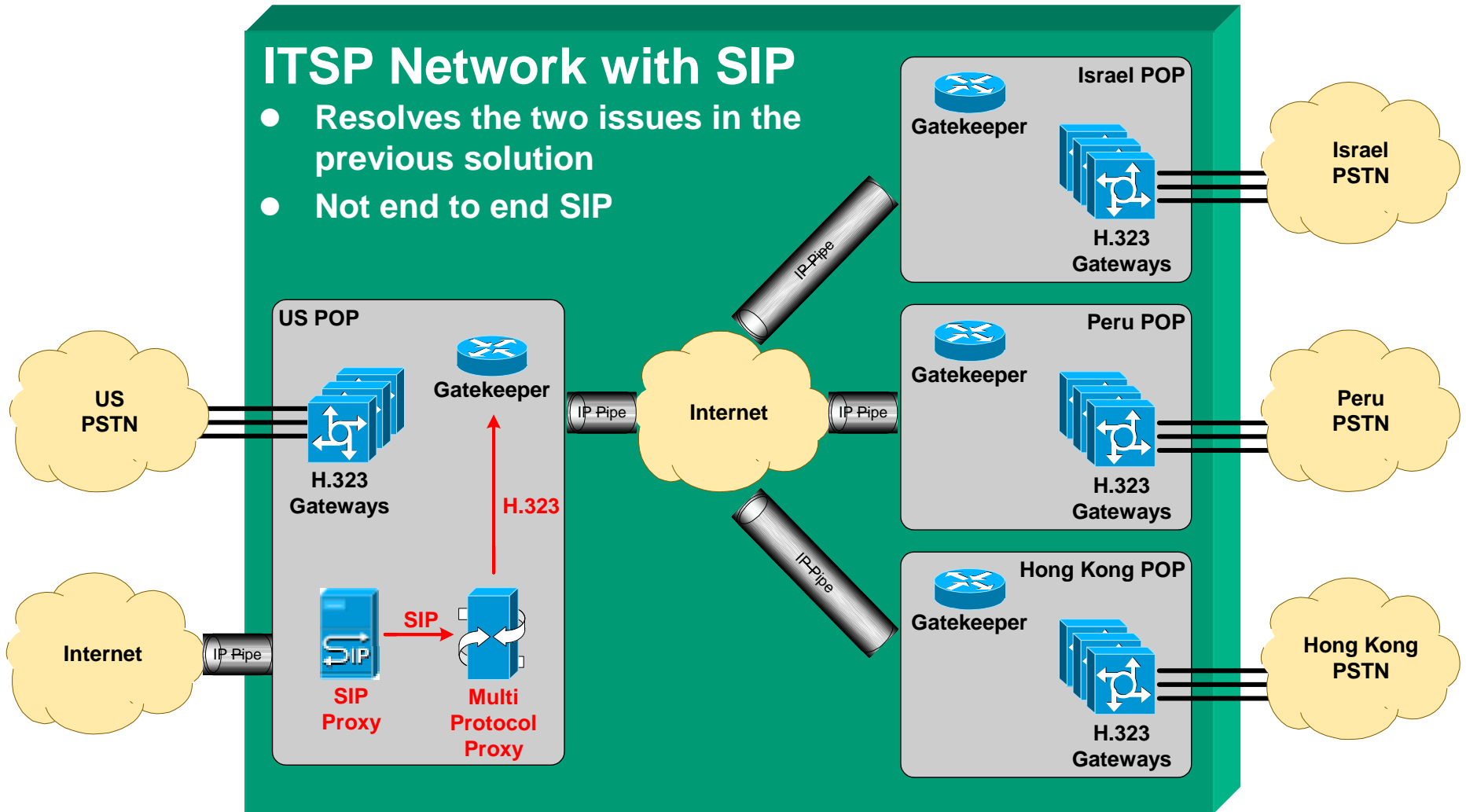


Multi-protocol Proxy Inter-networking

iBasis

ITSP Network with SIP

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

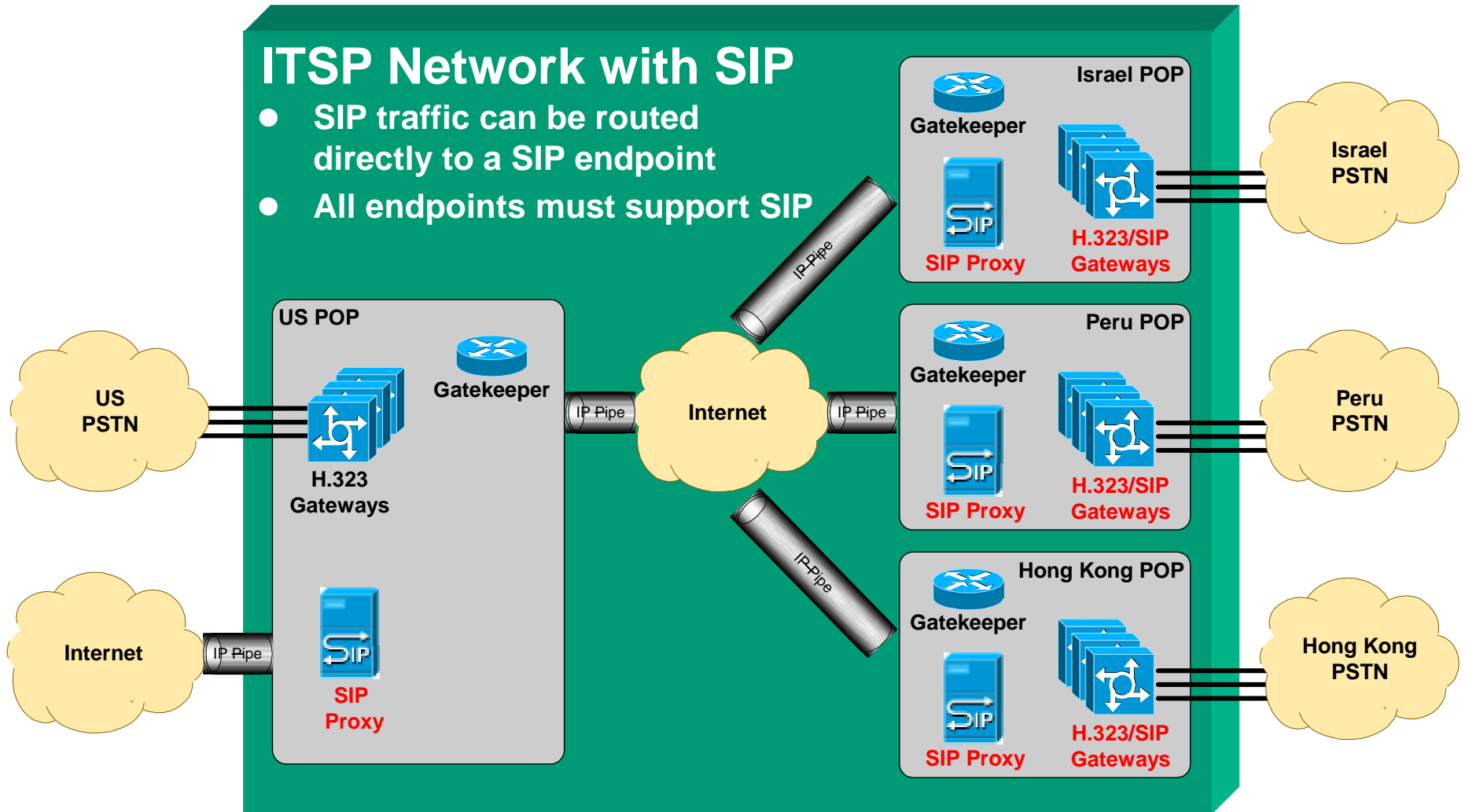


Dual-stacked Gateway Inter-networking

iBasis

ITSP Network with SIP

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



Migration Challenges

iBasis

- **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**

iBasis SIP Self Certification Laboratory



- Test your own SIP clients in our laboratory
 - Register at <http://www.ibasis.net/services/sip.htm>

The screenshot shows the iBasis SIP Self-Certification Lab interface. The page title is "The iBasis SIP Self-Certification Lab" and the subtitle is "CDRs and Protocol Trace Output". The search criteria is "Calling-Party 17815057301". The results are displayed in a table with the following columns: called_station_id, acct_session_id, vclp_setup_time, and vclp_connect_time.

called_station_id	acct_session_id	vclp_setup_time	vclp_connect_time
17815057301-181.177.100	17815057301-181.177.100	2011-08-25 14:46:28	2011-08-25 14:46:30
17815057301-181.177.100	17815057301-181.177.100	2011-08-25 14:46:46	2011-08-25 14:46:50
17815057301-181.177.100	17815057301-181.177.100	2011-08-25 22:20:06	2011-08-25 22:20:27
17815057301-181.177.100	17815057301-181.177.100	2011-08-25 22:21:14	2011-08-25 22:21:30
17815057301-181.177.100	17815057301-181.177.100	2011-08-25 22:31:05	2011-08-25 22:31:52
17815057301-181.177.100	17815057301-181.177.100	2011-08-25 22:32:28	2011-08-25 22:32:53
17815057301-181.177.100	17815057301-181.177.100	2011-08-25 22:34:16	2011-08-25 22:34:26
17815057301-181.177.100	17815057301-181.177.100	2011-08-25 22:46:18	2011-08-25 22:46:21
17815057301-181.177.100	17815057301-181.177.100	2011-08-27 16:31:07	2011-08-27 16:31:40
17815057301-181.177.100	17815057301-181.177.100	2011-08-28 16:21:14	2011-08-28 16:21:40

The screenshot shows the iBasis SIP Self-Certification Lab interface. The page title is "The iBasis SIP Self-Certification Lab" and the subtitle is "CDRs and Protocol Trace Output". The search criteria is "Call-ID c3943000-15329-16382d6-2e353032@205.181.177.209". The results are displayed in a text area with the following content:

```
log_message
[Time: Tue 23 16:46:30 2011] Received SIP packet from 205.181.177.209:5060
INVITE sip:17815057301@181.177.100 SIP/2.0
Via: SIP/2.0/UDP 205.181.177.209:5060
From: *Anonymous* <sip:17815057301@181.177.100>
To: <sip:17815057301@181.177.100>
Call-ID: c3943000-15329-16382d6-2e353032@205.181.177.209
CSeq: 101 INVITE
Expires: 300
User-Agent: Cisco IP Phone/8.0.2 SIP-ED/4000
Accept: application/edp
Contact: sip:17815057301@181.177.100:5060
Content-Type: application/edp
Content-Length: 337

INVITE
c3943000-15329-16382d6-2e353032@205.181.177.209
R=181.177.100
r=0
m=audio 17815057301/205.181.177.100
mptime:0.0000
mrtt:0.0000
mrttvar:0.0000
mrttmin:0.0000
mrttmax:0.0000
```

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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