Future of VoIP

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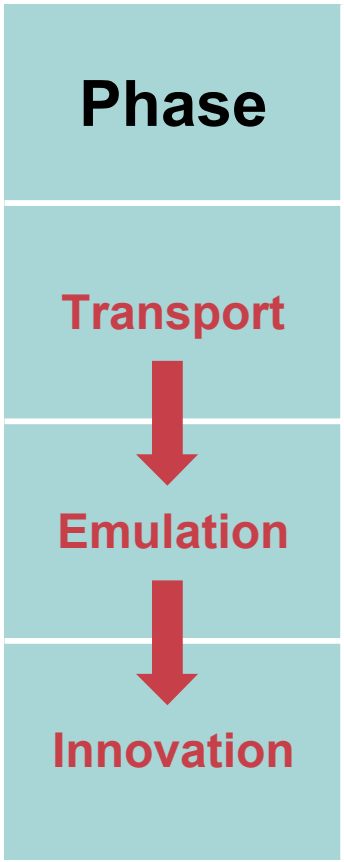
Evolution of VoIP



Views on VoIP Signaling

- **Cynic**

  First we had H.323, then we had MGCP, then we had Megaco, Cisco's got SCCP, what a mess*

  This VoIP stuff is going to take years to get to the point it works well enough

- **Pragmatist**

  I really like the consolidated network, but all my PBX features have to work

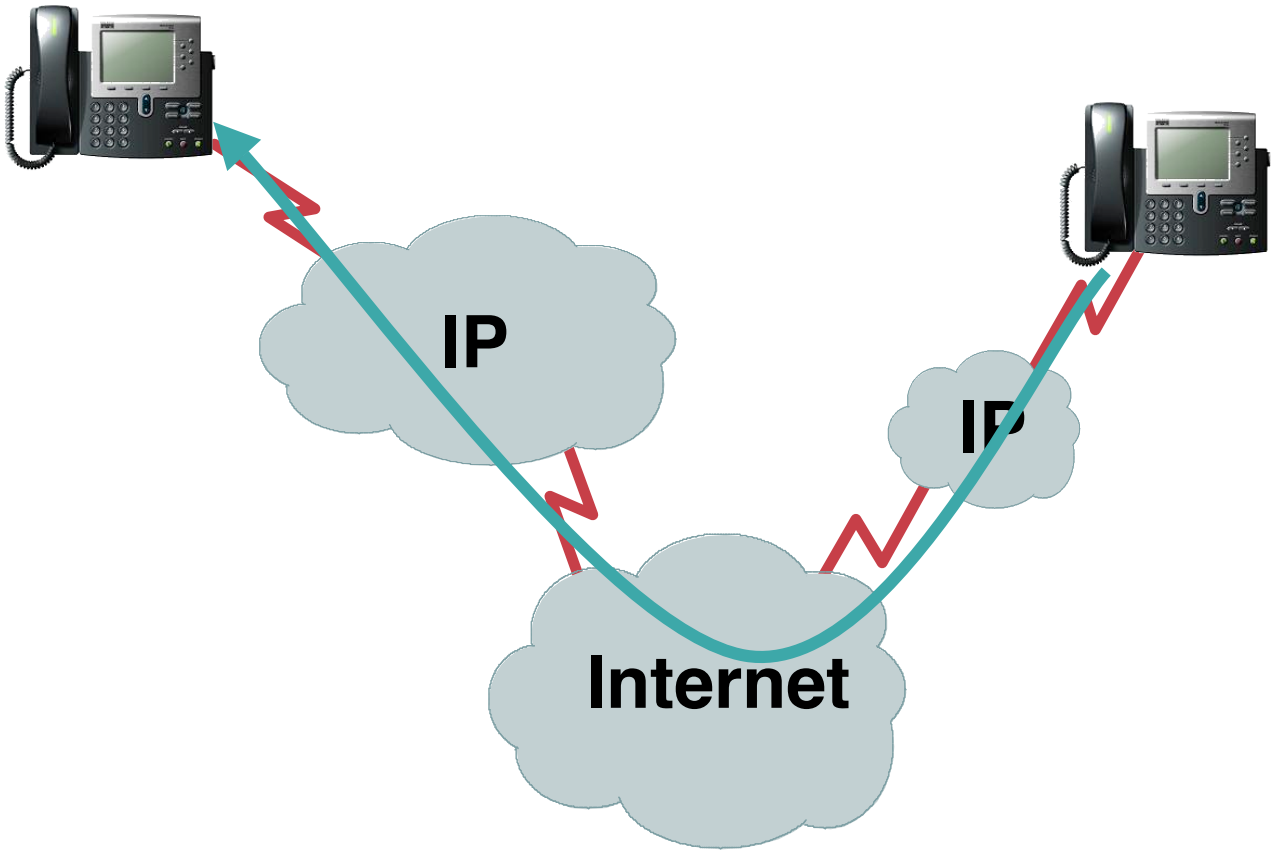
  The phones have to work; everything else is secondary

- **Wild-eyed radical**

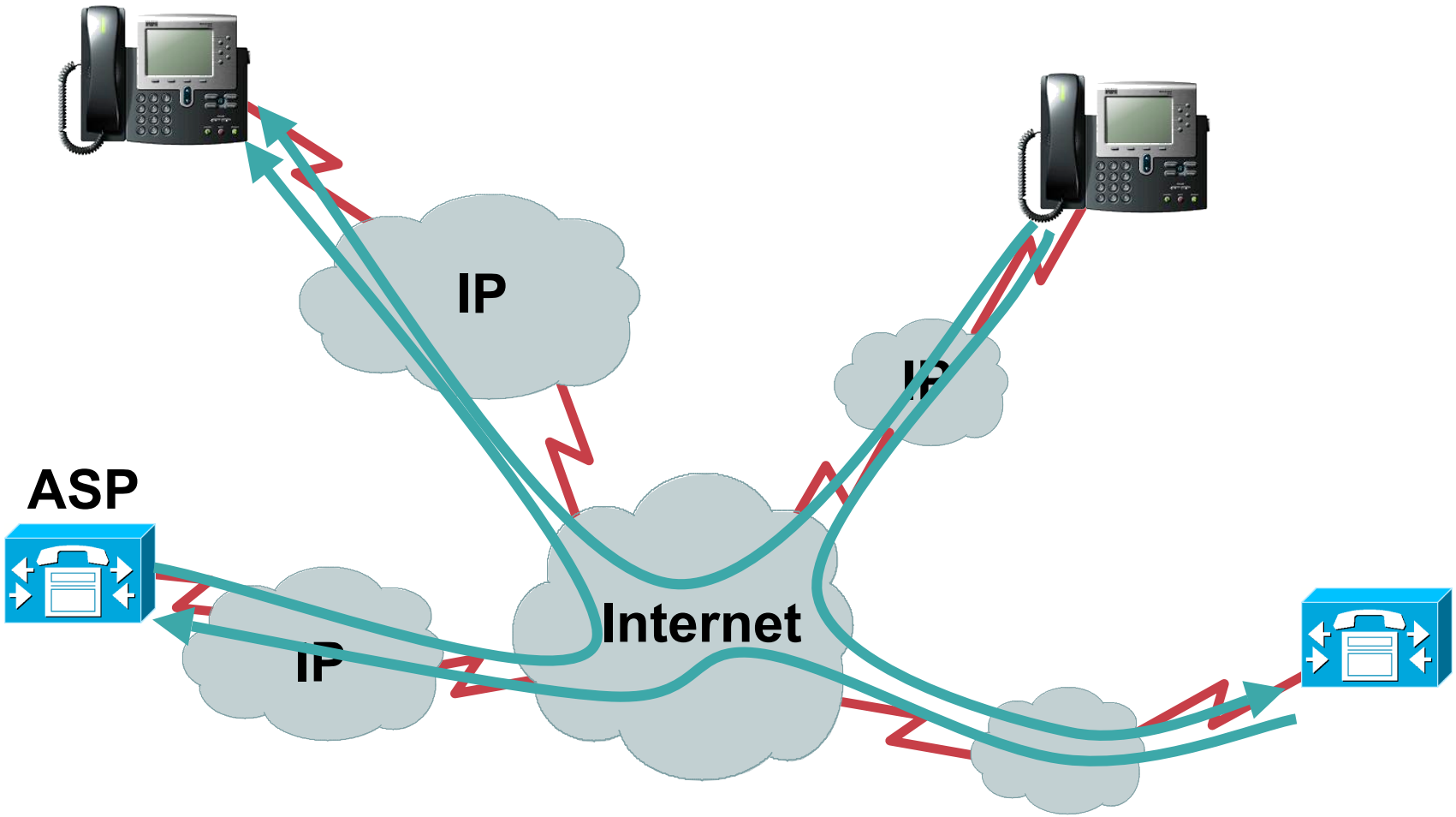
  Uh, what's a phone?

*** Yes, the intention is to show acronyms you might not understand, because the number of acronyms is part of the problem**

Voice over Internet

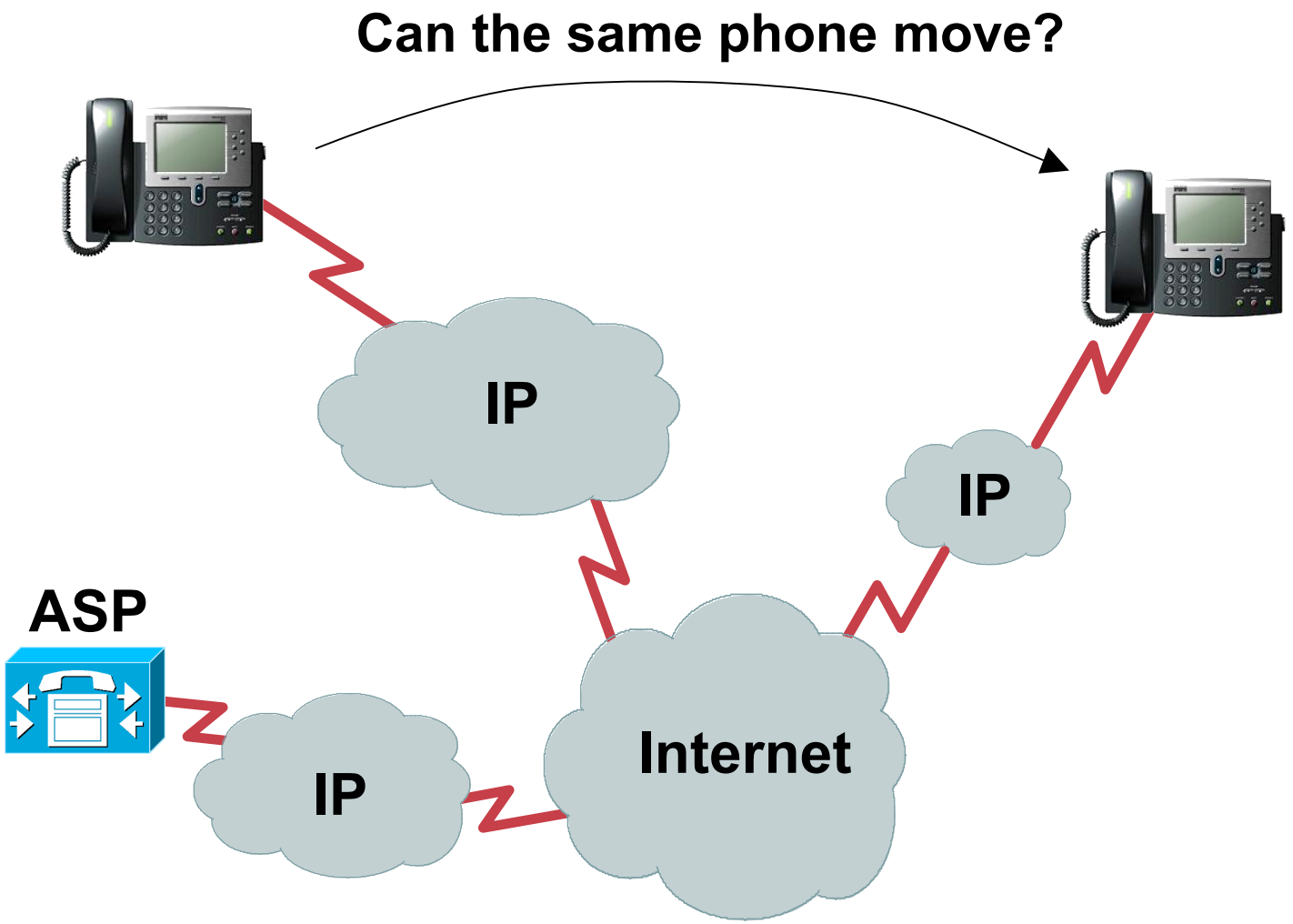


In reality we have more players



Application Service Provider ≠ Internet Service Provider

Nomadic VoIP?



All VoIP is nomadic

- The only question is how far a phone can move
  - In some networks, the phone can not move even inside an apartment
  - In other networks, the phone can move all over the Internet
- How far a phone can move is up to the VoIP provider, and his (potential) arrangement with the ISP
  - In some cases the ISP and VoIP provider is the same organization
  - In other cases, the ISP is unaware of the existence of VoIP in his network (and he doesn't care)

Implications on regulation

- Current regulation take for granted a voice provider have control both over the physical and logical infrastructure
- Further, it doesn't take into account a phone can move
  - Already cellphone usage is problematic (emergeny calls)
- VoIP (Internet in general) force a differentiation on what regulation is on a service provider and what is requirement(s) on the ISP

How Many Business Cards Do You Have?

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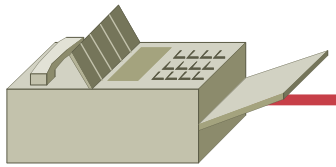
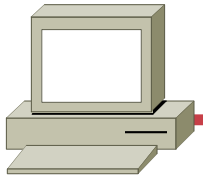
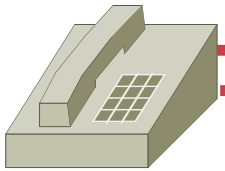
Today, Many Addresses







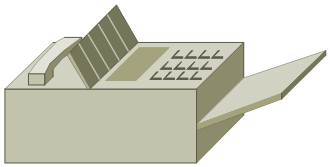
tel:+46-8-971234

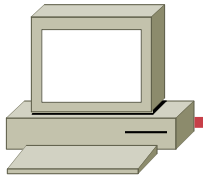
mailto:paf@example.com

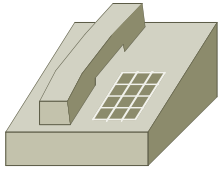
tel:+46-706051234

sip:paf@example.com

With ENUM, Only One







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ENUM

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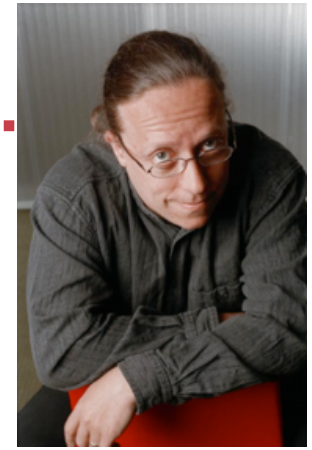


Give one number
to friends:
+46-8-971234

Trials

Delegated Codes	
+31	+246
+33	+247
+36	+290
+40	+358
+41	+374
+43	+420
+44	+421
+46	+423
+48	+882 34
+49	+878 10
+55	+971
+65	
+86	

Trials	
Austria	Japan
China	Finland
Sweden	France
Taiwan	Ireland
UK	…
Norway	
Poland	
Germany	
Slovakia	
South Korea	

Future of addressing

- If we have SIP based VoIP, the addresses in SIP look very much like an email address
- Is not the goal for people to have something like this:

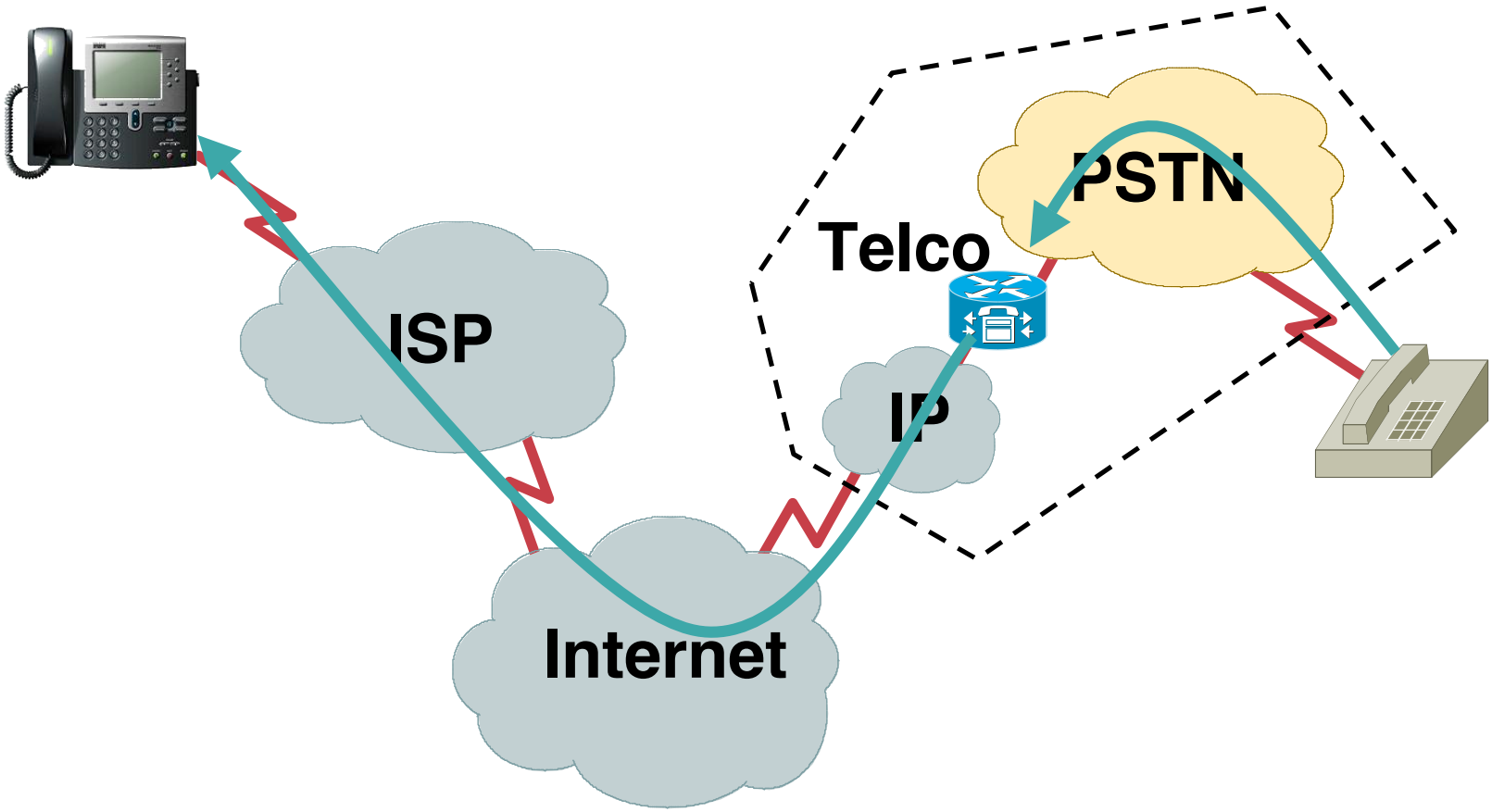
  http://paf.se/

  mailto:paf@paf.se

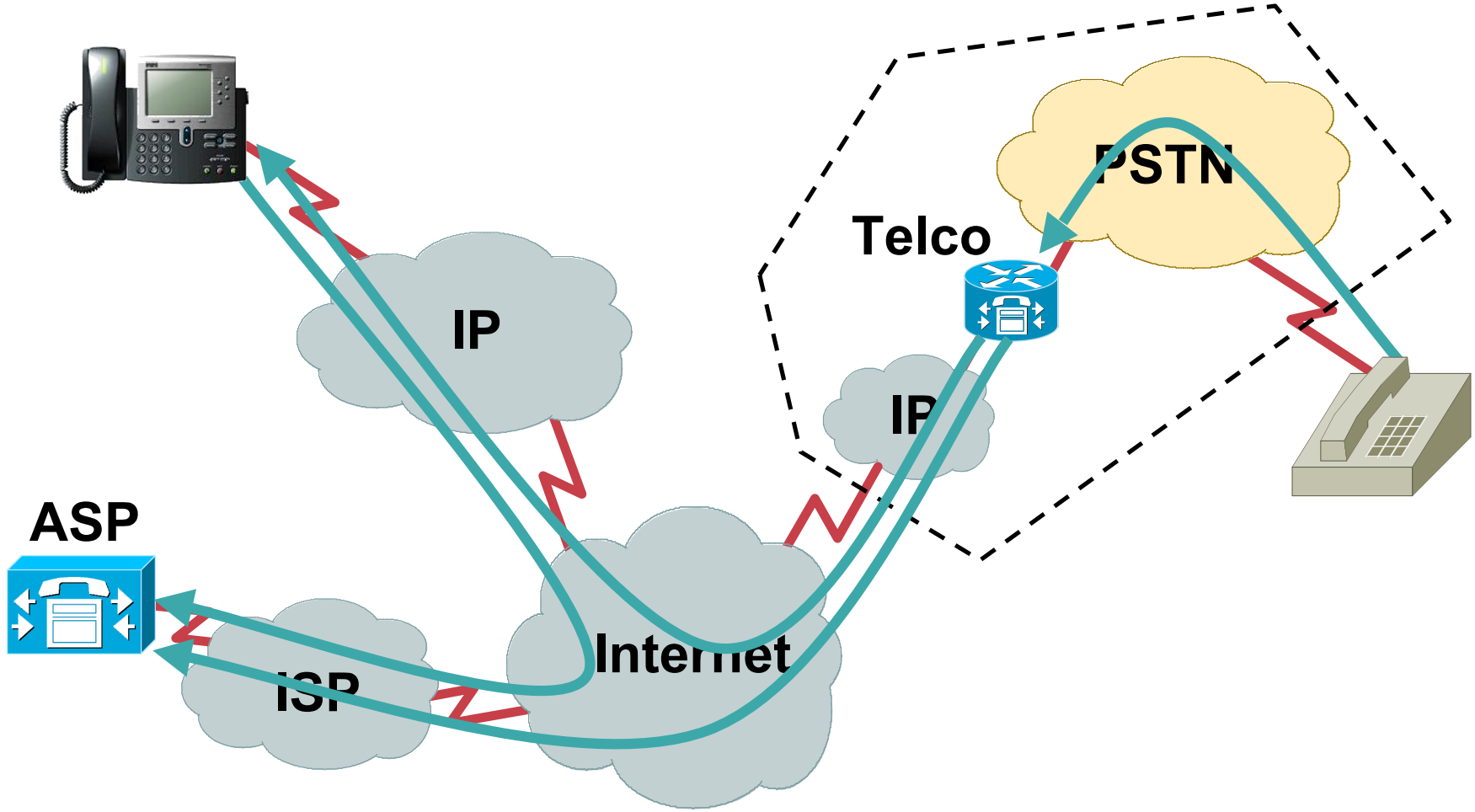
  sip:paf@paf.se

  jabber:paf@paf.se
- If so, why use E.164 numbers?

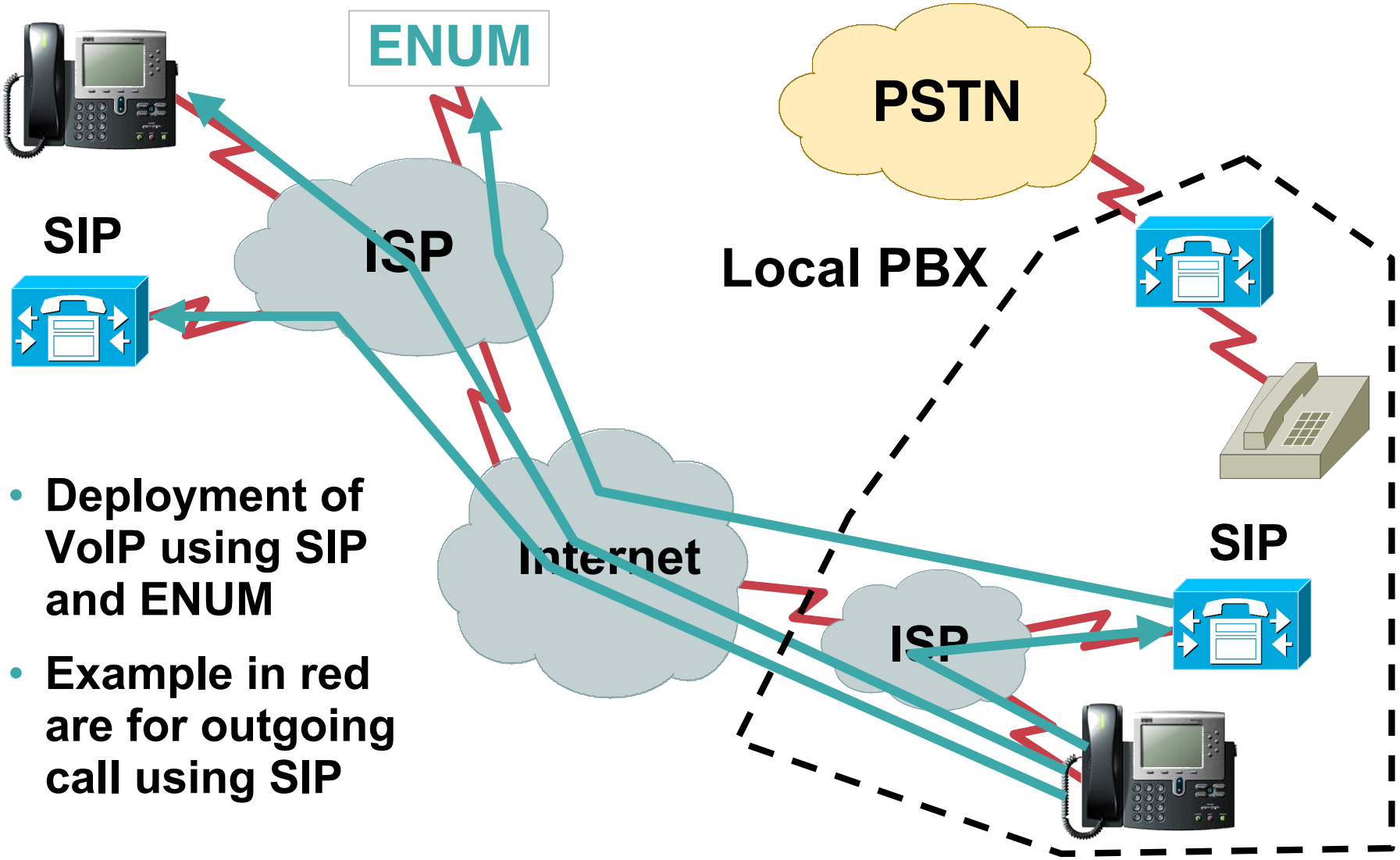
Call from traditional telephony



Call from traditional telephony

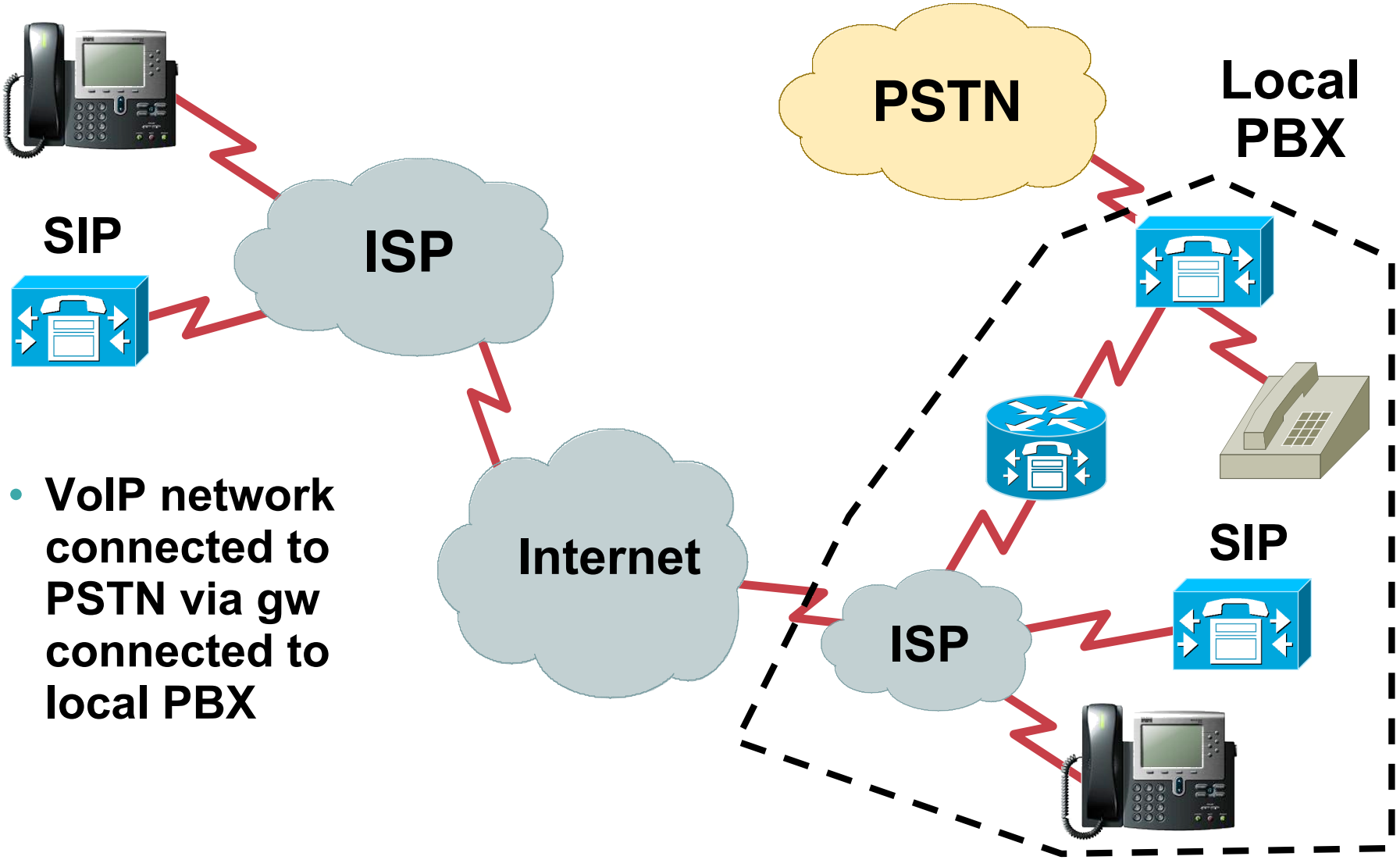


ENUM/VoIP Transition (1)



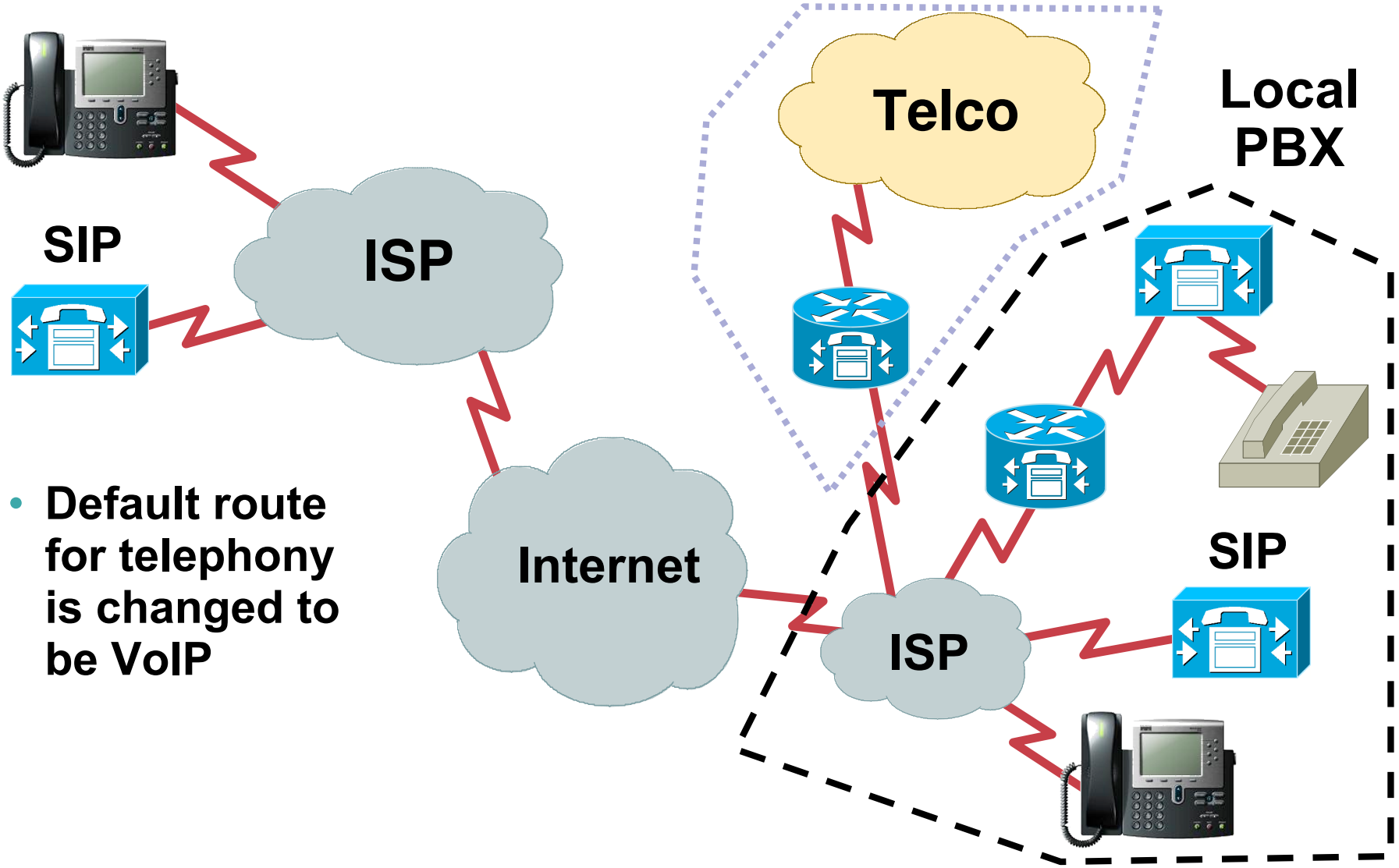
- Deployment of VoIP using SIP and ENUM
- Example in red are for outgoing call using SIP

ENUM/VoIP Transition (2)



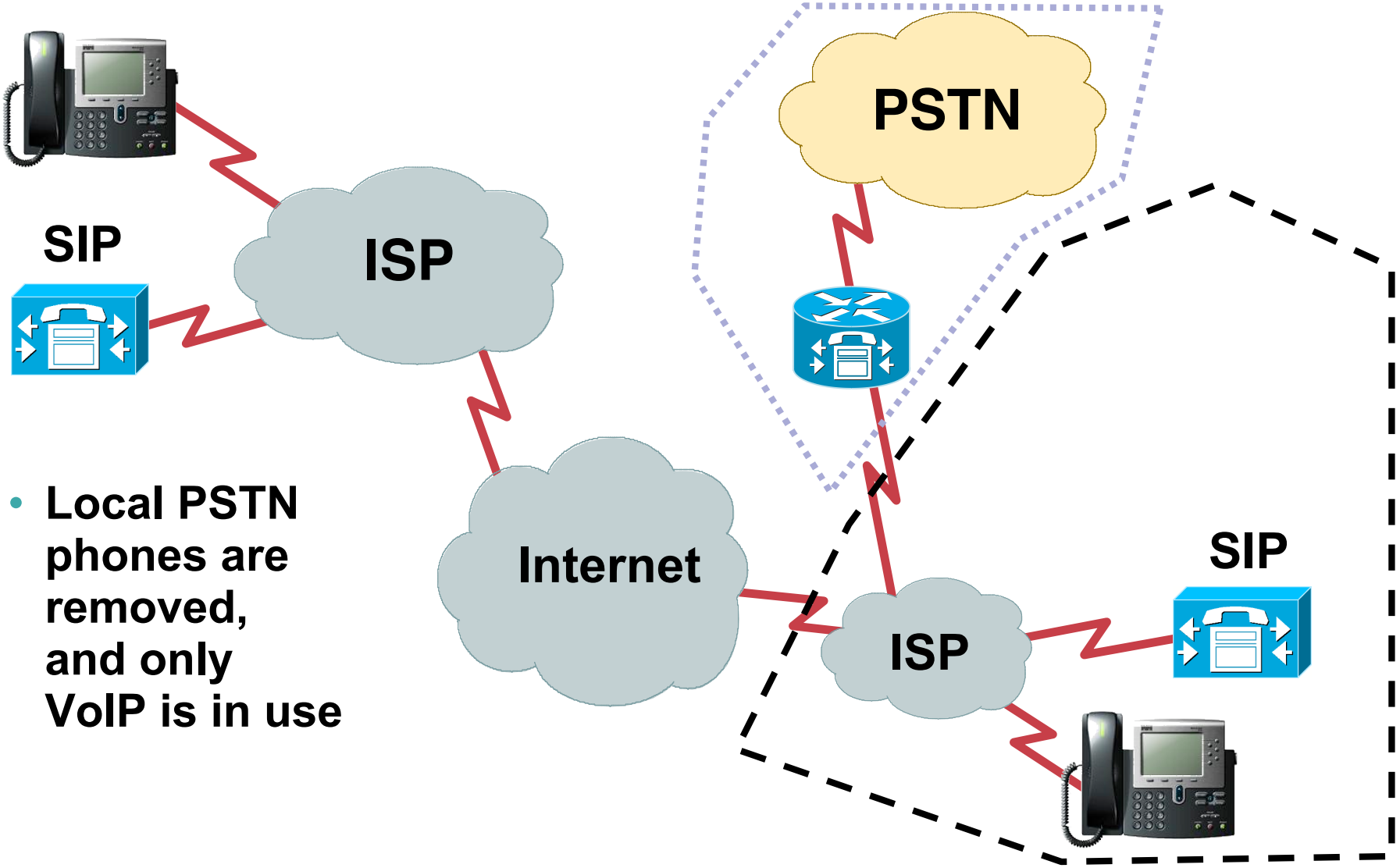
- VoIP network connected to PSTN via gw connected to local PBX

ENUM/VoIP Transition (3)



- Default route for telephony is changed to be VoIP

ENUM/VoIP Transition (4)



- Local PSTN phones are removed, and only VoIP is in use

What different kinds of VoIP exists?

- Enterprise have IP based PBX
  - Might or might not use VoIP internally
  - Might or might not route calls externally using VoIP
- Providers running VoIP services
  - Works very much like electronic mail
  - Anyone with “always connected” status to Internet can have their own VoIP server, just like their own email server
- VoIP telephony
  - Adapter that convert from analog telephony to VoIP
  - Software running on a computer
  - Hardware VoIP phone
- Multiple protocols
  - Vendor specific
  - SIP (Session Initiation Protocol)

One large problem, emergency calls

- Routing the call to the correct emergency center
- Identification of the caller
- Localization of where caller is

What is the goal?

- When a call is received at some call center, the receiver of the call want to know where the call originates from

What is the problem?

- The receiver of the call (only) get the IP address of the peer of the VoIP flow
- As the application service provider is not (always) the Internet Service Provider, he doesn't know where the IP address is (geographically) where the call originates from

What can the solution be?

- Require Internet Service Providers to keep track of where every IP address is
- Require Application Service Providers to know where the phones are
- Routing the call to (just) the closest emergency center

- Both have pros and cons

(A) ISP know where IP address is

- **“When call arrives at call center, call center query ISP’s for location of IP address”**
- **Plus:**
  - **We might need this information in other applications**
  - **ISP’s might know this already (no changes)**
- **Minus:**
  - **The IP flow might be “forwarded” from another location**
  - **The IP address might be the egress point of the network of an Enterprise**

(B) ASP know where phone is

- **“When call arrives to call center, the location of the caller is part of the signalling”**
- **Plus:**

  The location of the phone is independent of the network topology

- **Minus:**

  In reality, the phone have to know where it is, which imply we need to change all phones

(C) Routing to closest emergency center

- **“A call is to be routed according to network topology where the client is (not VoIP server)”**
- **Plus:**

  A call will always reach the closest emergency center

- **Minus:**

  Emergency center ingress points have to exist “everywhere” on the Internet, for example connected to all ISP’s in a given area

  In most cases, the VoIP server is doing the call routing, and not the VoIP client -- and information on where the phone is geographically will still be needed

  How to implement this is technically complicated

Current design goals

- The phone get its location from built-in mechanisms (GPS, manual configuration,…) or from ISP (DHCP or other auto-configuration) which tell the phone where it is
- The phone have to tell the receiver of the call where the call originates from
- Receiver of the call must accept that “it is unknown where the call is from”

Plus and minus

- **Plus**
  - Convergence done the correct way will save money
  - Cost for services will go down
  - More features
    - What's the difference between a phone call and chat with added voice? What's the difference between email and instant messaging?
- **Minus**
  - Convergence done the wrong way will cost money
  - Many companies make money on termination fees
  - Traditional view on regulation doesn't work

Questions?

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