Phone Bootup

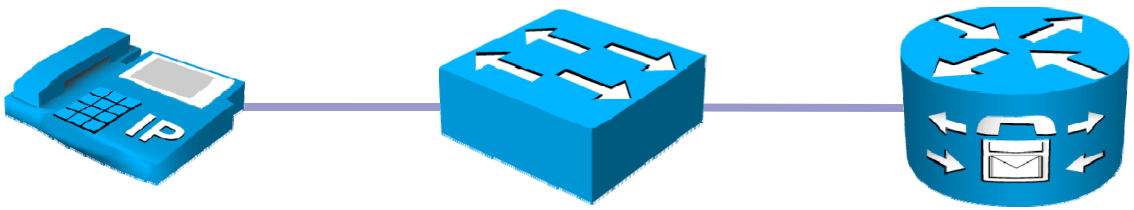
The IP phone powers on.

The phone performs a POST.

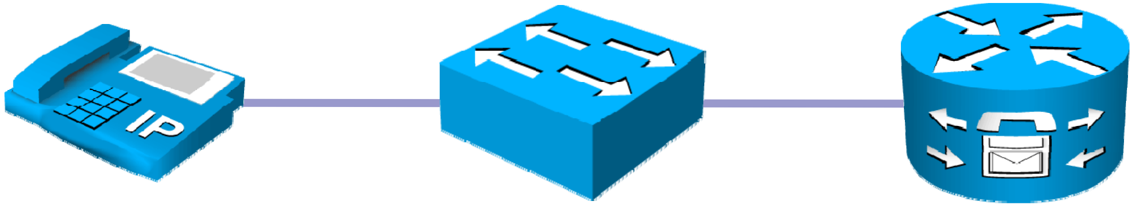
The phone boots.

The phone uses Cisco Discovery Protocol to learn the voice VLAN.

The phone initializes the IP stack.



Phone Bootup (Cont.)



The IP phone sends a broadcast requesting an IP address.

The DHCP server selects a free IP address from the pool and sends it, along with the other parameters, including option 150.

The IP phone initializes, applying the IP configuration to the IP stack.

The IP phone requests a configuration file from the TFTP server defined in option 150.

The configuration file will contain the IP address of the call agent to register to.

Network Time Protocol

- Correct clock synchronization is important for performance, troubleshooting, and CDRs.
- Each Cisco device has an internal system clock that can be set from a number of sources, such as an internal calendar system and NTP.
- NTP allows network devices to synchronize to a clock master.
- The local NTP server can have an attached clock or can synchronize with a more authoritative source.
- There are free NTP servers available on the Internet.

Network Time Protocol (Cont.)

- The IP phone gets its displayed time from the call control platform to which is registers.
  - Cisco Unified Communications Manager
  - Cisco Unified Communications Manager Express
- The time of the internal clock of the Cisco Unified Communications call control platform should be synchronized with an NTP server.
- The time of the Cisco Unified Communications call control platform is used to stamp all syslog and trace messages.

Introducing Cisco Unified Communications Manager Express (CME)

CME Key Features and Benefits

- Supports deployments of up to 240 phones on a single router
- Extends capabilities to the small office that were previously available only to larger enterprises
- Is based on Cisco IOS Software
- Can be administered by GUI or CLI

Supported Platforms

- Cisco Unified Communications Manager Express supports these Cisco platforms:
  - Cisco 2800 & 2900 Series Integrated Services Routers
  - Cisco 3800 & 3900 Series Integrated Services Routers
  - Cisco Unified Communications 500 Series for Small Business

Examples of Cisco Unified IP Phones

Cisco Unified
IP Phone
7942G

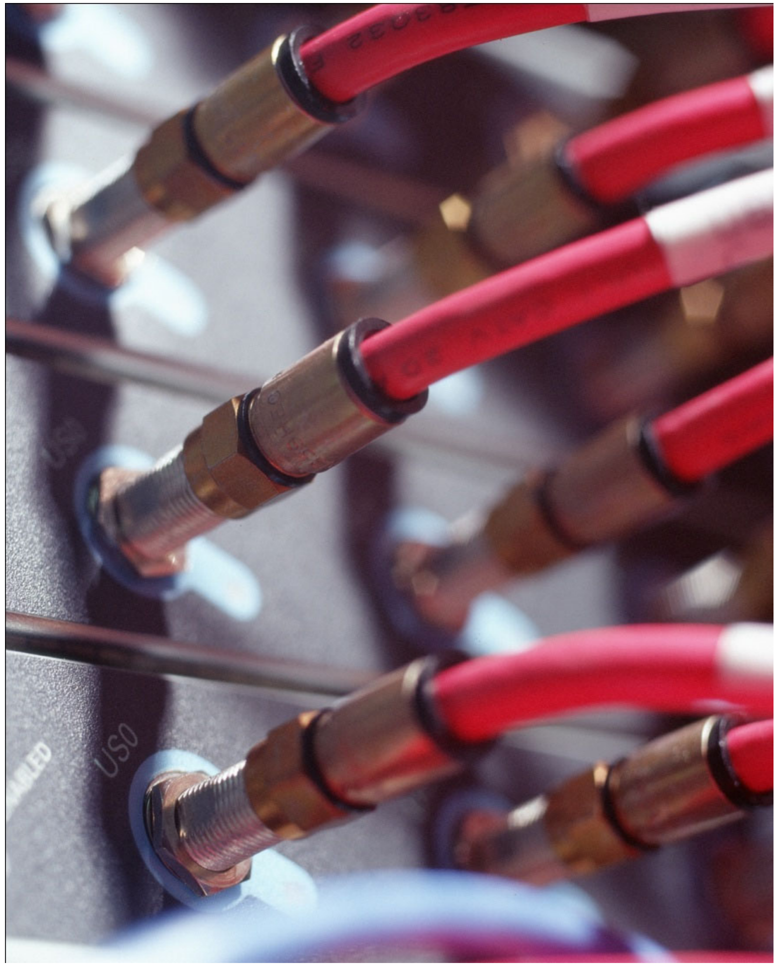
Cisco ATA 186
and 188

Cisco Unified
Wireless IP
Phone 7920

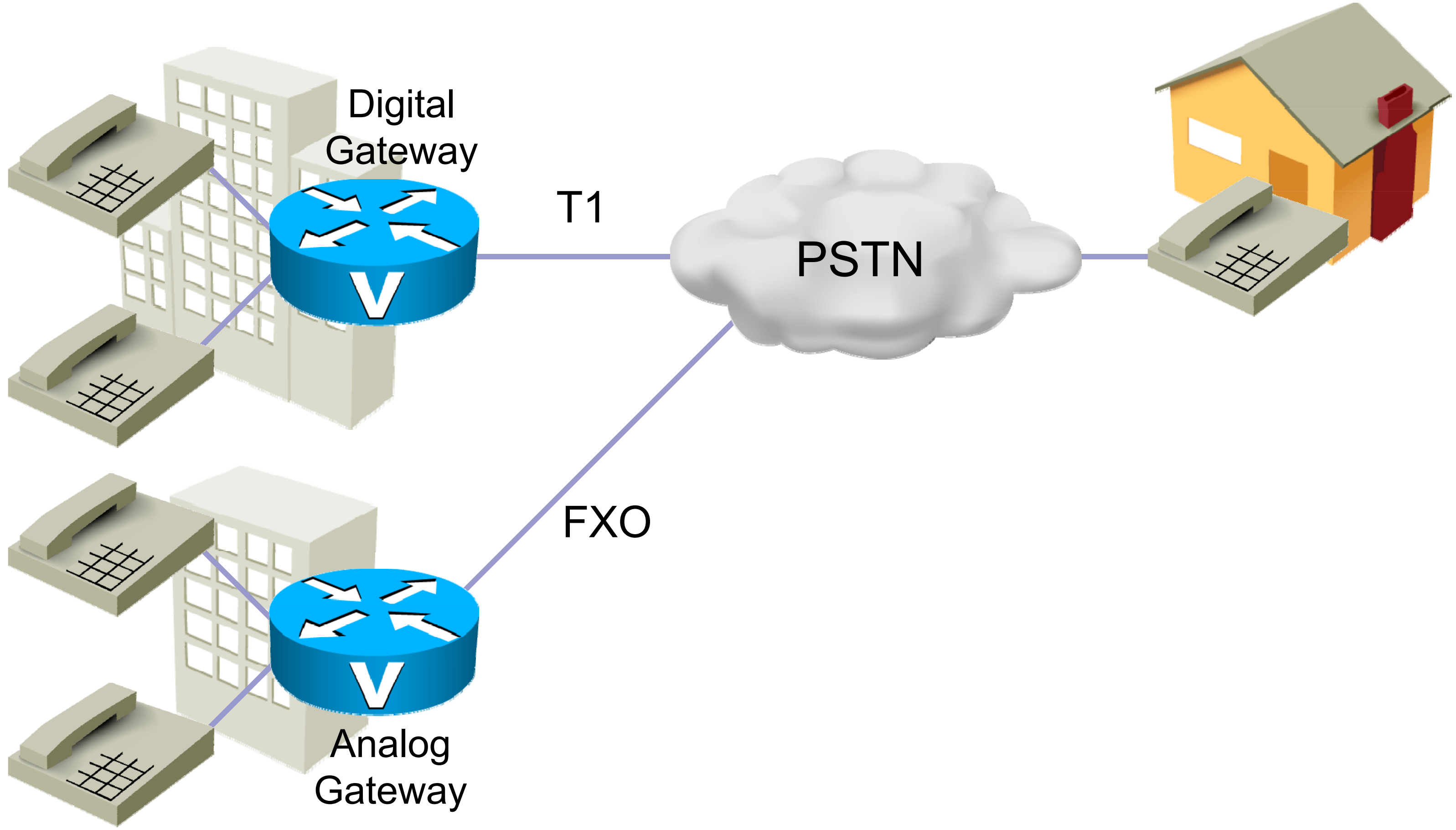
Cisco Unified IP
Phone 7962G

Gateways

- Translate between different networks
- Require DSP resources to perform the translation
- Can be analog gateways:
  - Analog station gateways
  - Analog trunk gateways
- Can be digital gateways

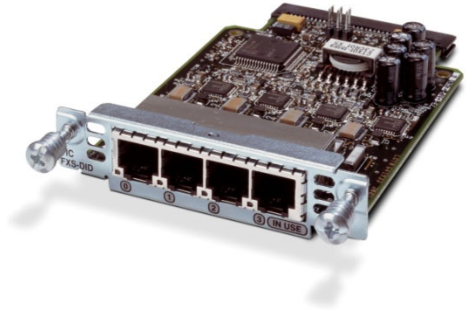


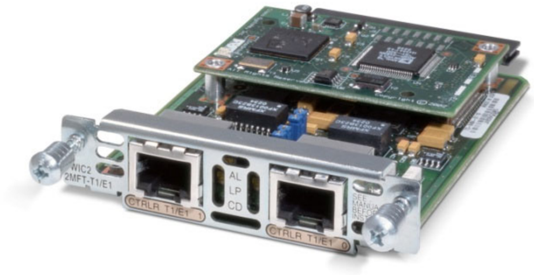
Gateway Function—Example on Cisco Unified Communications Manager Express



Voice Ports

- Analog ports
  - FXS
  - FXO
- Digital ports
  - CAS T1/E1
  - PRI T1/E1
  - BRI





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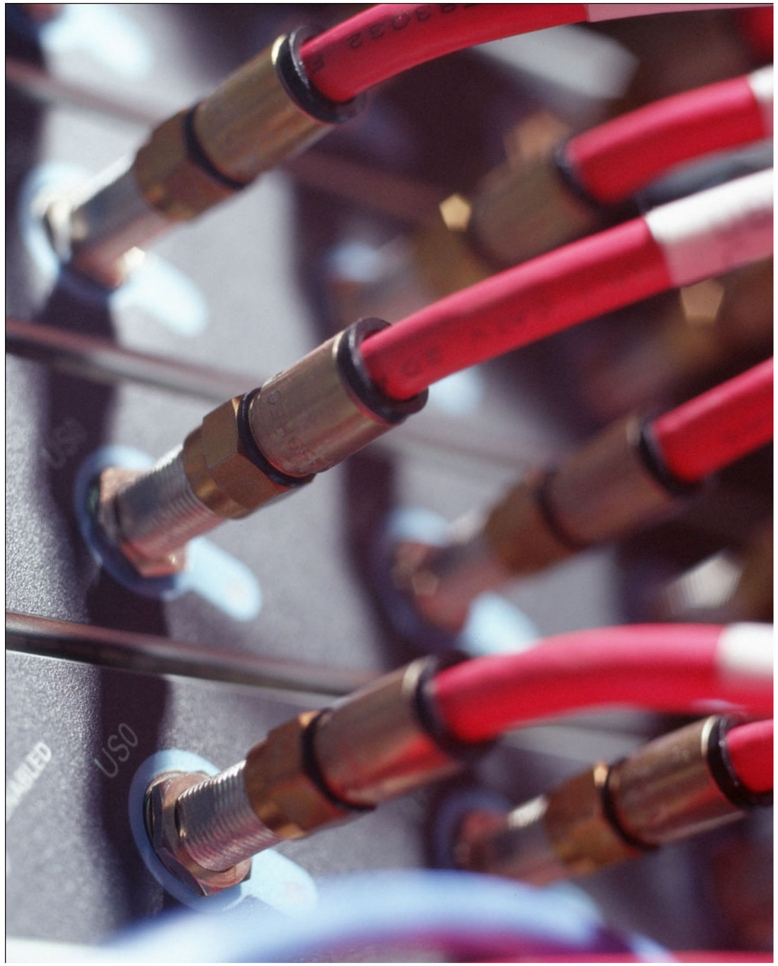

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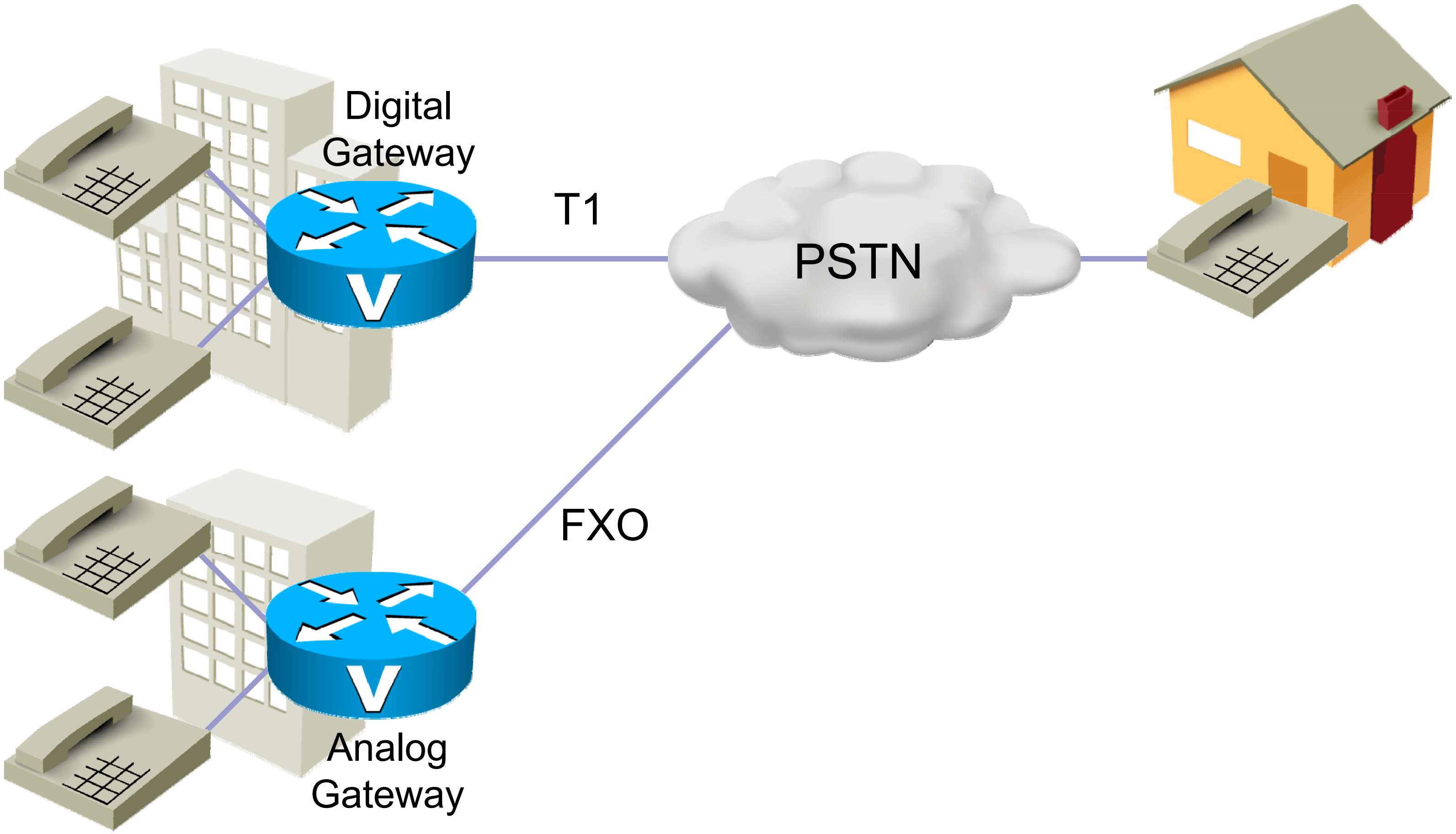

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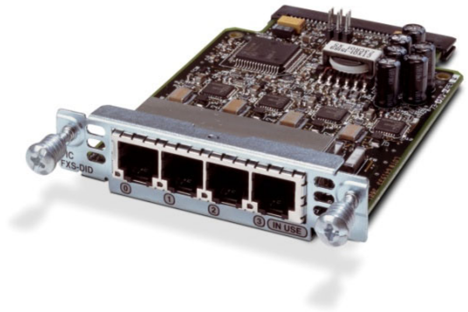
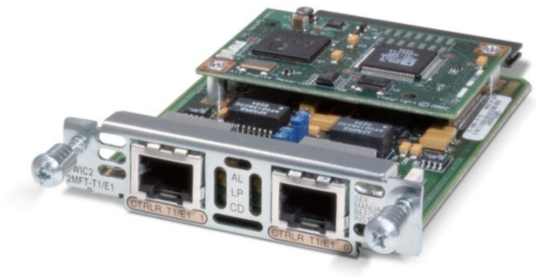
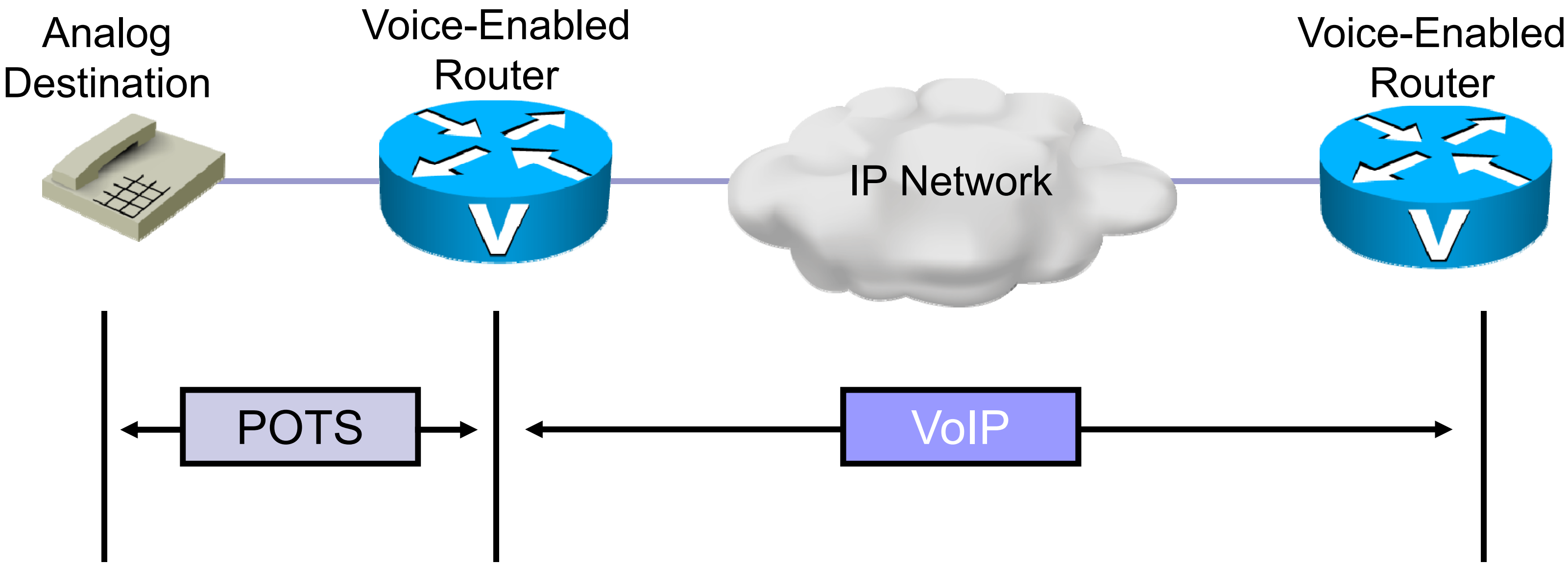


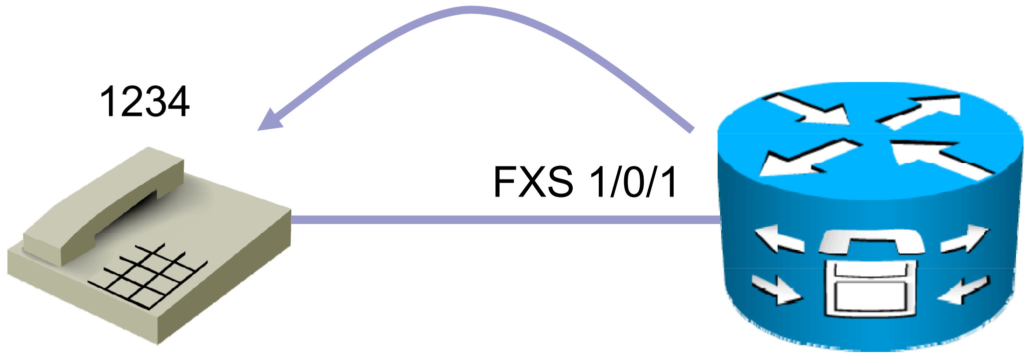


Dial Peers (Cont.)



You create dial peers using the CLI.

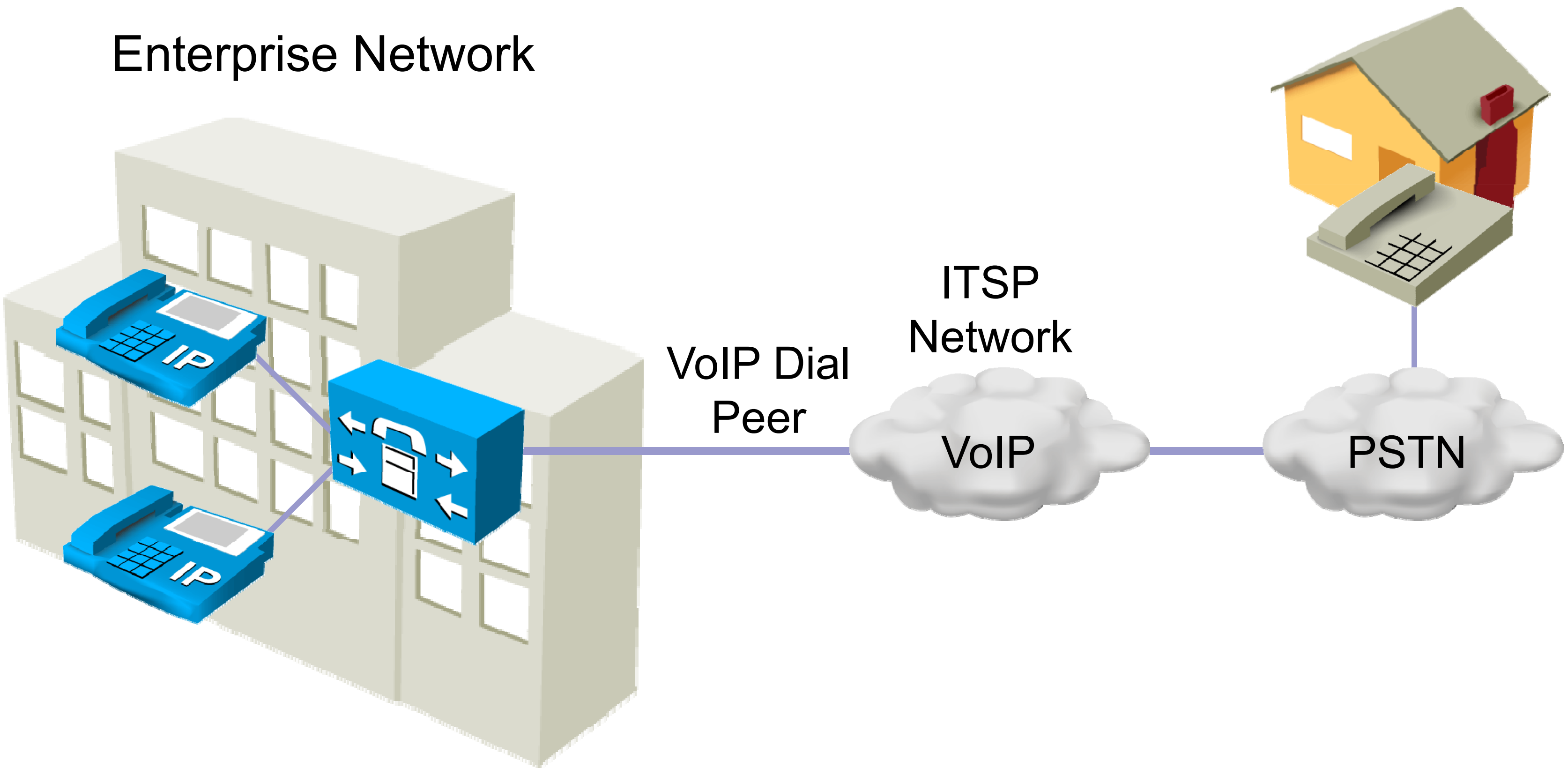
POTS Dial Peers



Dial peer 20 will be used to match outbound when the router receives a call setup message for 1234.

```
CMERouter(config)#dial-peer voice 20 pots
CMERouter(config-dialpeer)#destination-pattern 1234
CMERouter(config-dialpeer)#port 1/0/1
```

Internet Telephony Service Providers (Cont.)



Dial Peers and Destination Patterns Summary

- Gateways translate between two different networks. They can be analog or digital.
- Voice ports are used to terminate a traditional telephony interface on the voice gateway.
- Call legs represent segments in the call path that connect two devices.
- Dial peers represent programming on the voice gateway that defines what to do when a call setup message is received.
- An ITSP trunk is an IP connection to the carrier for PSTN calls.
- You can use **show** commands to verify dial peer and dial plan configurations.