DPI DC

شرکت داده پردازی ایران

IP telephony Products

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Overview

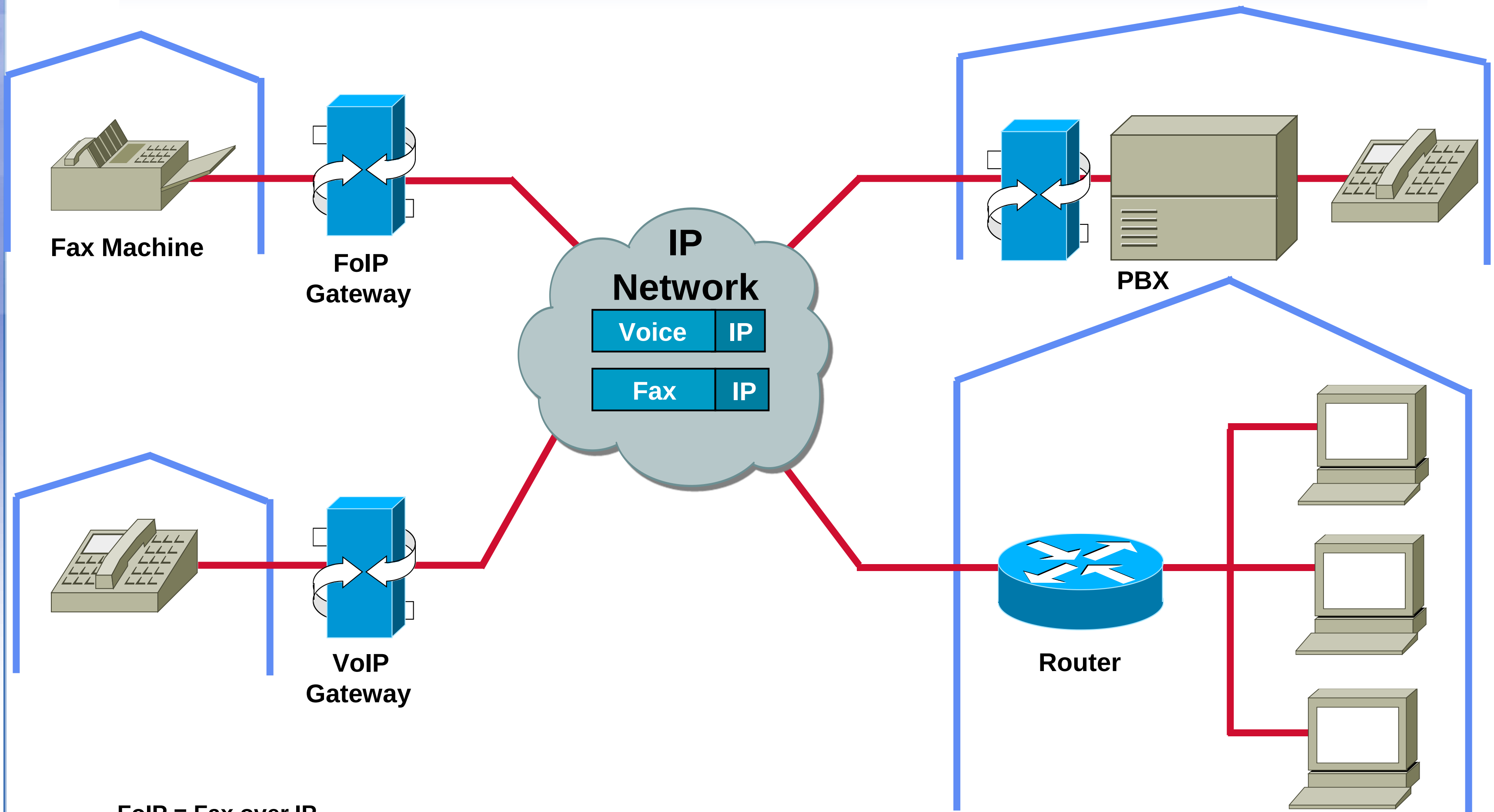
- *VoIP, An overview*
- *DSG Products & Solutions*
- *Cisco Products*
- *[Putting It All Together]*
- *[Technical discussions]*
- *Summary*

VoIP, An overview

VoIP – Voice Over Internet Protocol

- *The transmission of human or synthetic speech in IP datagrams*
- *Can be real-time (conversion) or non-real-time (messaging)*
- *Can be transported over public or private networks using a variety on WAN or LAN technologies*
- *Can be phone-to-phone, PC-to-phone, or PC-to-PC*
- *Does not imply specific standards for voice coding, signaling, or network performance*

VOICE AND FAX OVER IP



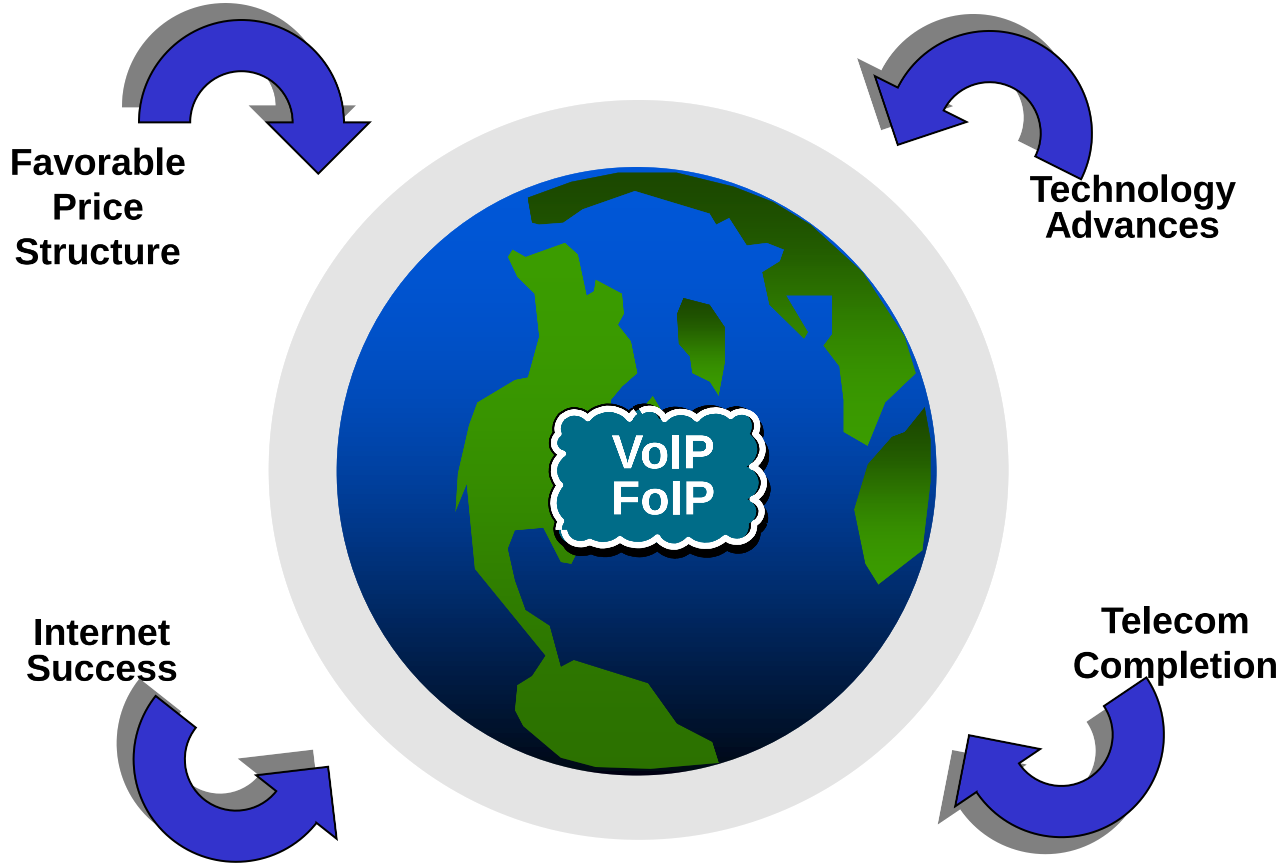
FoIP = Fax over IP
VoIP = Voice over IP

IP = Internet Protocol

Why the Interest?

- ***Short-Term: cheaper long distance, especially international***
- ***Mid- to Long-Term: the promise of a more efficient infrastructure***
    - *Combine all traffic in a packet network*
    - *Use of Web tools for simple management and service access*
- ***Long-Term: the opportunity to put voice networking and applications on the same growth and learning curve as the Internet and WWW***
    - *Promise of open interfaces for service creation*
    - *Take advantage of silicon economics*

WHY THE INTEREST NOW ?



VoIP =Voice over IP
FoIP = Fax over IP

Why the Interest Now?

- *The telecom world is opening to competition*
  - *Deregulation in Europe*
  - *Local competition in North America*
  - *VoIP proponents expect an opportunity to provide a more flexible and feature-rich network at a lower cost*
- *Technology advances make it possible*
  - *Widespread use of echo cancellers is now affordable*
  - *Speech compression reduces bandwidth requirements for toll quality by a factor of 6 to 10*
  - *Network backbone technologies like ATM and Giga-routers can support high bandwidth, low latency transmission*
  - *Dense Wavelength Division Multiplexing makes bulk bandwidth more affordable than ever before*

Why the Interest Now?

- *Incredible success of the internet*
  - *Access to most countries*
  - *De Facto standards and user interfaces based on IP and the Web*
- *Favorable pricing structures*
  - *ITSPs are classified as enhanced service providers in the US and are not subject to access fees*
  - *In most foreign countries ITSPs pay network termination charges much like a business PBX but they are not subject to international settlement charges*

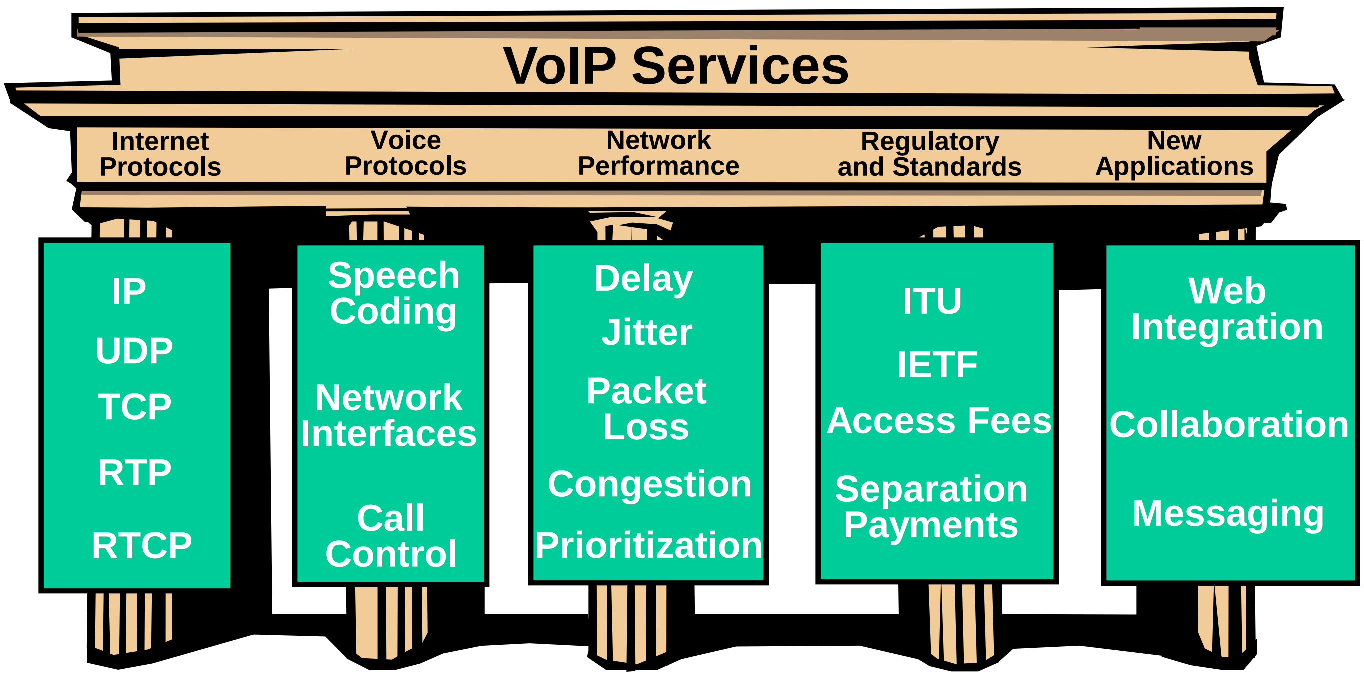
The VoIP Industry Reaches Adolescence

- *ITSPs are rapidly building out their networks*
  - *Initial focus is cheap long distance*
  - *Longer-term focus is enhanced services*
  - *Much consolidation can be expected*
- *Companies explore Voice and Fax over IP in their private networks*
  - *Main driver is cost savings*
  - *Performance is more controllable*
- *There are many issues which must be addressed*
  - *Standards for interoperability*
  - *Regulatory status*
  - *Numbering plans*
  - *Interworking with PSTN*
  - *Billing and net management*

Some Interesting Comments from Speakers at the Voice on the Net Conference

- *In 1998, there were more than one trillion minutes of POTS usage*
- *The US market for telephony services is about $250 billion and the global telecom services market is around $800 billion*
- *The cross-over for wide-area data traffic exceeding voice traffic is happening about now, but voice revenues are still much greater than data revenues*
- *By 2004, 5% to 20% of long distance calls will be VoIP*
- *Circuit switching will be dead by 2005*
- *Voice will be only 1% of the total global network traffic by 2008*

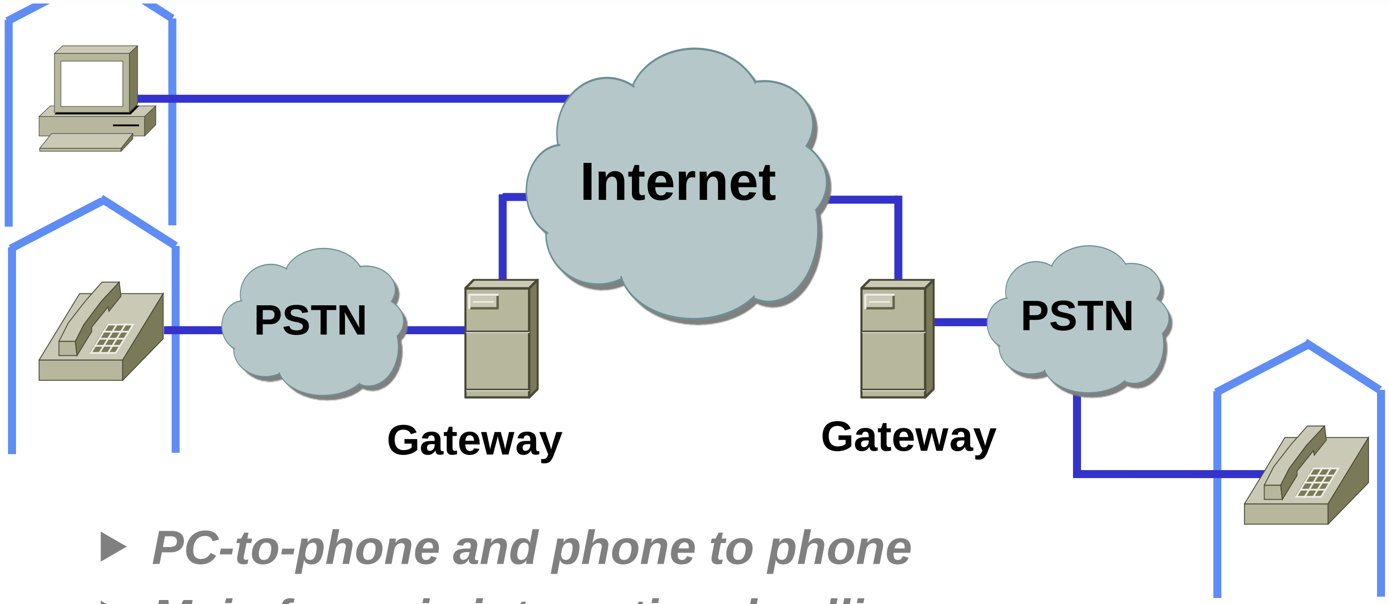
VoIP BUILDING BLOCKS



VoIP Components

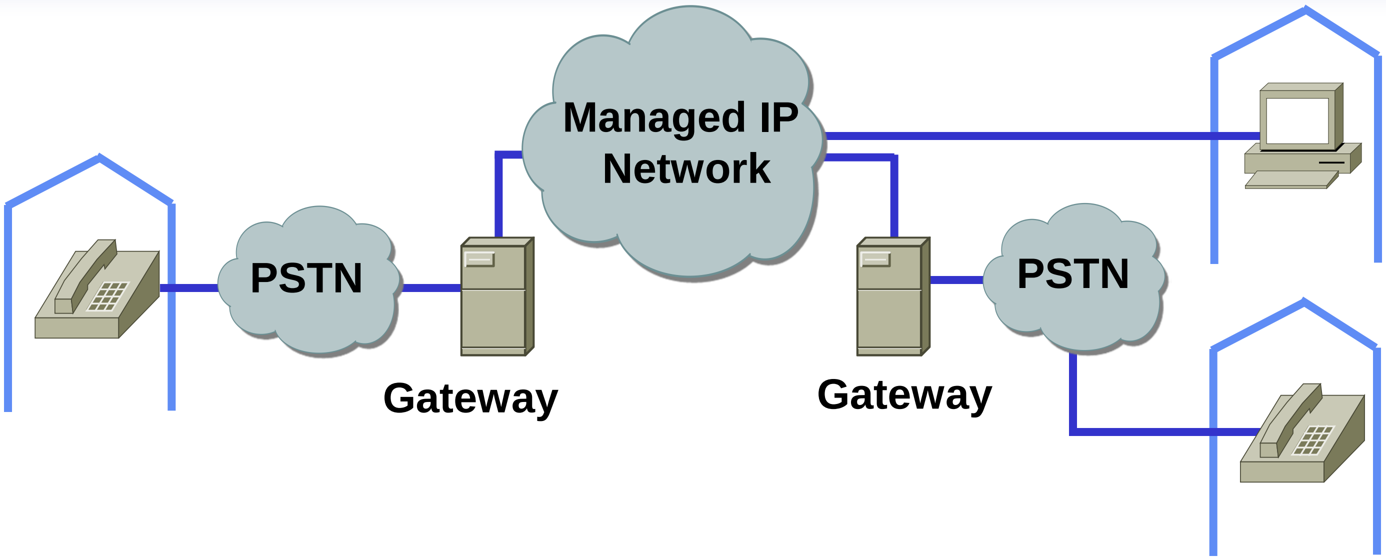
- *ISPs, ITSPs & providers*
- *Gateways*
  - *Phones, IP PBXs,*
- *GateKeepers:*
  - *Authenticating/Accounting system*
  - *Billing system*
  - *management system*
- *Inter(a)net Connection*
- *[PSTN connection]*

ISPs Deploy Gateways for PSTN Connections



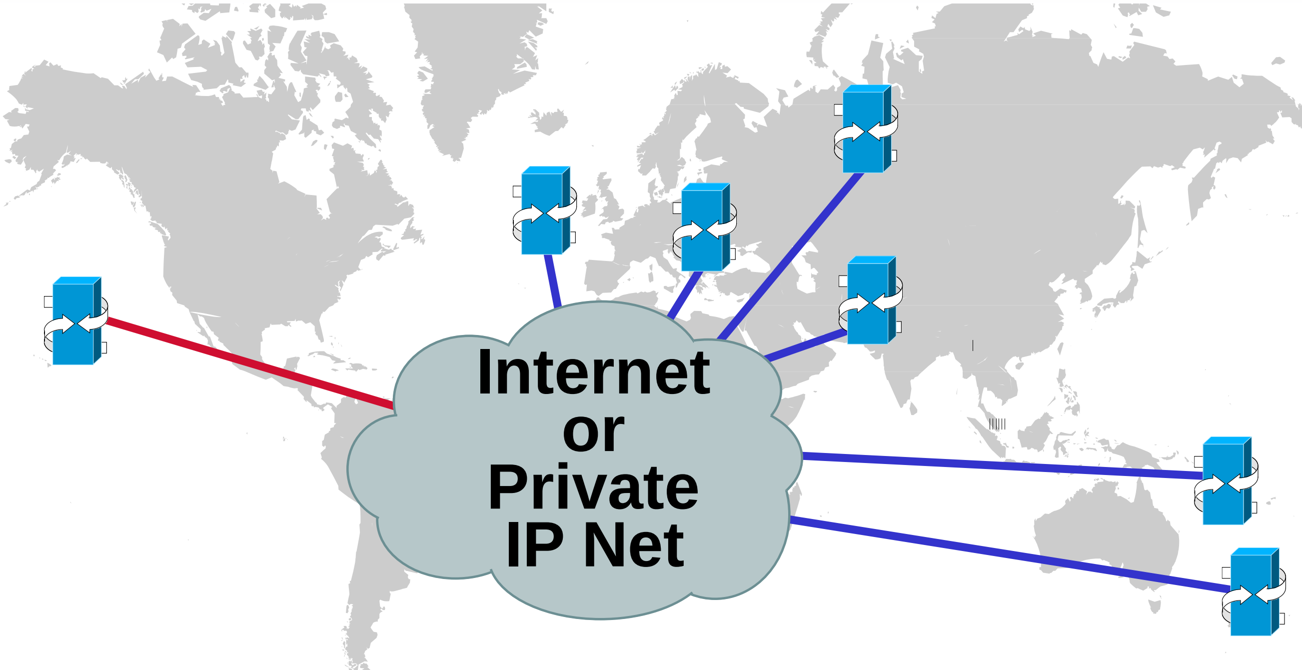
- PC-to-phone and phone to phone
- Main focus is international calling
- Limited to calling locations in which gateways are placed
- Discounts of 30-70% compared to PSTN
- Delay is still a problem

Internet Telephony Service Providers Must Control Network Performance



- Control the network's performance for voice traffic
- Carries strictly voice traffic or provides priority of voice over data
- Positioned as enhanced service provider from regulatory perspective
- Offers cheap long distance as initial service

Internet Telephony Service Providers



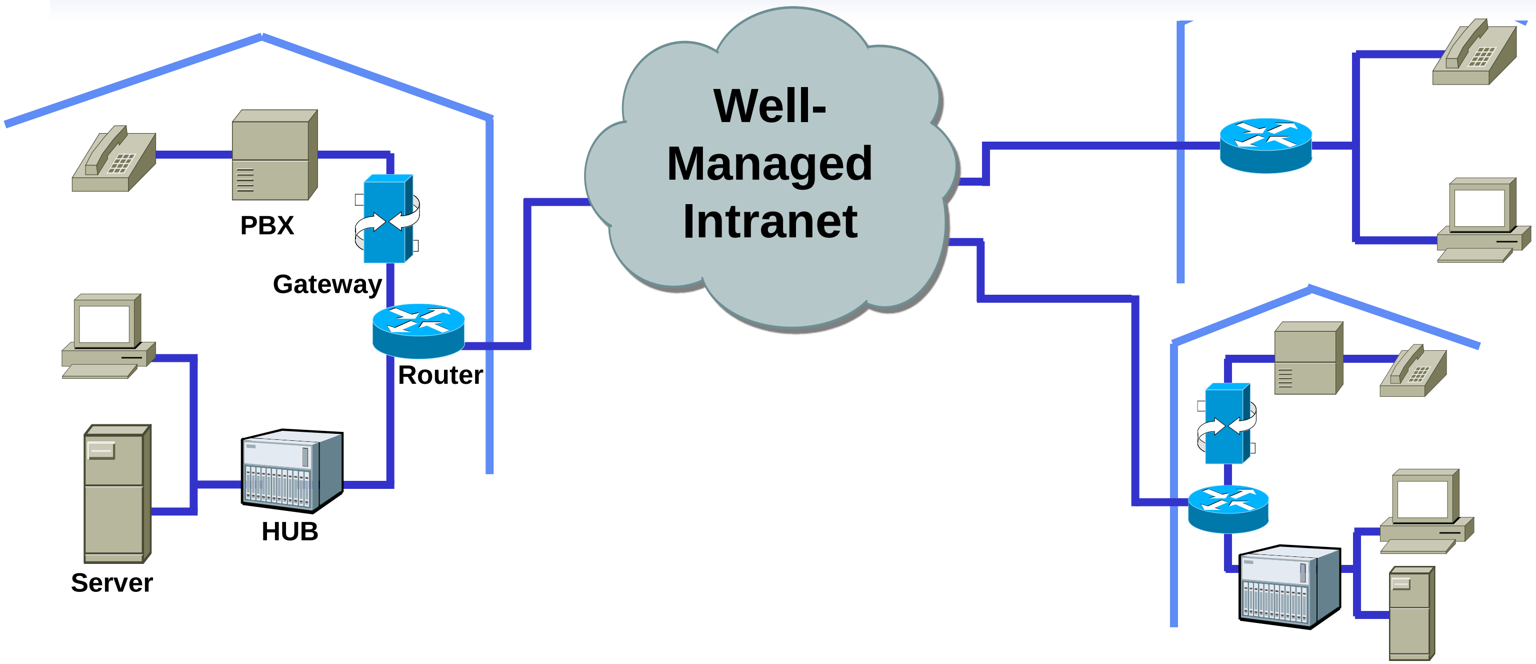
- Place gateways in countries with high international tariffs and/or high calling volumes
- Gateways support standard end user business interfaces to PSTN
  - Analog, T1, E1, ISDN
- May be subject to performance inconsistency of Internet

ITSP Examples

- *Access Power (North America)*
- *Delta Three*
- *FNET*
- *GlobalNet (Europe)*
- *iBasis (public Internet)*
- *IncomTel TG (Russia)*
- *INTERLINE*
- *Level 3*
- *Net2Phone (PC-phone)*
- *Net Communications Inc.*
- *POPTEL (public Internet)*
- *Qwest*
- *Telematrix (Asia)*
- *USA Global Link*
- *WIN (PC-phone)*

. . . and many others!

Gateways are Deployed in Private Networks

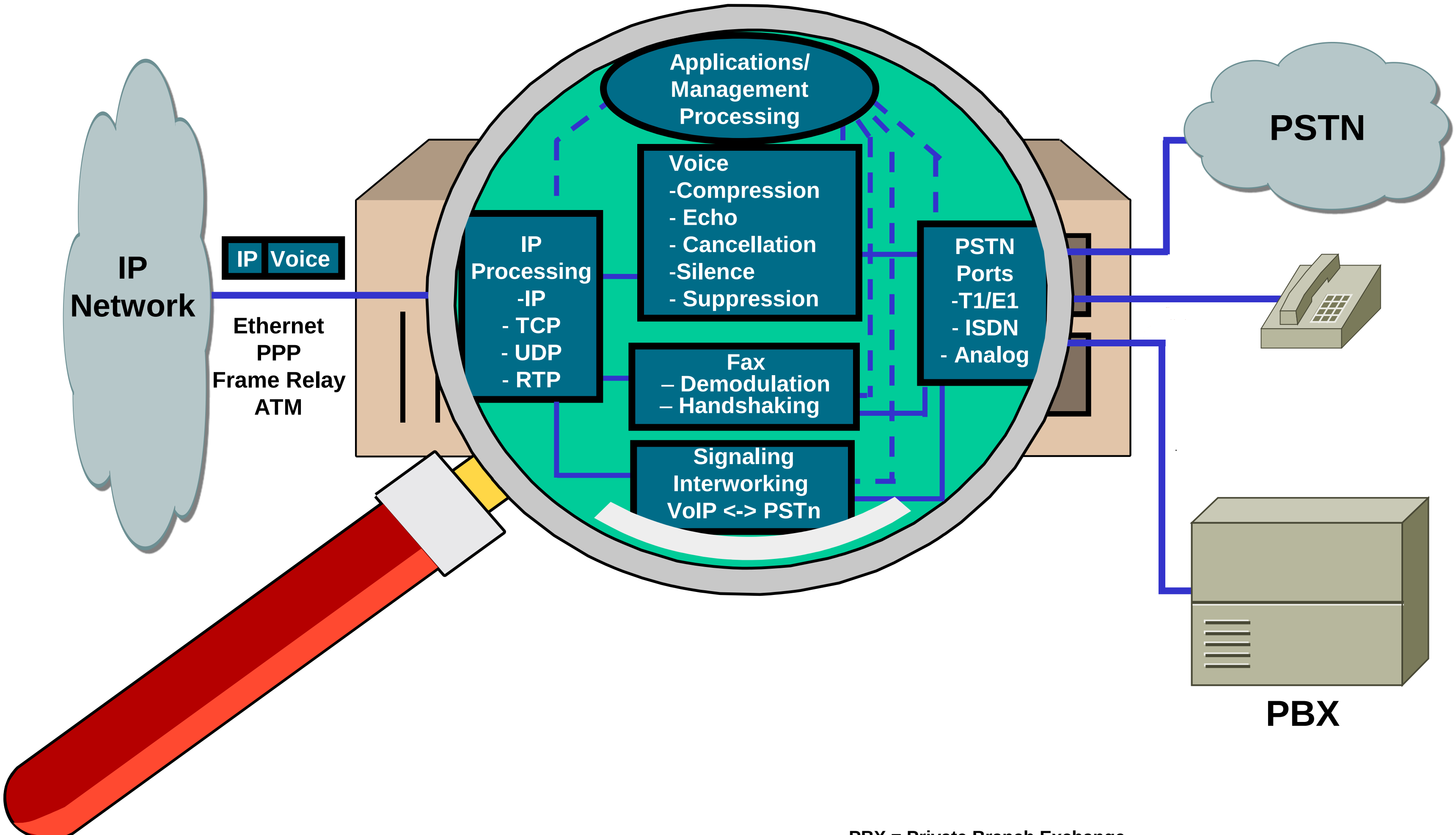


- *Adds voice to existing IP network*
- *Model similar to Voice over Frame Relay*
- *Intra-company calls and tail-end hop-off*
- *Delay more manageable than Internet*

Internet Telephony Gateway Some Capabilities

- *Traditional PSTN interfaces*
    - *Analog, T1, E1, ISDN*
- *Data Communications interfaces for IP transport*
    - *Ethernet, ATM, Frame Relay, PPP*
- *Voice Processing*
    - *Compression*
        - *G.723.1, G.729, proprietary*
    - *Silence suppression*
    - *Echo cancellation*
    - *Playout algorithms*
- *Fax Processing*
    - *Demodulation*
    - *Fax protocol support (T.4, T.30)*
    - *Real-time or store-and-forward*
- *Voice services/ call control support*
    - *ISDN and POTS signaling*
    - *H.225/245 or proprietary*
- *Variable delay buffers to remove jitter*
- *Additional features, e.g., voice mail*

Generic Gateway



PBX = Private Branch Exchange
PSTN = Public Switched Telephone Network

Sample VoIP Product Vendors

- *Cisco*
- *Clarent*
- *elemedia (SW)*
- *Ericsson*
- *Inter-tel*
- *Linkon (SW)*
- *Lucent*
- *Nortel Networks*
- *(MICOM)*
- *Motorola*
- *NetSpeak*
- *NeTrue*
- *Nokia*
- *Nuera*
- *RADVision*
- *TI Telogy (SW)*
- *VocalTec*
- *DSG*

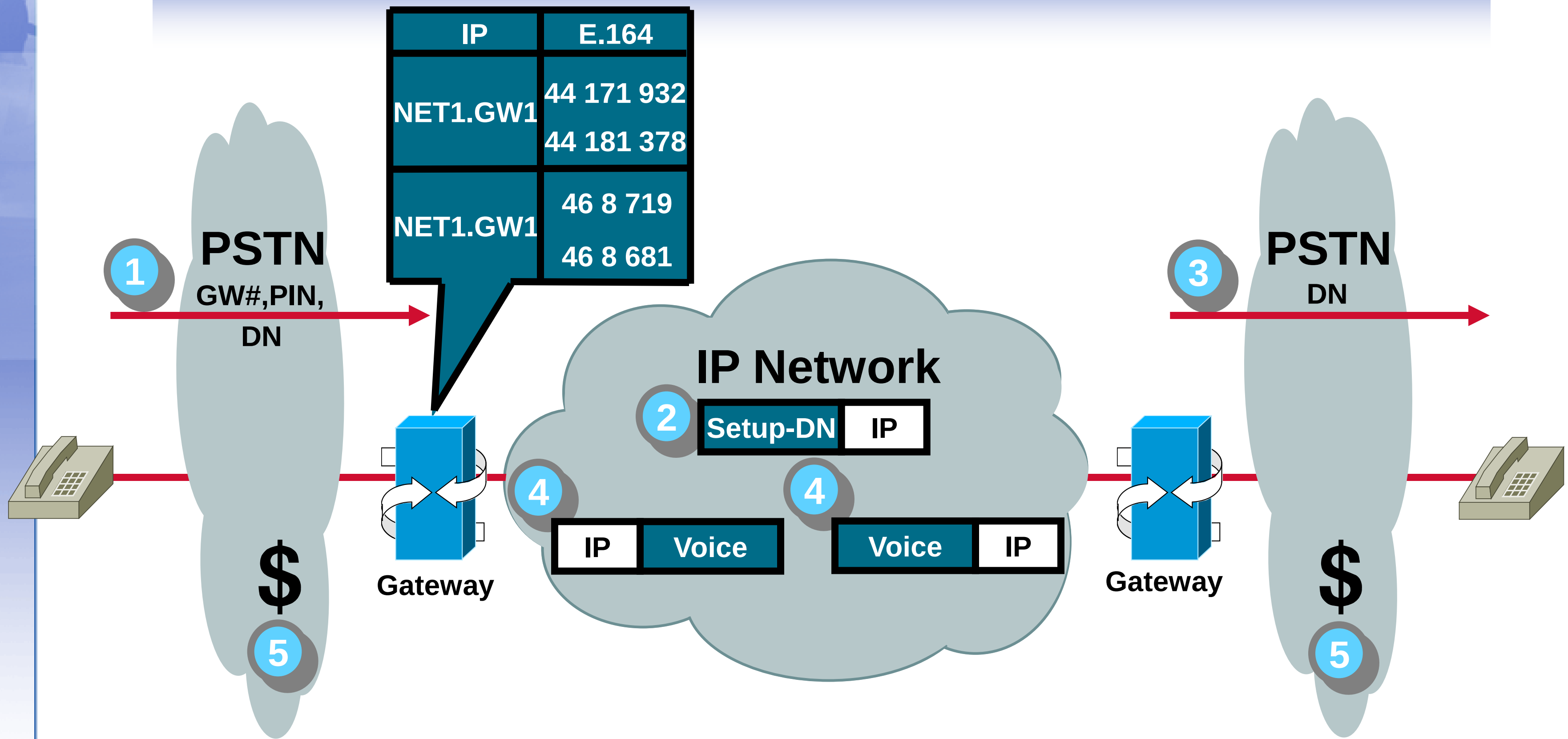
. . . and many others !

SW = Software only vendor

Making an ITSP Call

1. Multi-stage dialing
   - Call gateway via local access or toll free number
   - Key in PIN, usually for prepaid minutes
   - Key in destination number
2. Gateway identifies remote gateway nearest called number
   - E.164 to IP address resolution
   - Call “setup” sent via IP to remote gateway
3. Remote gateway calls destination number
4. Gateway convert speech for transport via IP
5. PSTN sees two local calls

Making an ITSP Call



DN = Destination Number
PIN = Personal Identification Number
GW# = Gateway Number

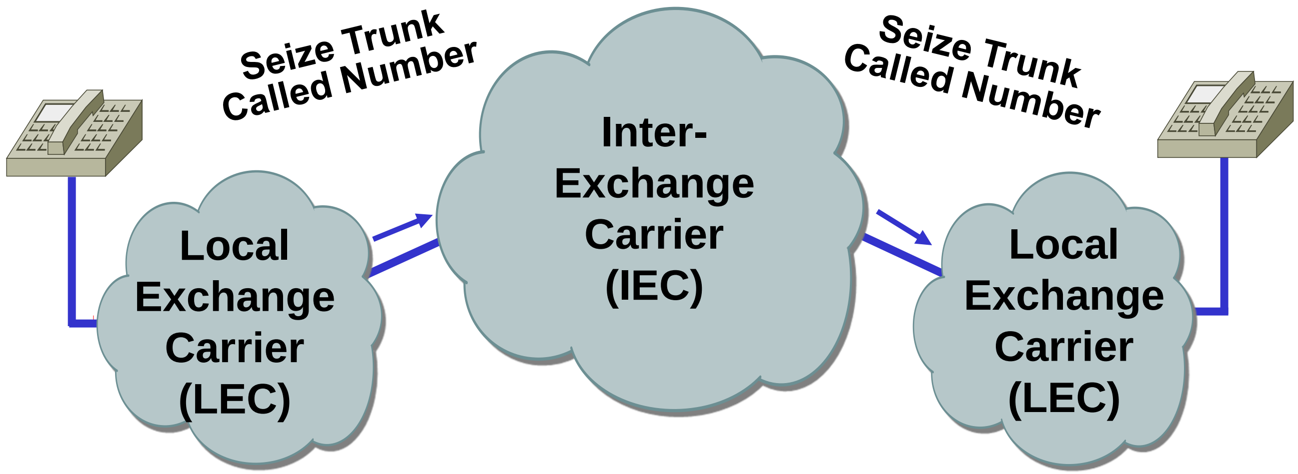
Characteristics of an ITSP Call

- *Longer set-up times*
  - *Dialing similar to calling card calls*
  - *Post dial delay typically greater than PSTN*
- *Quality depends on the network, typically*
  - *Noticeable drop compared to PSTN*
  - *But, perfectly acceptable for many applications*
  - *Similar to digital cellular on both ends of call*
  - *Quality will continue to improve as technology evolves*
- *Can be more expensive for corporation with volume discounts*
- *Can be significantly cheaper for individuals making international calls*

Why is ITSP Service Cheaper ?

- Less expensive infrastructure
  - No voice switches, SS7 or AIN (Adv. Intelligent)
  - Sharing inherent in packet networks
  - Not clear whether an infrastructure providing services and reliability equivalent to PSTN would be cheaper
- Simple prepaid fees
  - No printing and mailing of bills
  - No collections for bad debts
- Favorable regulatory status of enhanced service providers
  - No access fees
  - No international settlement payments

Access Fees



- LEC charges IEC for originating and terminating access
- Makes IECs single largest customer of LECs
- Single largest expense for IECs

Regulatory Status

- *Most of the world has taken a hands-off position on regulating Enhanced Service Providers and the Internet*
- *In general, Internet Telephony Service Providers have successfully positioned themselves as ISPs*
- *As a result, ITSPs typically do not pay carrier interconnection fees such as access fees or settlements*
- *But, Internet telephony is regulated in some countries and illegal in others*

VoIP Regulations in the US

- *So far, the FCC and other government agencies have classified ITSPs as Enhanced Service Providers*
- *In an April 10, 1998 report to Congress, the FCC:*
  - *Reaffirmed that Internet Service Providers were not subject to Universal Service Obligations, access fees or rate regulation*
  - *Recognized that phone-to-phone services over the Internet fit the definition of "telecommunication services" but chose to take no action*
- *BellSouth and U S West informed ITSPs in their regions that they would be treated as Interexchange Carriers but the FCC told them to slow down*
- *U S West has formally filed with the FCC to treat ITSPs as Interexchange Carriers*

ISTP = Internet Telephony Service Provider
FCC = Federal Communications Commission

VoIP Regulations in Canada
Calls Terminating to the Public Network Pay "Contribution"

- *CRTC 97-590*
  - *Calls which use the Internet as the underlying transmission facility shall be subject to contribution*
- *CRTC 98-28*
  - *Enhanced service providers are not eligible for an exemption (Shadow Tel Communications)*
- *About 4 cents per minute*

CRTC = Canadian Radio-television and Telecommunications Commission

Regulations in Europe

- European Union criteria for regulation of Internet telephony
- It must:
  - Be commercially offered
  - Be provided to the public
  - Be to and from the public switched network terminating points on a fixed telephony network
  - Involve direct transport and switching of speech in real time
- Many ITSPs have taken this opportunity to operate without licenses in Europe
- There is growing consensus that providers of PC-to-phone and phone-to-phone services should be licensed as voice telephony operators

ISTP = Internet Telephony Service Provider

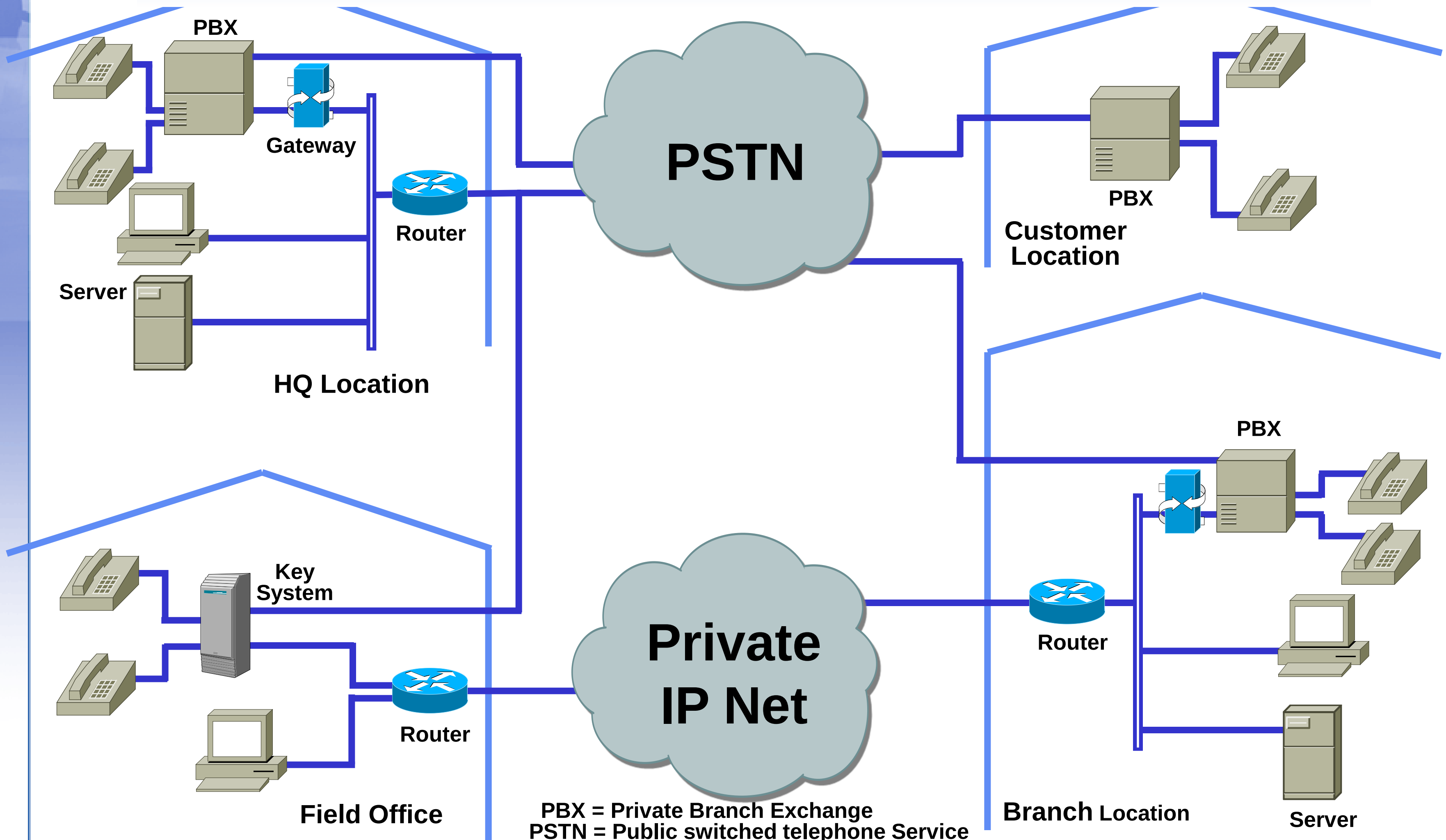
VoIP Regulations in Asia

- *There is no Pan-Asia regulatory body, status varies by country*
- *In many countries unlicensed termination of voice calls (e.g., toll and separation payment bypass) is illegal and gateways of violators may be confiscated, e.g., China, India, Myanmar, Pakistan, Singapore, Taiwan*
- *In some countries, newly licensed competitive carriers are using VoIP now or will use it when they begin offering service, e.g.,*
  - *Now: Japan, China*
  - *Future: Taiwan, Singapore*
- *In some countries, the incumbent carriers have begun to offer VoIP services, e.g., China, Korea*

Private VoIP Networks

- Primarily driven by cost savings
  - Reduce minutes paid to carriers for intra-company calls
  - Reduce access costs by combining voice, fax and data on common facility
  - Support tail-end hop-off for some PSTN calls
- Relatively small investment
  - Telephony gateways
  - Incremental bandwidth in IP network
  - Router software to prioritize Voice over Data
- Capital costs typically recovered in less than 1 year, depending on current voice tariffs

Private VoIP Networks



PBX = Private Branch Exchange
PSTN = Public switched telephone Service

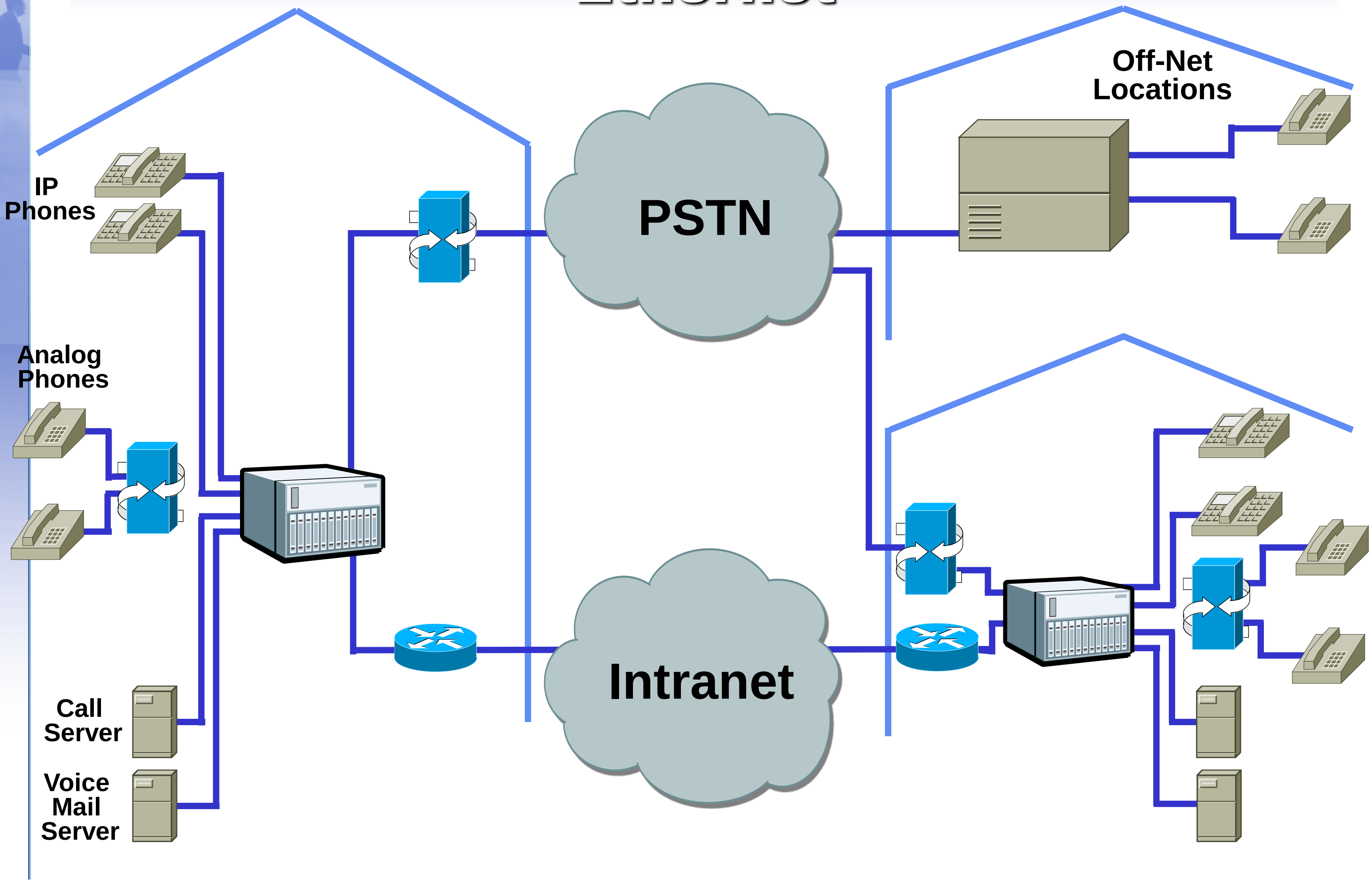
Private VoIP Networks

- *Gateways*
  - *Available from a variety of vendors*
  - *Can be stand-alone, PC card, router card or PBX card*
  - *Typically support voice and fax*
- *Dialing options*
  - *Integrated with Private Branch Exchange (PBX) least-cost routing, so transparent to user*
  - *Or requires special access code*
- *No regulatory issues*
- *Voice quality will depend on network design*
  - *Delay and delay variation must be managed*
  - *May be poorer than PSTN*

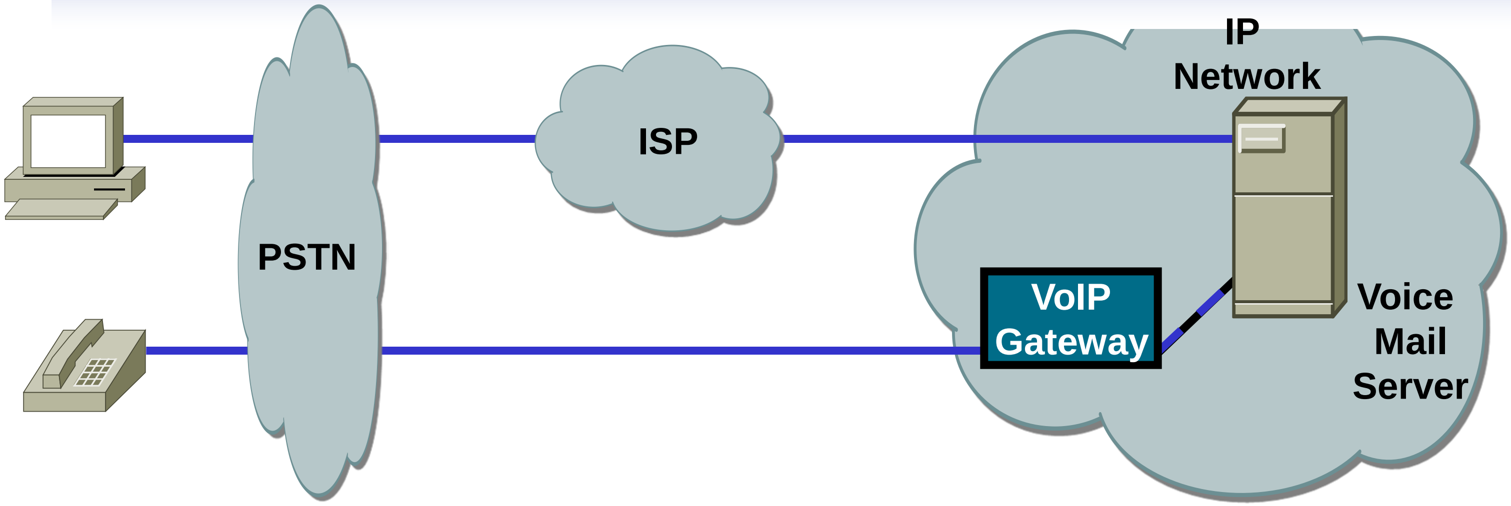
IP PBX

- *Support voice and data on switched/higher speed Ethernet infrastructure*
- *Selling Points*
  - *Consolidated wiring plan*
  - *User configuration of services via browser interface*
  - *Easy moves and charges via DHCP*
  - *Open interfaces for service creation*
  - *Cost advantage of standard components vs. proprietary PBX*
- *Question Areas*
  - *Reliability*
  - *Feature set*
  - *Scalability*
  - *Cost of IP phones*
  - *Voice quality*

IP PBX
Voice Overlay on Switched/High Speed Ethernet



Voice Mail



- Calls diverted to VoIP gateway on busy/don’t answer for voice mail customer
- For phone retrieval
  - Circuit-switched connection required only to VoIP gateway
  - Delay not an issue for voice message playback
- For PC retrieval
  - Access to voice messages via Internet
  - Conversion to speech at PC

VoIP Issues

- *Voice quality*
  - *Can delay be acceptably reduced?*
  - *Can VoIP be as good or better than POTS?*
  - *How good is good enough?*
- *Reliability*
  - *How long until VoIP can provide lifeline service?*
  - *Will companies trust VoIP as their only long distance service?*
- *Internetworking with the Public Switched Telephone Network*
  - *SS7/IN capabilities*
  - *OAM&P capabilities*
  - *Number portability*
- *Standards*
  - *Call control*
  - *Service access*

POTS = Plain Old Telephone Service
IN = Intelligence Network
OAM&P = Operations, Administration, Maintenance and Provisioning

VoIP Issues (continued)

- *Interoperability*
  - *Vendor gateway to vendor gateway*
  - *Vendor gateway to PSNT for advanced features*
- *Incumbent carrier migration*
  - *Circuit switched net migration to IP*
  - *Does VoIP make sense for POTS subscribes?*
- *Numbering plans*
  - *E.164 vs. IPv6 vs.?*
- *Regulations*
  - *What happens to industry growth if arbitrage opportunity disappears?*

DSG Products

DSG Technology

- http://www.dsgtechnology.com/
- *Founded in 1997 in Taiwan*
- *started worldwide Phone-to-Phone Internet Telephony Network with*
  - *NT-based VoIP gateway,*
  - *Gatekeepers and Billing Systems*
  - *embedded dial-up CPE in 1998 and Ethernet CPE in 1999*
  - *Internet Telephony Networks*

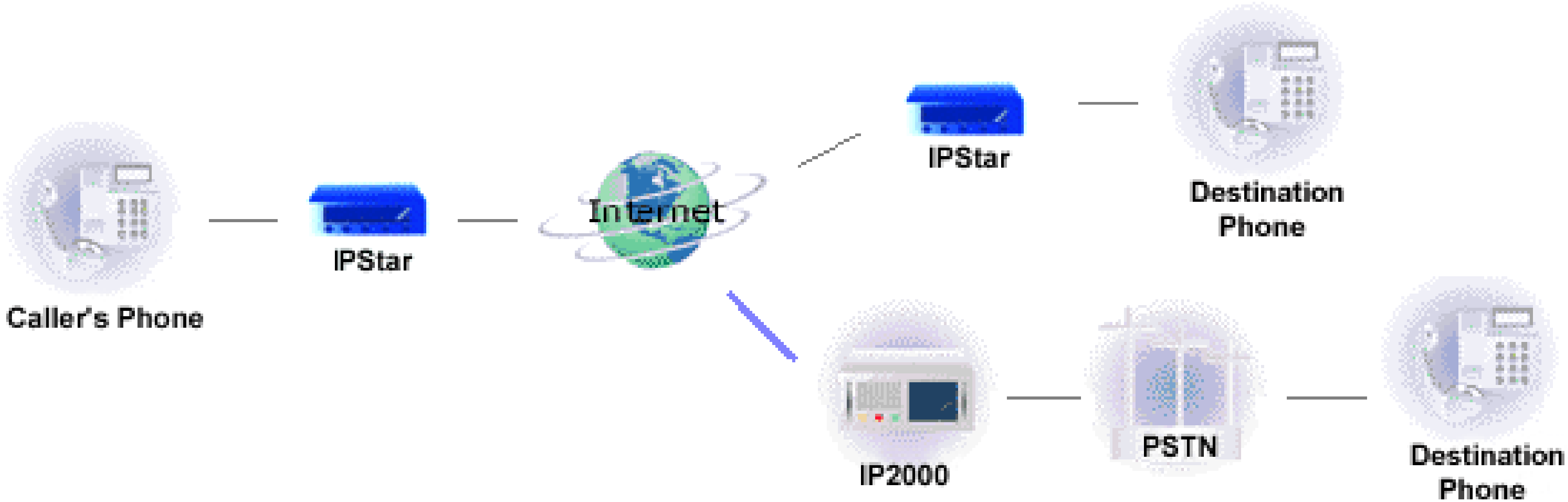
Solutions

- *Dialup (IPStar)*
- *Leased Line (InterStar, InterPhone)*
- *ITSP (IP2000)*
- *VPN (IP1000)*

- *InterPBX*
- *VoiceMail*
- *Voice Recording system*

DialUp (IPStar)

Internet access: PSTN

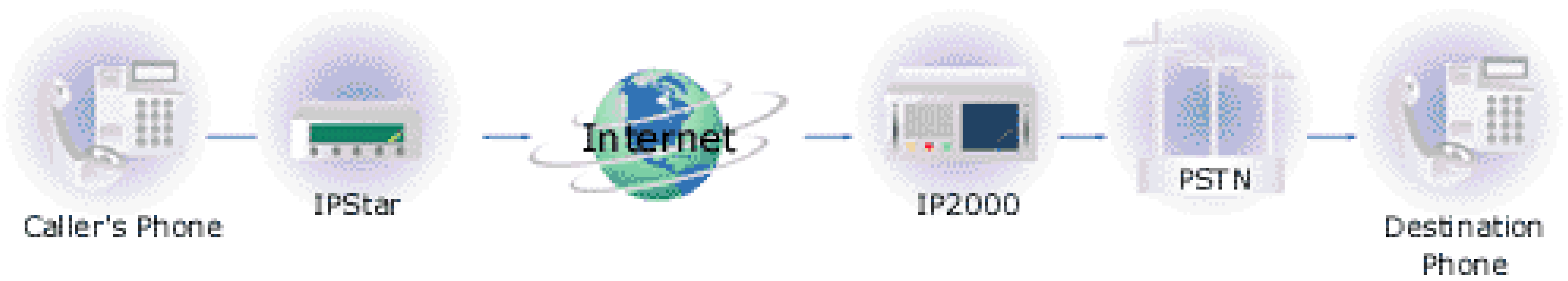


IPStar

- Internal Modem : ITU-T V.34, 33,600bps
- two RJ11 jacks to line and phone
- Voice Compression: G.723.1 compliant
- Internet Protocols :TCP/IP, PPP, PAP, CHAP
- IPStar-to-IPStar Mode

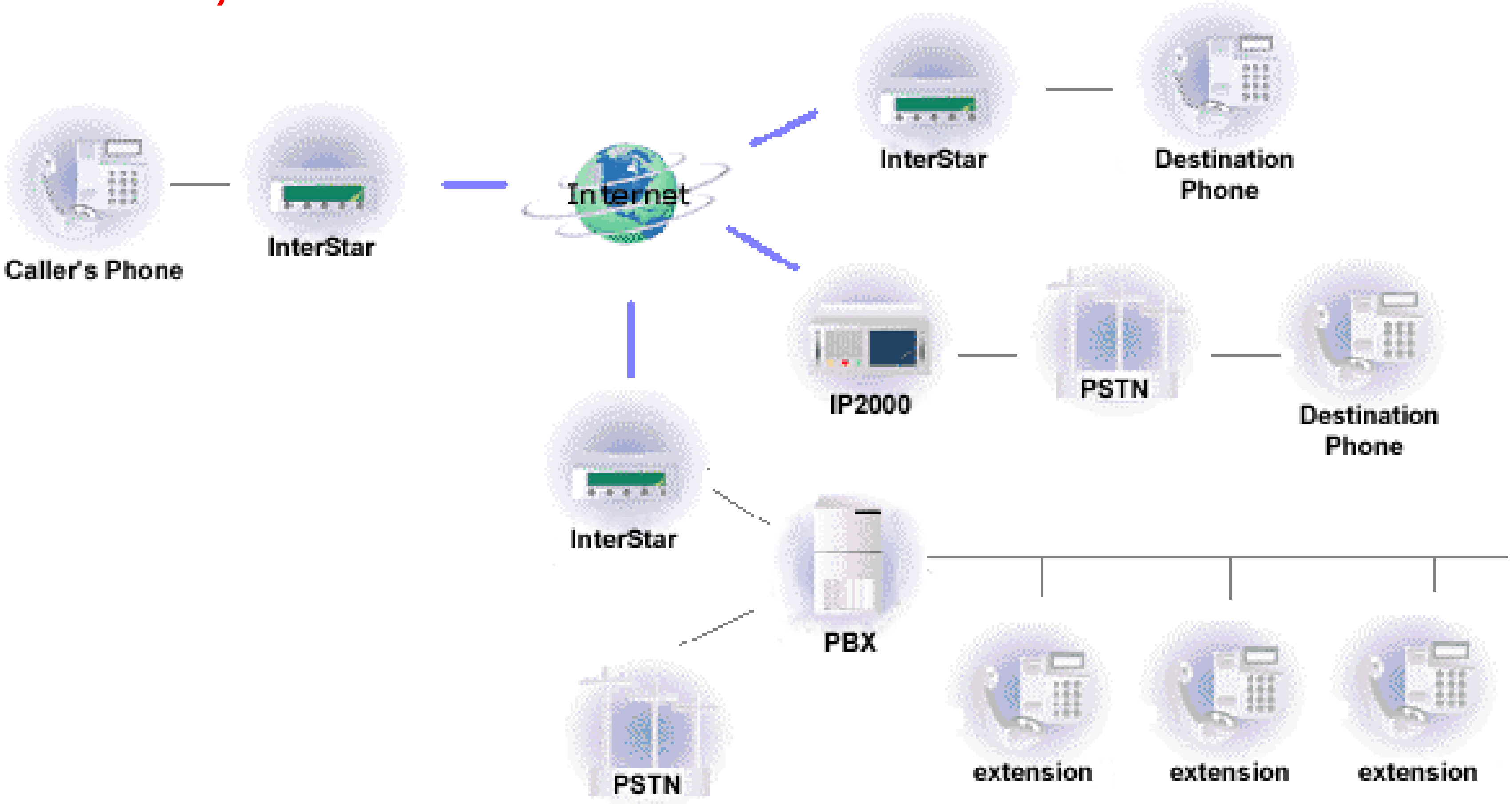
IPStar

- IPStar-to-Phone Mode
- Direct-Link Mode (6-digit ID)
- Direct IP Mode



Leased Line (InterStar)

Internet access: **Ethernet, Leased Line, xDSL, Cable Modem, ISDN and Home- PNA**



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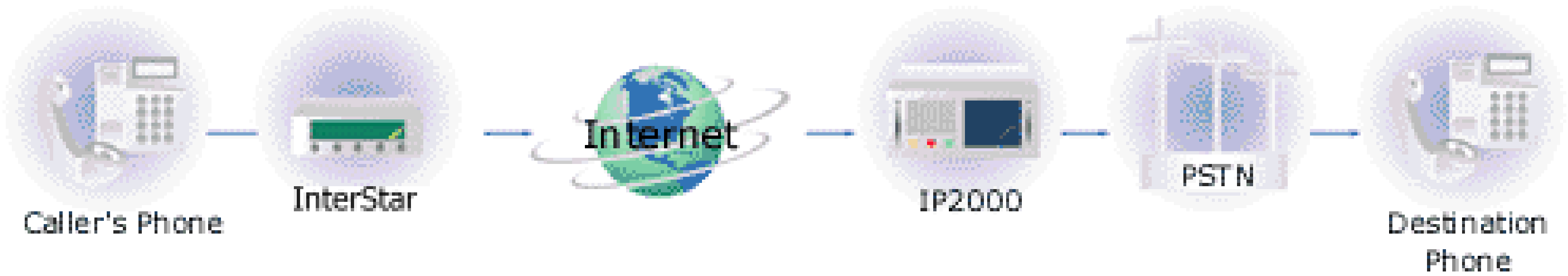
InterStar

- ***InterStar-to-InterStar Mode (IP add)***

- ***InterStar-to-IPStar Mode (Unique ID)***

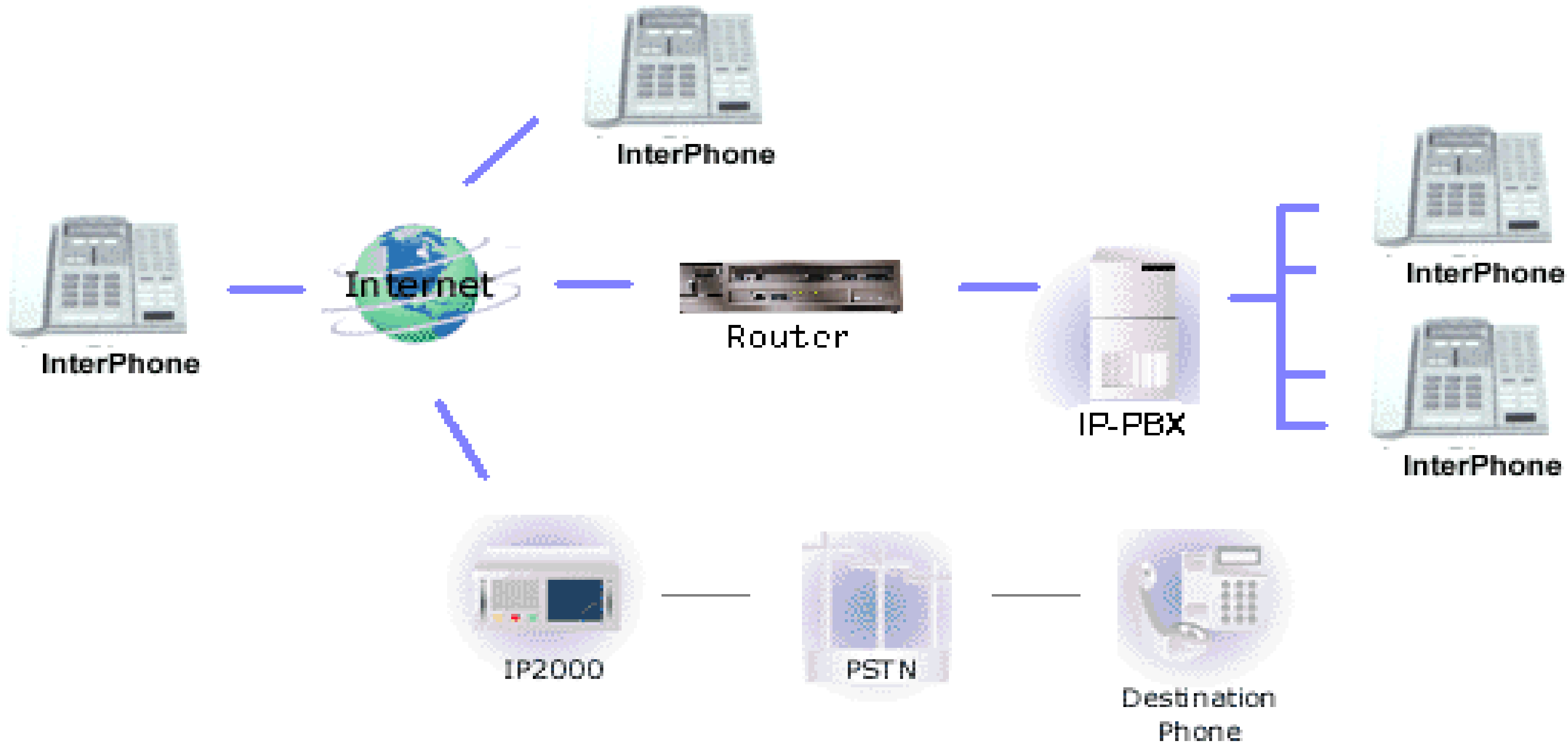
InterStar

- *InterStar-to-Phone Mode*
- *Phone-to-InterStar Mode*



Leased Line (InterPhone)

Internet access: Ethernet, Leased Line, xDSL, Cable Modem, ISDN and Home- PNA

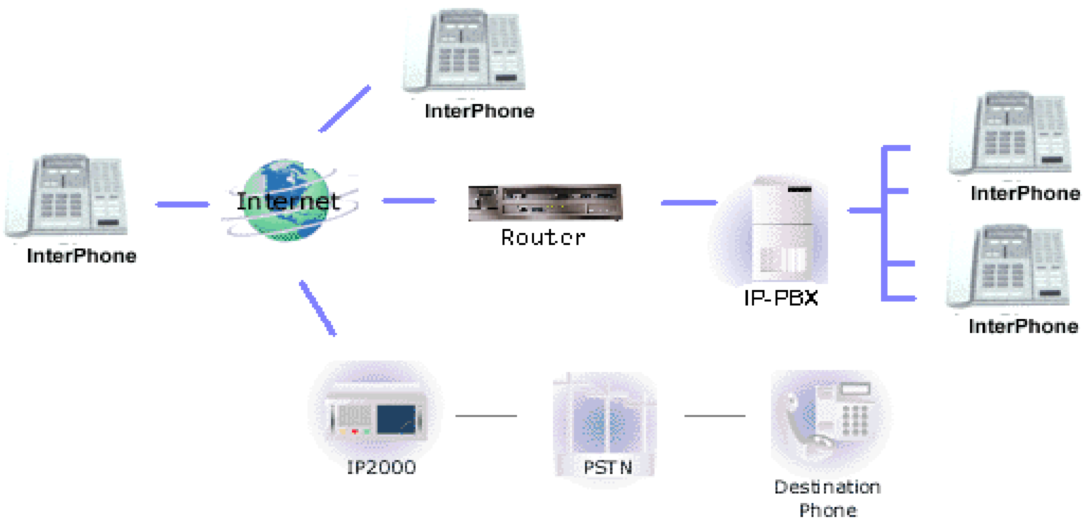


Leased Line (InterPhone)

- *Two-ports 10BaseT Ethernet hub*
- *a 20x2 LCD display*
- *dial keypad,*
- *programmable buttons*
- *one-button memory speed dial (2x21)*
- *Clear Voice Quality*
- *TCP/IP, UDP, DHCP, BootP, Telnet*
- *Free On-Line Upgrade*

Leased Line (InterPhone)

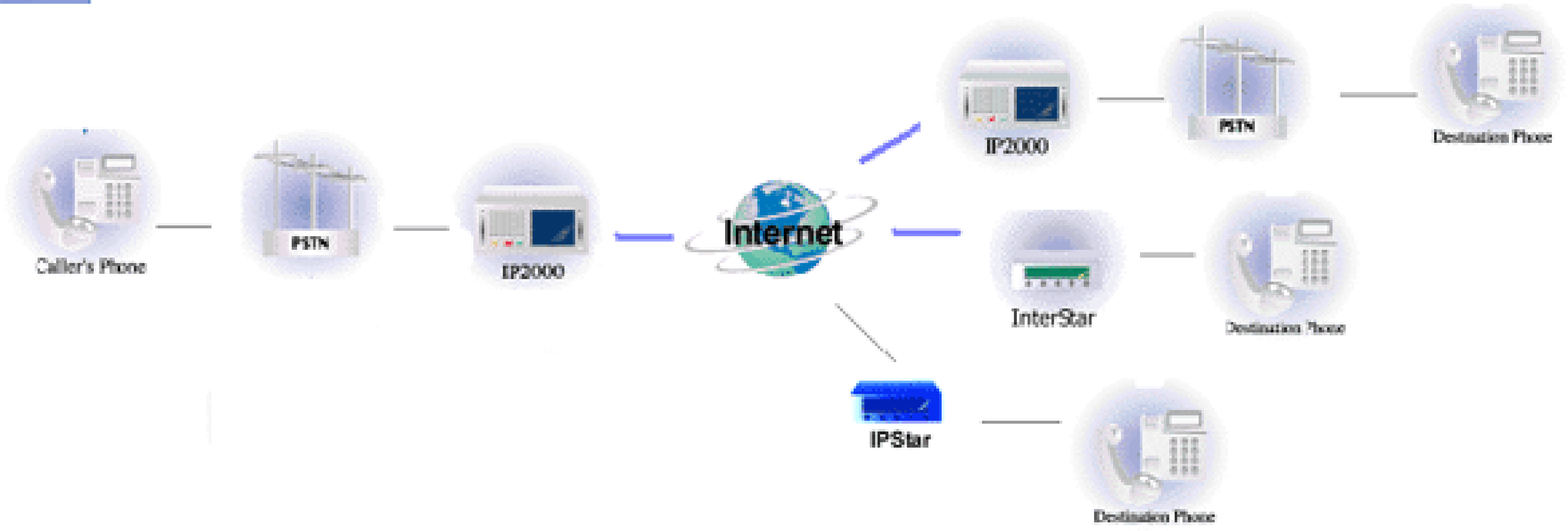
- InterPhone-to-InterPhone/InterStar Mode
- InterPhone-to-IPStar Mode
- InterPhone-to-Phone Mode



IP2000

ITSP (IP2000)

Internet access: Leased Line



IP2000

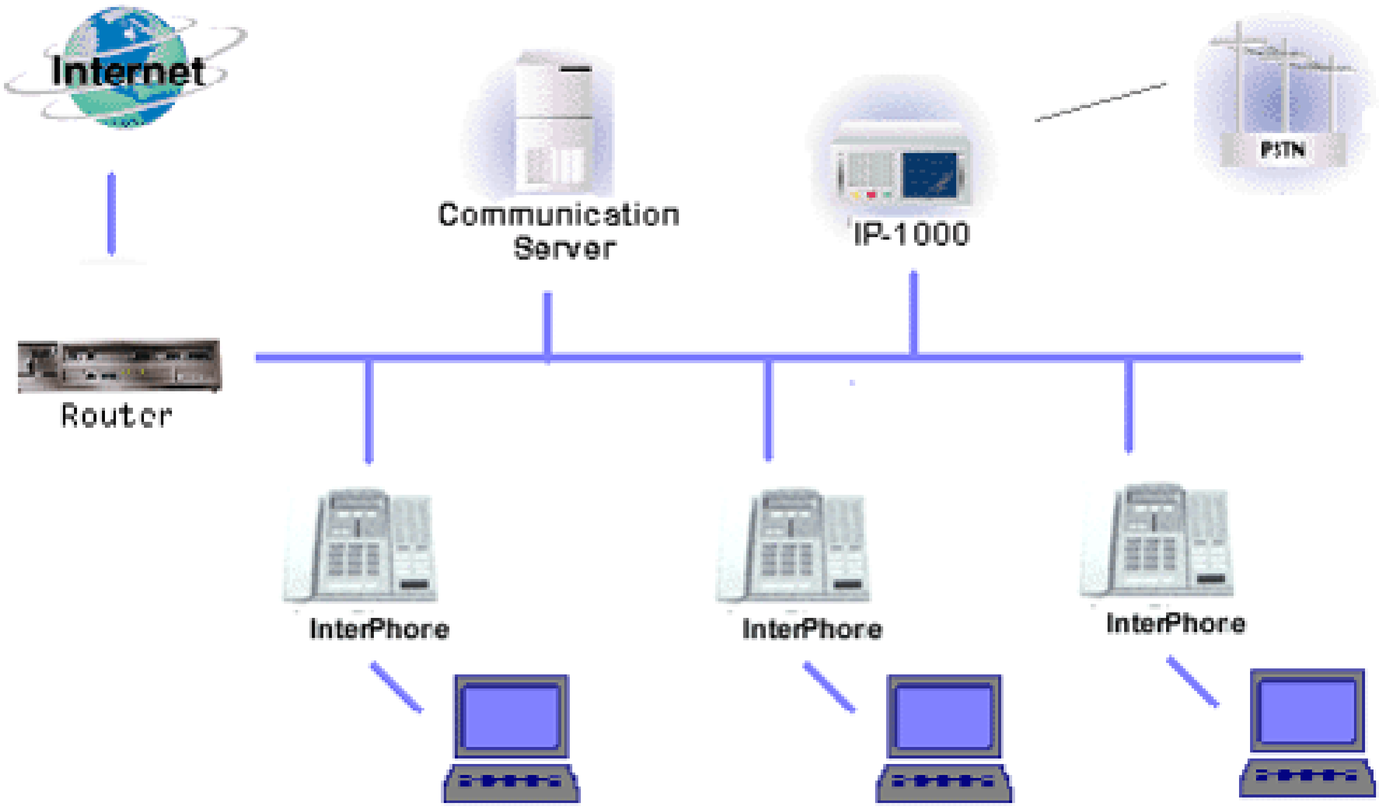
- *Based on IPC machine with Windows NT Workstation 4.0*
- *IPV4-A (analog) ($5000), IPV12-D (T1/E1 interface)*
- *From 4-port up upto 32-port (Analog)*
- *Up to 96 ports (digital interface)*
- *Phone-to-Phone mode (ITSP/VPN)*
- *Device-to-Phone mode (ITSP)*
- *ITU G.723.1 6.3/5.3Kbps, Proprietary: 4.8/4.1Kbps*

IP2000

- *Pre-paid (PPC) business*
- *NT-Based with IPGw & PSS program with lock*
- *Rates: Inbound+DSG center+outbound*
- *Rates: different from IP1000*

VPN (IP1000)

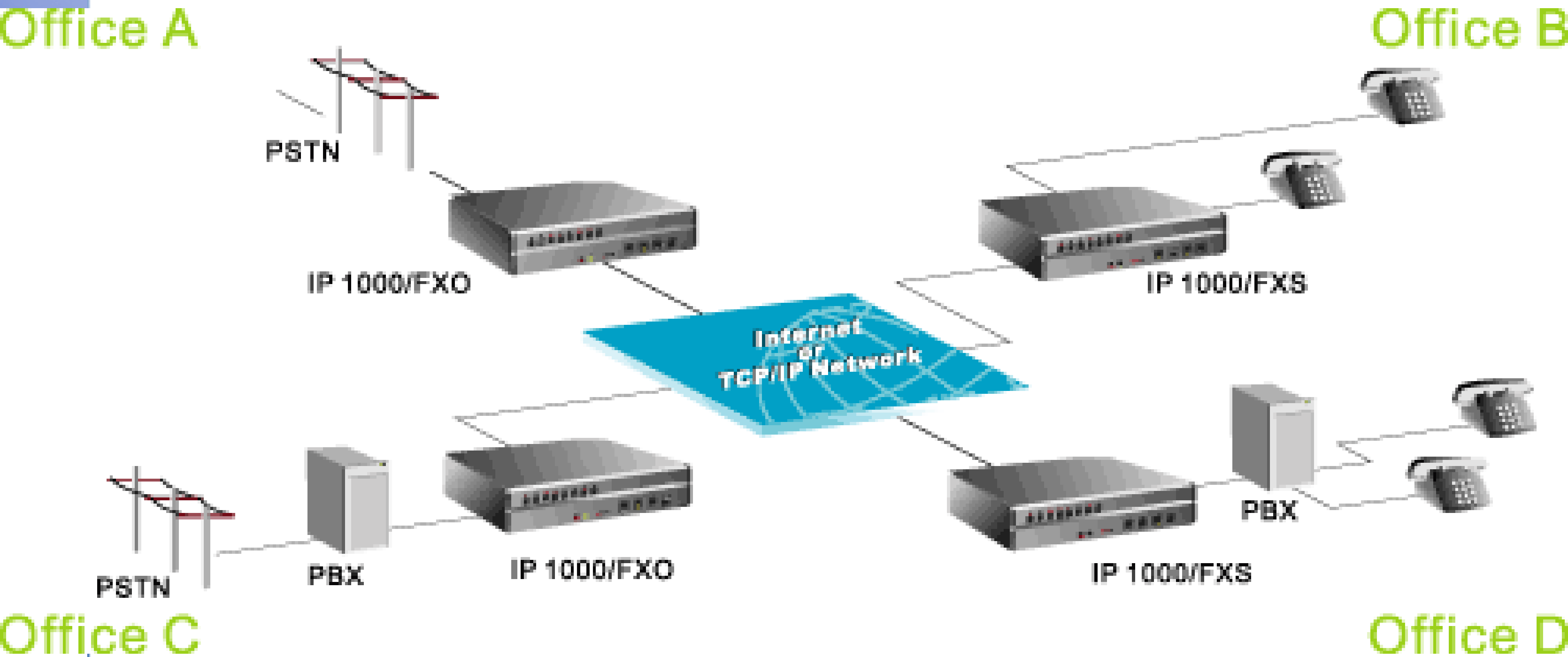
Internet access: Leased Line



IP1000

- G.723.1 standard
- TCP/IP, DHCP, Telnet
- IP1000-4 or IP1000-8
- FXO & FXS RJ-11 ports
- Password protected ports
- 10BaseT Ethernet connection
- VPN & ITSP mode
- $1000 for 4 port

IP1000-VPN mode



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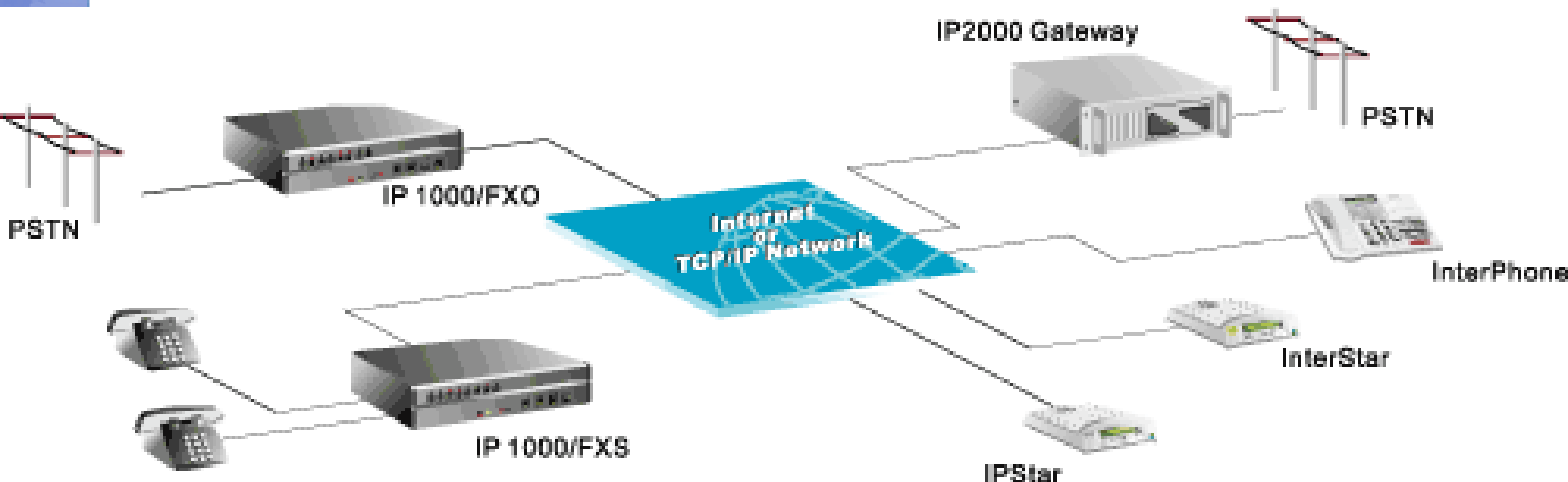
IP1000-ITSP mode



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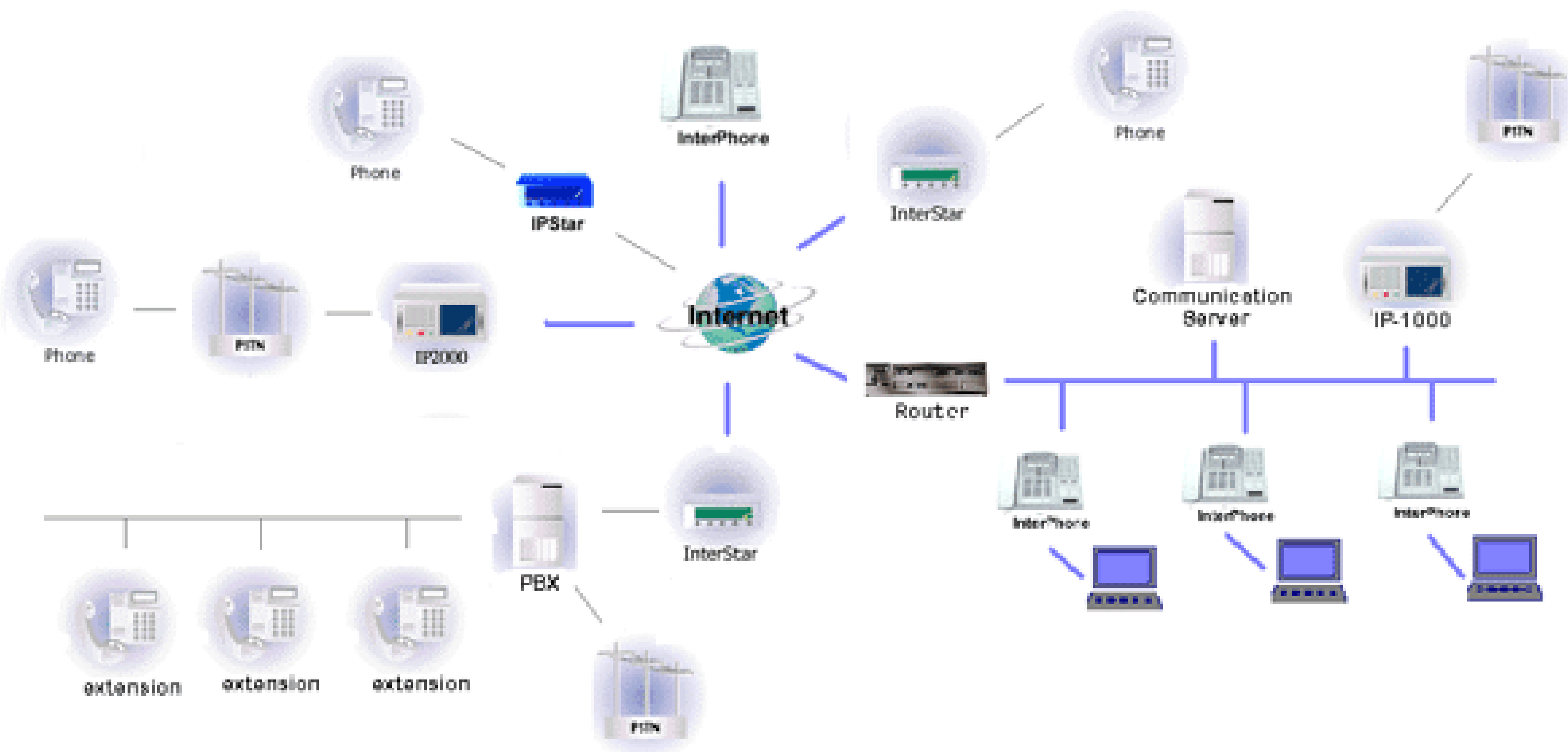
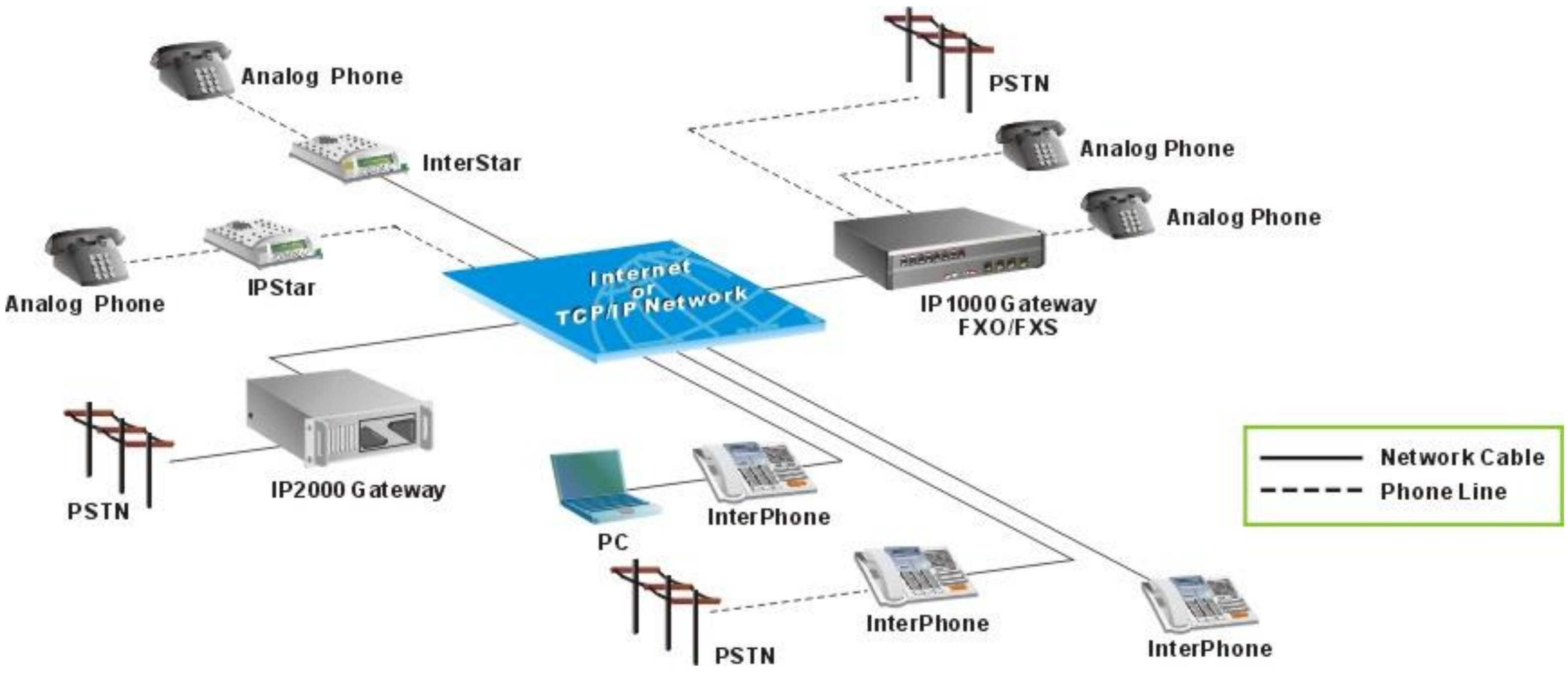
DSG Total VoIP Solution



DSG Network Connection

▶ *Connection:*



Installation Issues

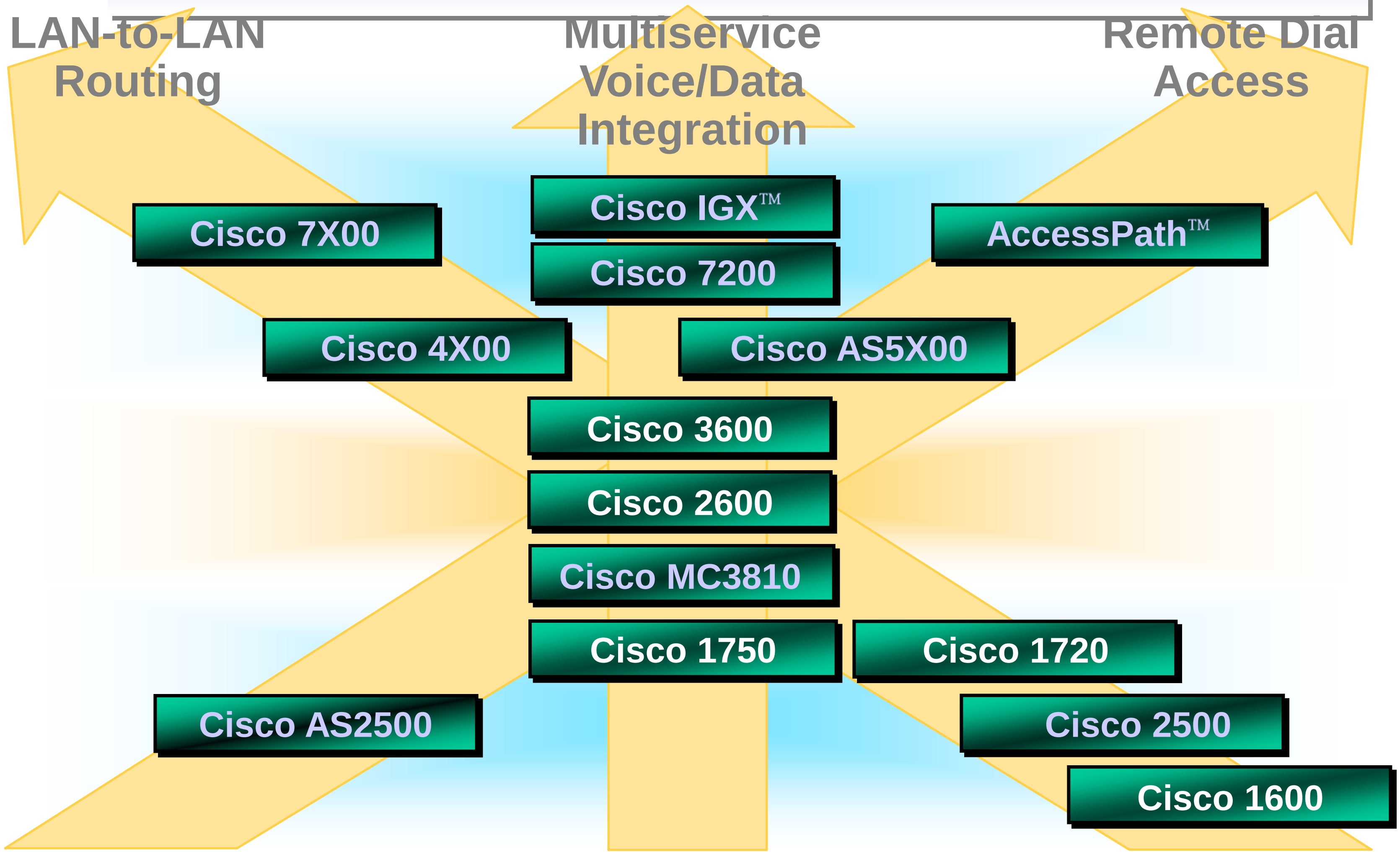
- Customer PBX:
  - Pulse or Tone ?
  - busy tone type?
- Internet Connection:
  - Bandwidth ?
  - Simultaneous data & voice?
- Billing:
  - Per user
  - Per device

Installation Issues

- *Lines:*
  - *Internal*
  - *Direct*
- *Internet Connection:*
  - *Bandwidth ?*
  - *Simultaneous data & voice?*
- *Billing:*
  - *Per user -> IP2000*
  - *Per device -> IP1000 & InterPhone*

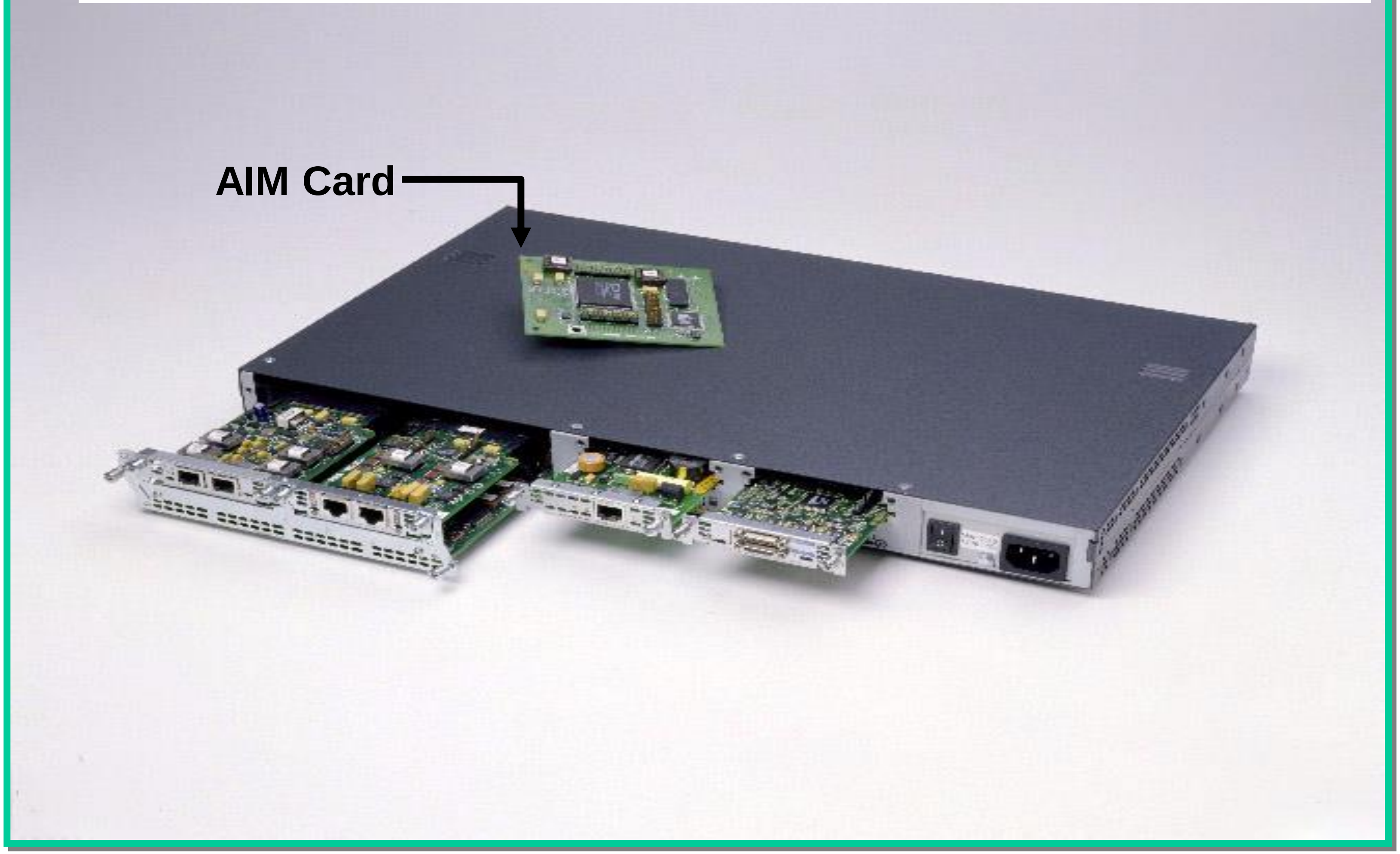
Cisco VoIP Products

Cisco Product Portfolio



Cisco 1750 Router

Cisco 2600 Series



Cisco 3600 Family Platforms

3660

3640

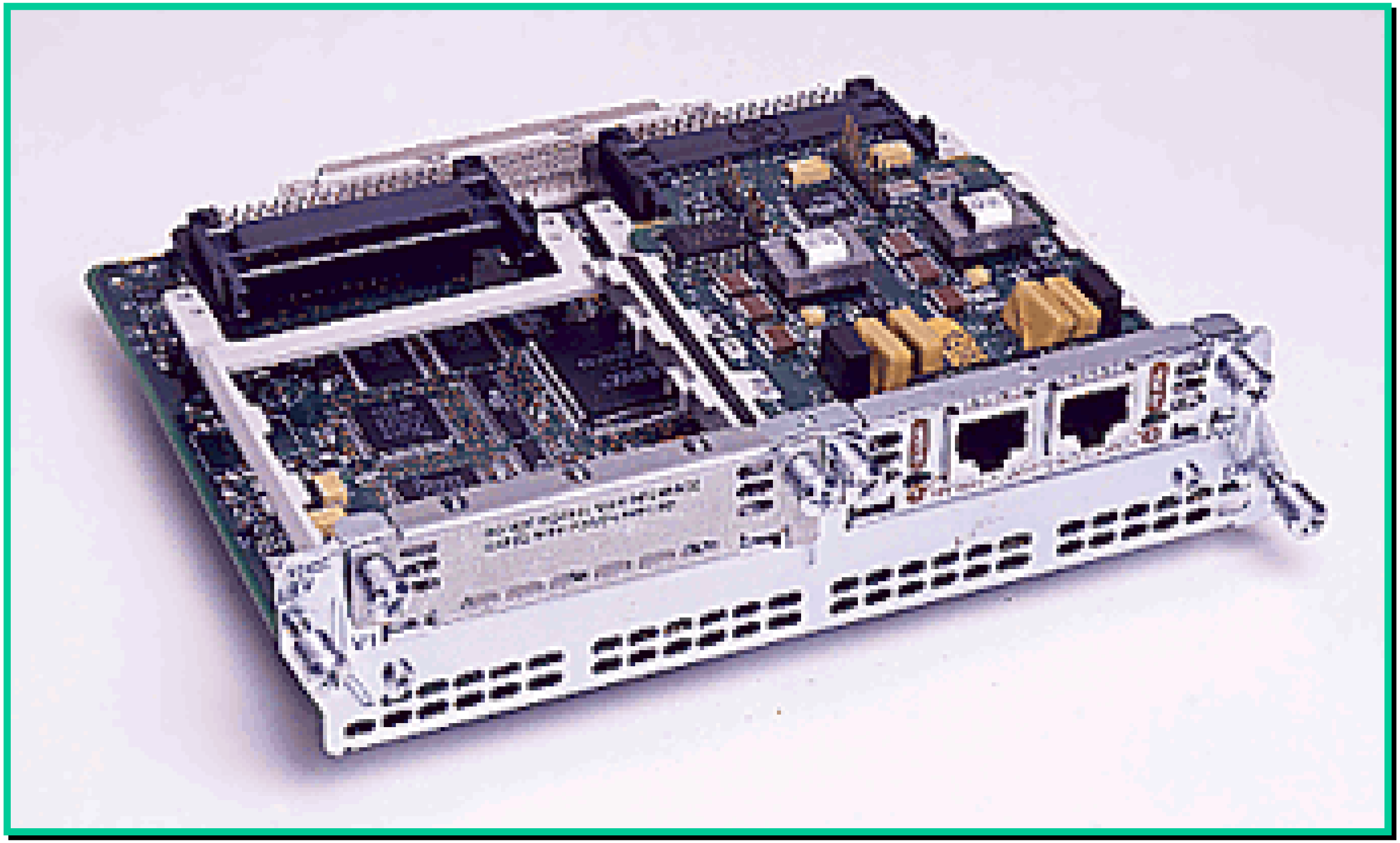
▶ *3620*

Cisco 3620 Modular Access Router

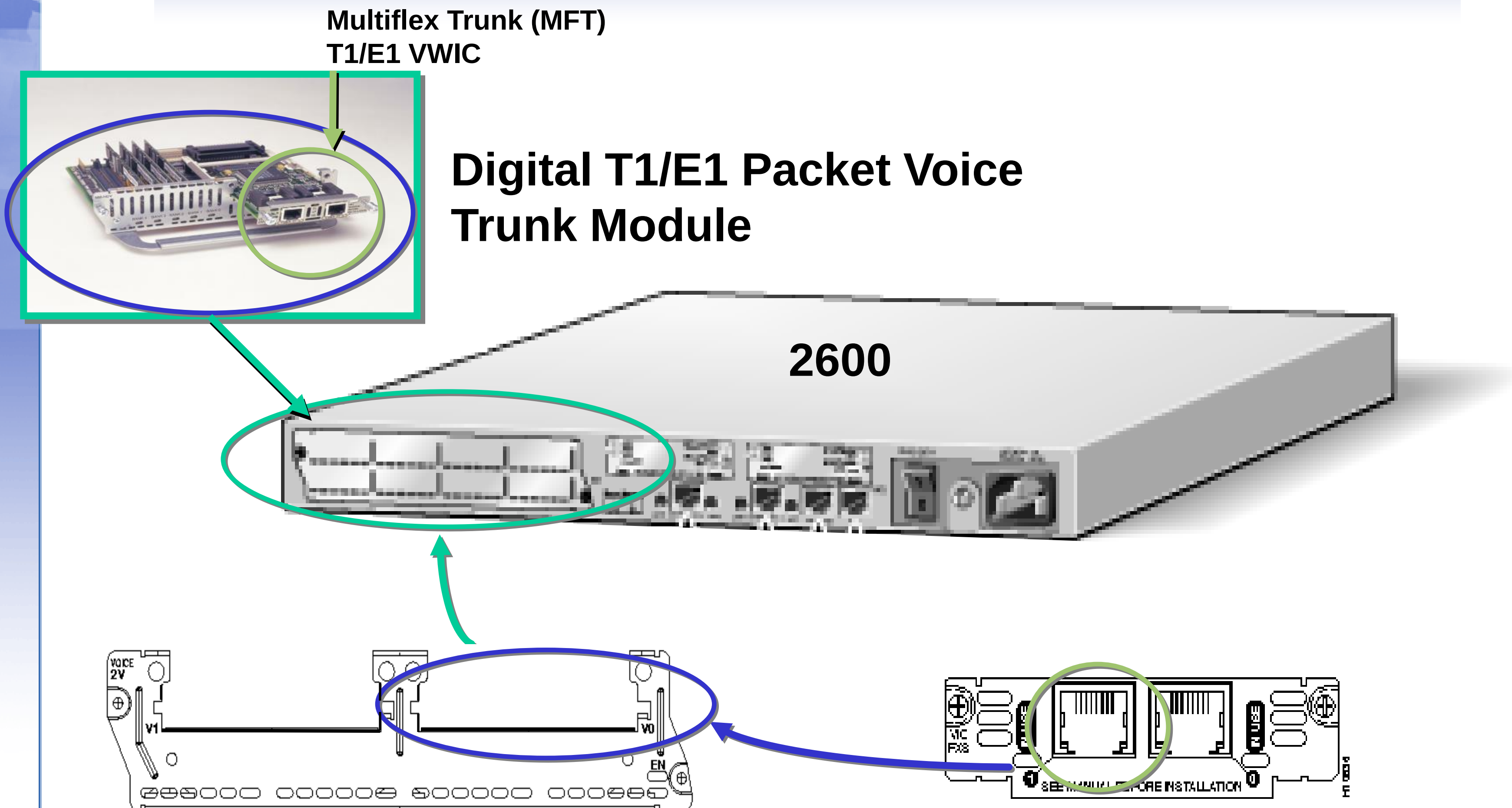
Cisco 3640 Modular Access Router

Cisco 3660

Voice/Fax Network Module



VICs, VWICs, and VNMs



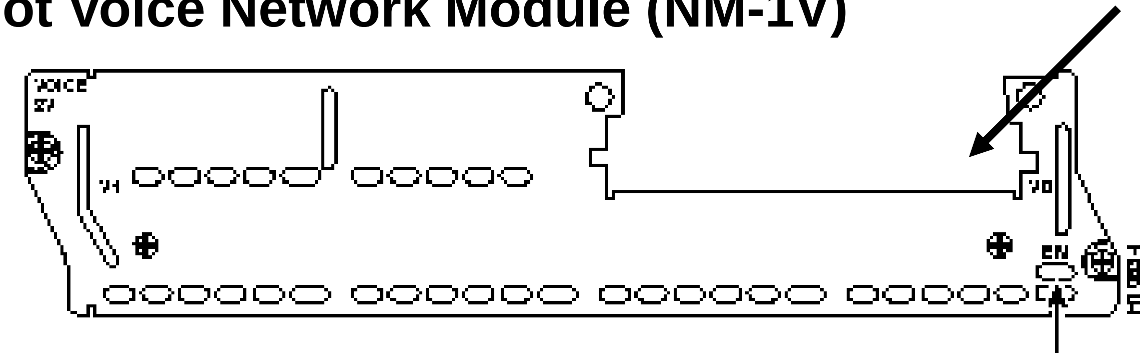
Voice Network Module

Voice Interface Card (analog)

Selecting the Voice Network Module

One-Slot Voice Network Module (NM-1V)

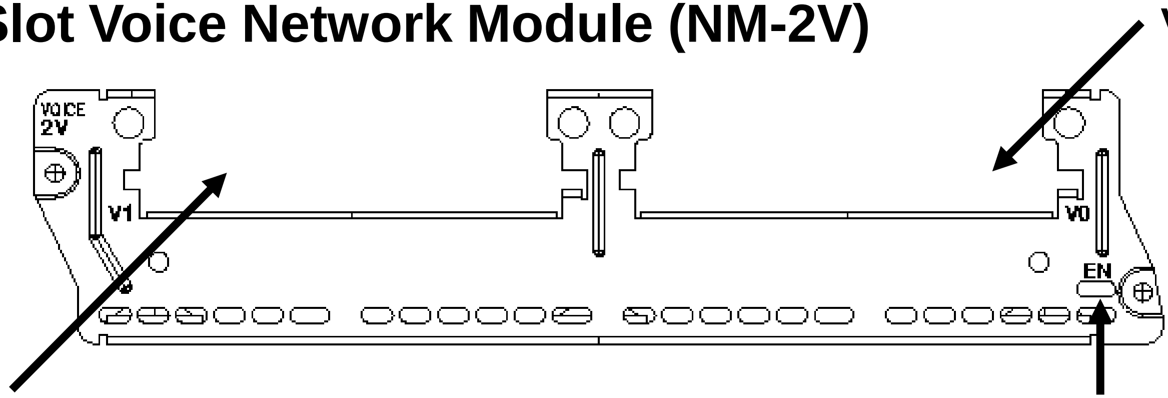
Voice slot V0



Enable LED

Two-Slot Voice Network Module (NM-2V)

Voice slot V0

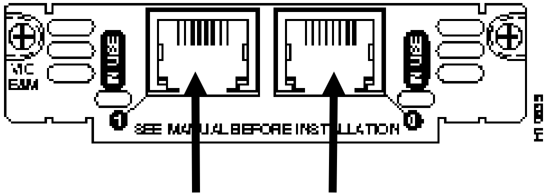


Voice slot V1

Enable LED

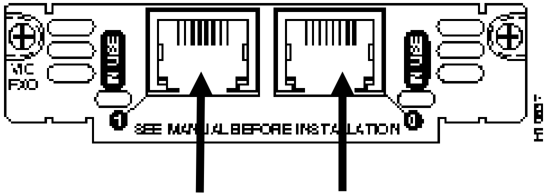
Selecting the Voice Interface Card

E&M VIC



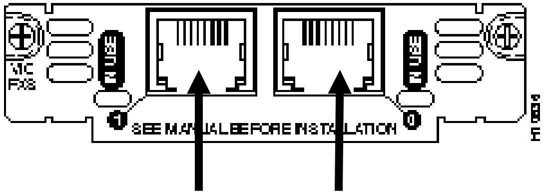
Voice port 1 Voice port 0

FXO VIC



Voice port 1 Voice port 0

FXS VIC

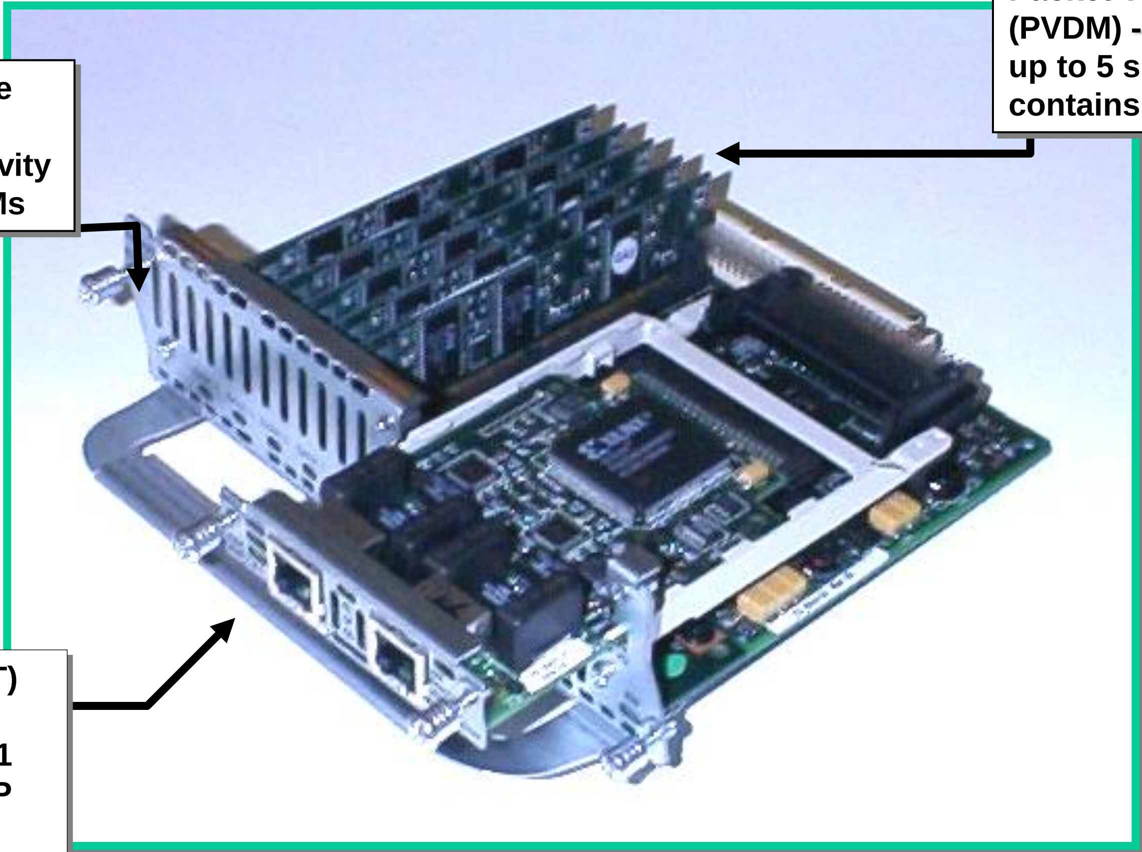


Voice port 1 Voice port 0

Digital T1/E1 Packet Voice Trunk Module

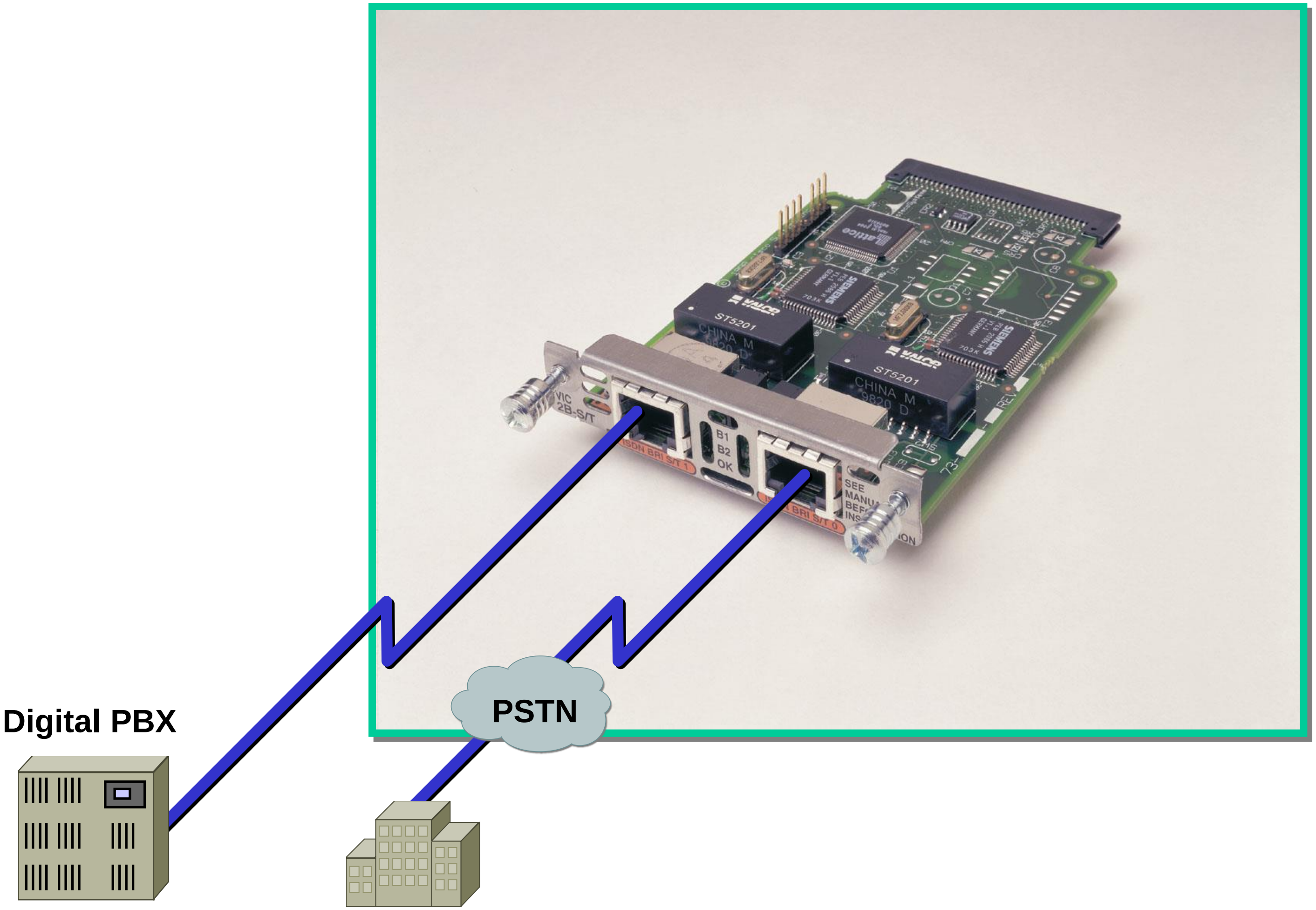
High-Density Voice (HDV) Module - provides connectivity for MFT and PVDMs

Packet Voice DSP Module (PVDM) - up to 5 slots; each PVDM contains 3 TI-549 DSPs

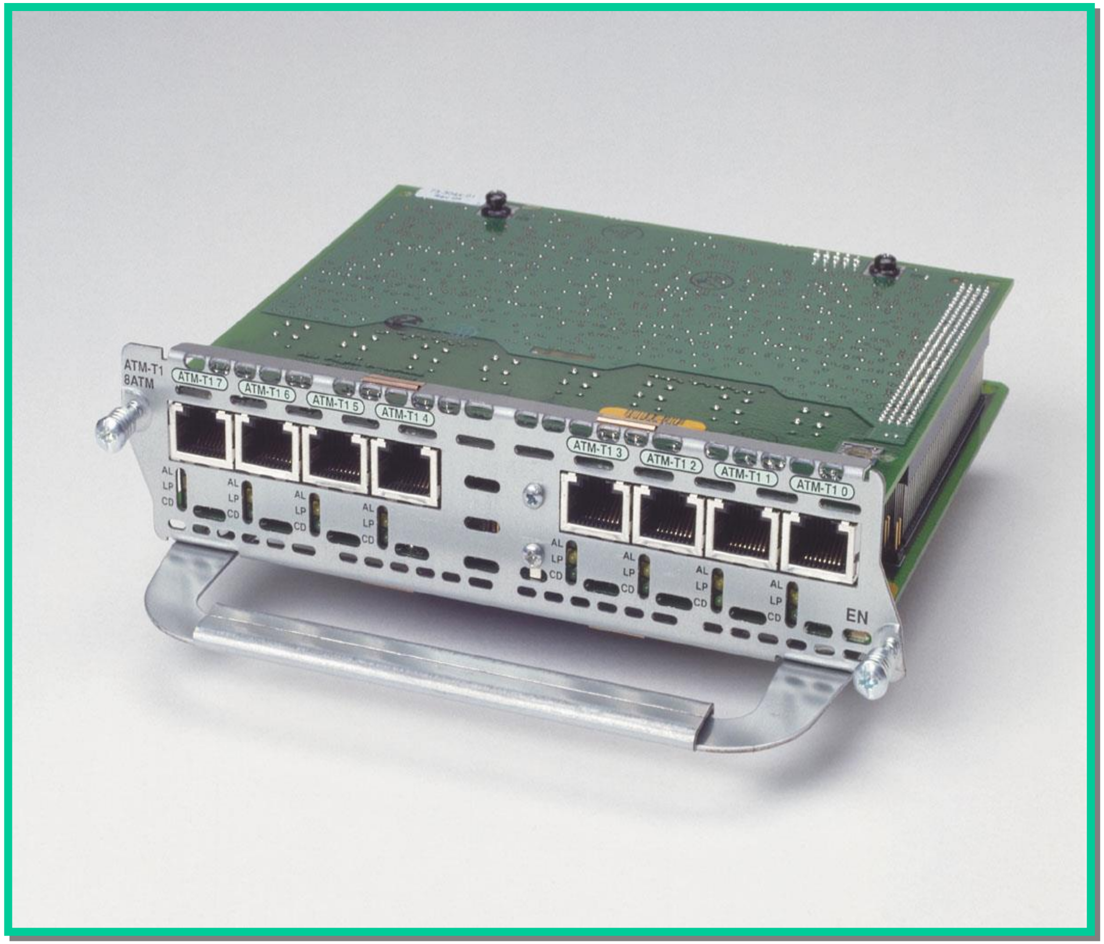


Multiflex Trunk (MFT) T1/E1 VWIC - provides physical T1 connectivity for VoIP or Drop & Insert

BRI Voice Interface for C2600/C3600



Multiport T1/E1 ATM IMA for 2600/3600



Cisco MC3810 Platform

3810 DSP Refresh

Configurations:	Slot 2	Channels	Slot 5	Channels	Total Channels
Analog, 4 ports G.729	HCM2	4	Empty	0	4
Analog, 6 ports G.729a	HCM2	6	Empty	0	6
Digital T1 G.729	HCM6	12	HCM6	12	24
Digital T1 G.729a	HCM6	24	Empty	0	24
Digital E1 G.729	HCM6	12	HCM6	12	24
Digital E1 G.729a	HCM2	8	HCM6	24	24

Cisco AS5300 Platform

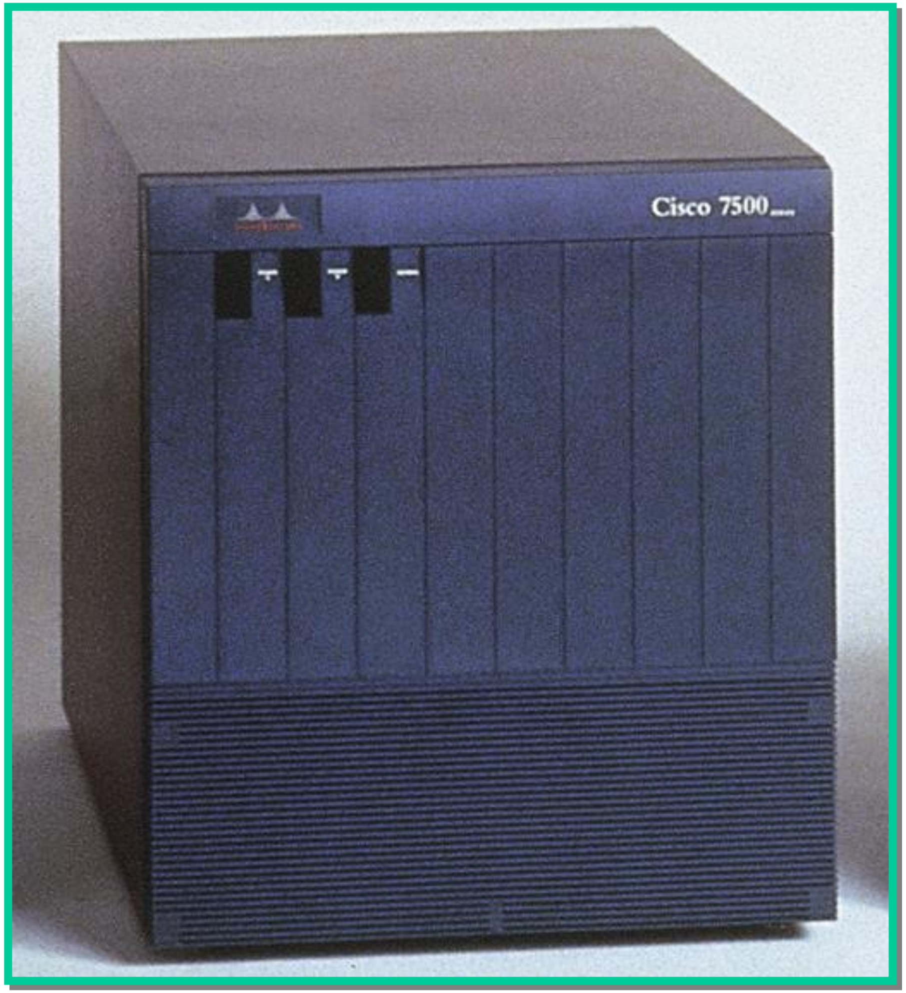
Product Line Overview—Data/Voice Integration

	Cisco AS5300	Cisco MC3810	Cisco 3600	Cisco 2600
Cisco IOS LAN-to-LAN Routing	Yes	Yes	Yes	Yes
Voice over IP	Yes	Yes	Yes	Yes
Voice over Frame Relay	No	Yes	Yes	Yes
Voice over ATM	No	Yes	Yes	Yes
ATM WAN Access	No	Yes	Yes	Yes
Integrated Dial	Yes	No	Yes	Yes
LAN Media	E/FE	E	E, TR, FE	E, TR, FE
WAN Media	PRI	Serial, BRI, ATM	PRI, MBRI, Serial, ATM	PRI, MBRI, Serial, ATM
Maximum Analog/ Digital Voice Density	0/120	6/24	24/180	4/30

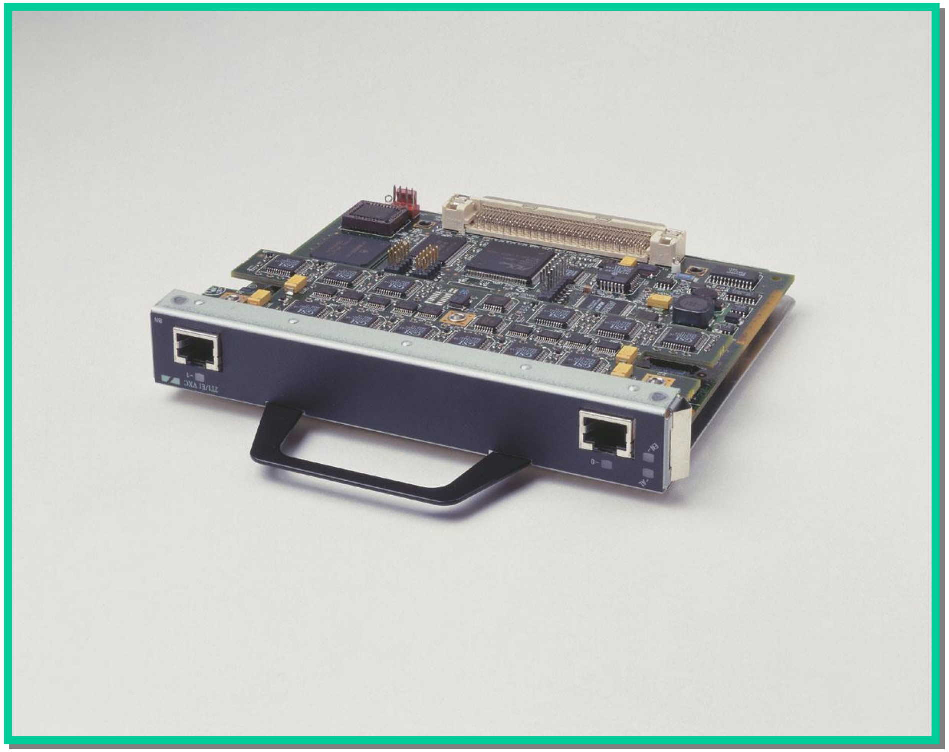
Cisco AS5800 Voice Gateway

Cisco 7200

Cisco 7500



Digital T1/E1 Voice Port Adapter for Cisco 7200 and 7500



DPI DC

Cisco IP Phones

7910

7914

7935

7940

7960

SoftPhone

Q&A

سؤال

جواب

Putting It All Together: Building A VoIP Network And Making A Call

Objectives

- *At the completion of this section you will:*
- *Know some general business questions you may ask depending on your business model*
- *Understand the role of the gateway and gatekeeper in a VoIP network*
- *Know how H.323 relates to a VoIP call over an IP network*
- *Understand the amount of protocol overhead needed to create a VoIP packet*
- *Understand all the steps necessary to complete a phone-to-phone call over a VoIP network*

Section Topics

- General Business/Service Questions
- Building a VoIP Network
  - Network equipment
  - Network transport options
- Making a Call
  - Call setup
  - Voice transmission

General Business/Service Questions

- *What's the objective?*
- *What problem are you trying to solve?*
- *Private or public applications?*
- *Low cost or enhanced function?*

General Business/Service Questions Internet Telephony Service Provider

- Market focus?
  - » Business or residential customers
  - » Cheap international voice/fax only
  - » ISP with voice/fax capability
  - » Facilities-based carrier providing full-service network
- Global footprint?
  - » Which markets to serve
  - » Build own network
  - » Join ISTP consortium
- Service quality?
  - » You get what you pay for
  - » Service guarantees
- Backbone options?
  - » Internet
  - » Virtual networks
  - » Dedicated lines

General Business/Service Questions Private Network

- *Overlay of existing data network?*
- *Applications?*
  - *Intra-location voice/fax*
  - *Non-interactive, real time (e.g., voice mail, integrated voice response)*
- *Locations?*
  - *Branch vs. HQ*
- *Is it really cheaper then voice VPN?*
  - *Maybe yes for international, but no for domestic*
- *Will users accept quality drop?*
- *Can priority mechanisms be set in*
- *existing network equipment?*

VPN = Virtual Private Network

Gateway Questions

- *Architecture*
  - » *PC card*
  - » *Router card*
  - » *Stand alone chassis*
- *Price per port*
- *PSTN port types?*
  - » *Analog, T1/E1, ISDN*
- *Support of PSTN protocols?*
  - » *SS7, Intelligent Networks*
- *Router functions incorporated?*

PSTN = Public Switched Telephone Network
ISDN = Integrated Services Digital Network

Gateway Questions (continued)

- *Call control Protocol?*
  - » *H.323*
  - » *SIP*
  - » *MEGACO*
  - » *Extensible to accommodate changing standards*
- *Fax support?*
  - » *Real-Time*
  - » *Store-and-forward*
- *Pass DTMF?*
- *Pass dial-up modem?*
- *Interoperability demonstrated with other vendors?*

DMTF = Dual Tone MultiFrequency

SIP = Session Initiation Protocol
MEGACO = Media Gateway Control

Gateway Questions (continued)

- *Voice coding algorithms?*
  - *G.723.1 for low bit rate voice/video*
  - *G.729 for voice only*
  - *G.72x or proprietary for performance, delay, modems*
- *Voice frame size*
  - *Performance vs. efficiency*
  - *Tunable*
- *Jitter buffer?*
  - *Delay range*
  - *Automatic vs. manual setting*
  - *Tunable*
- *Processing delay*
- *Reliability*
- *Priority requests compatible with backbone routers (e.g., RSVP)?*

RSVP = Resource Reservation Protocol

Gatekeeper Questions

- *Do you need Them?*
- *How many?*
- *H.323 V2 compliant?*
- *Performance?*
  - » *e.g., How many call setups per minute*
- *Extensible to support PSTN protocols?*
  - » *e.g., SS7/IN*
- *Authorization and billing functionality?*
- *Vendor interoperability demonstrated?*

PSTN = Public Switch Telephone Network

Network Backbone Options
Public Internet

- *Buy access from ISP at each gateway location*
- *Lowest cost*
- *Poorest performance*
- *May be appropriate for cheap international*
- *calling business model*

Network Backbone Options Dedicated Facilities

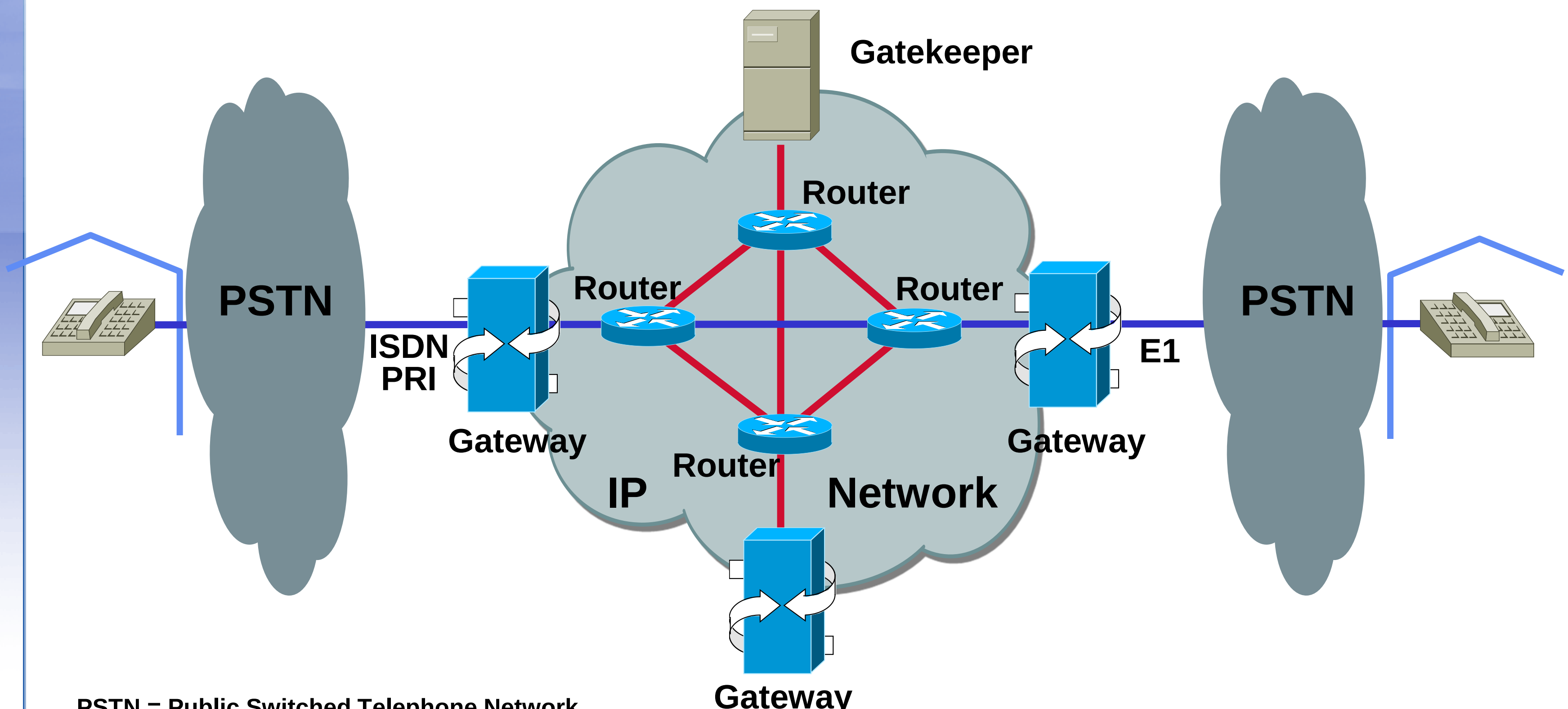
- *Company-owned or leased line from telco*
- *Best control over network performance*
- *Highest-cost infrastructure*
- *Best position for service carrier*
- *Opens additional questions about backbone*
  - *IP routers or Frame/ATM virtual circuit networks?*
  - *Voice/fax only or multi-service?*
  - *Adequate priority mechanisms within switching platforms?*

Network Backbone Options
Virtual Network From Facilities-Based Carrier

- *IP VPN, Frame Relay, ATM*
- *Typically lower cost then dedicated facilities*
- *IP routers at edge of virtual network*
- *Performance guarantees?*
- *Priority mechanisms?*
- *ISP in addition to voice/fax?*

VPN = Virtual Private Network
ATM = Asynchronous Transfer Mode

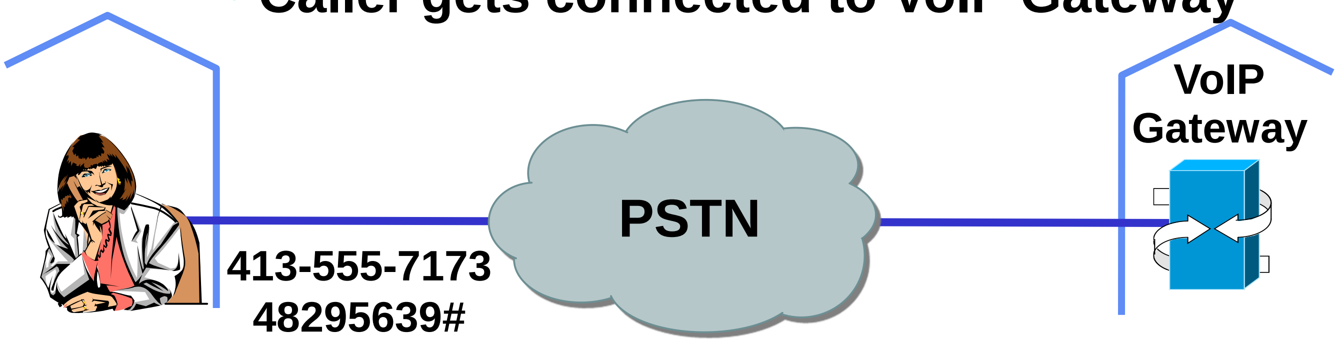
VoIP Network Example
Phone-To-Phone Service



PSTN = Public Switched Telephone Network
ISDN = Integrated Digital Network
PRI = Primary Rate Interface

IP Telephony Service Provider Use of H.323

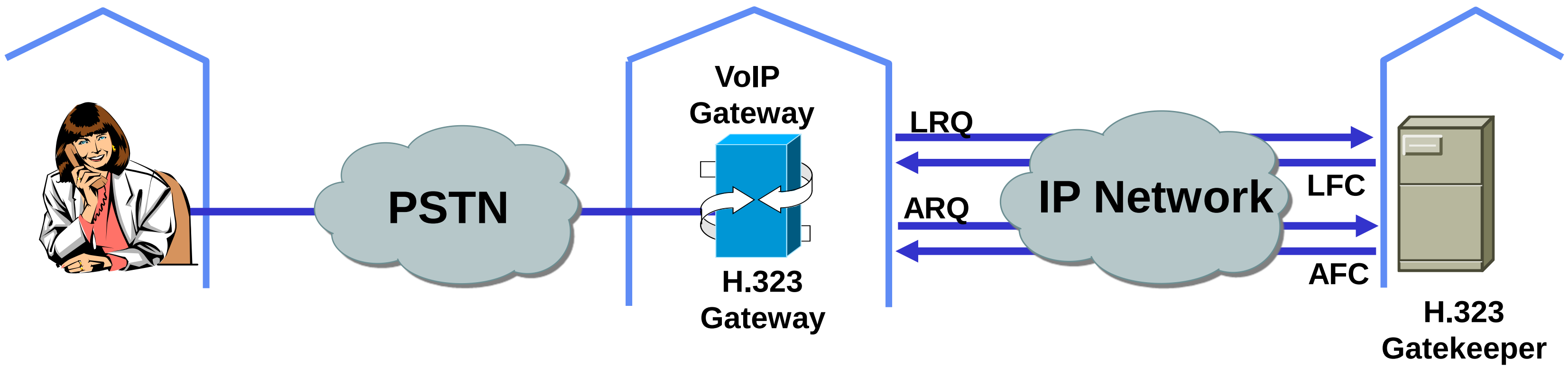
- Caller dials access number for ITSP
- Caller gets connected to VoIP Gateway



- Caller is prompted for destination Telephone number and account #/PN

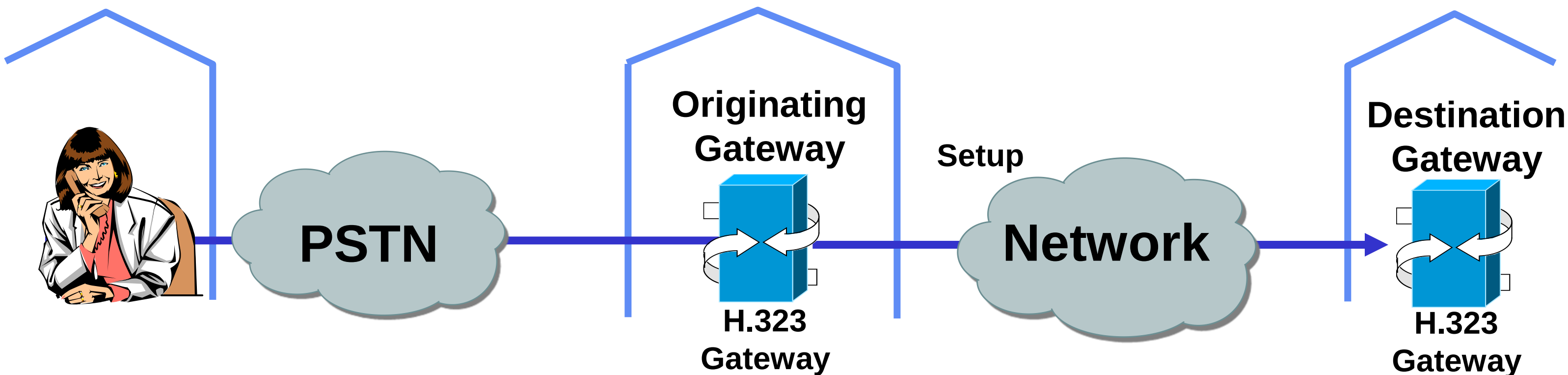
VoIP = Voice over IP
PIN = Personal Identification Number

IP Telephony Service Provider Use of H.323 V2 (Continued)



- Gateway makes request of Gatekeeper for IP address of destination Gateway based on telephone number
  - Location Confirm: 204.124.46.19
- Gateway makes request of Gatekeeper to establish call and use bandwidth
  - Admission Confirm: 16 Kb/s

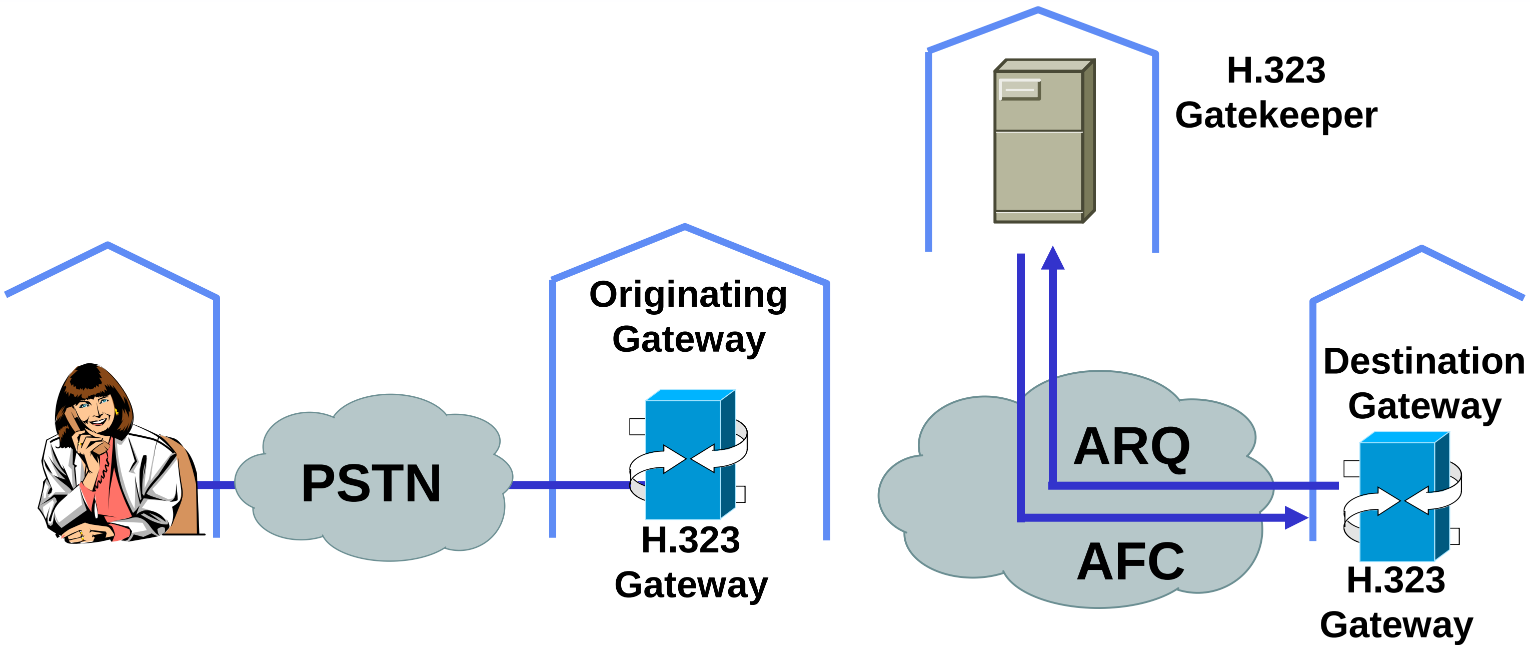
IP Telephony Service Provider Use of H.323 V2 (Continued)



- Originating Gateway sends Setup message to destination gateway using IP address received from gatekeeper:

Setup: 413-555-7173	Destination Telephone #
312-555-7083	Originating Telephone #
188.140.76.19	IP Address of originating Gateway
G.729 Audio	faststart element to open audio channel(s)

IP Telephony Service Provider Use of H.323 V2 (Continued)

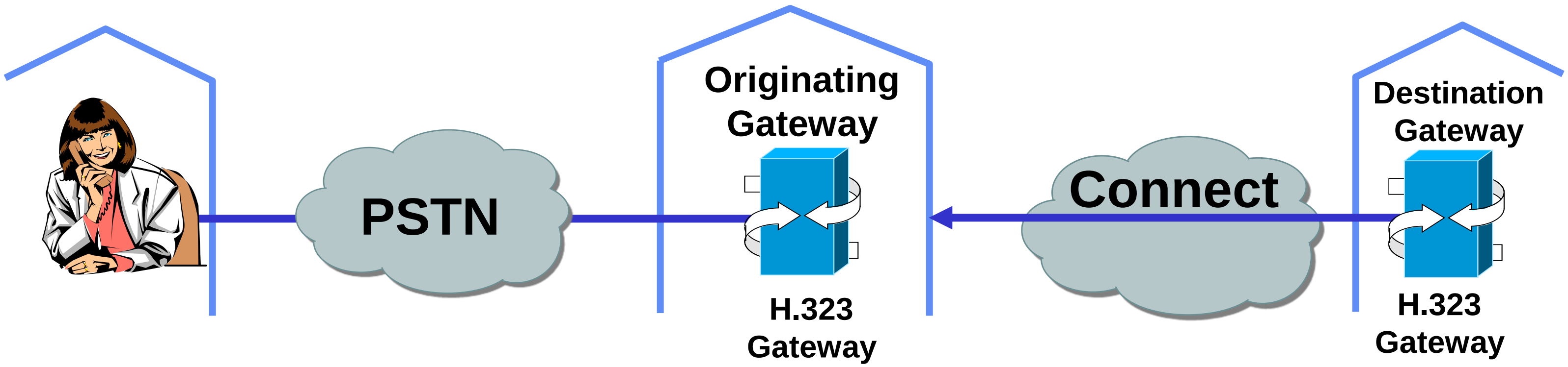


- Destination gateway makes request of gatekeeper to accept call from originating gateway, use bandwidth:

  Admission Request : 188.140.76.19,16 Kb/s

- Gatekeeper confirms request

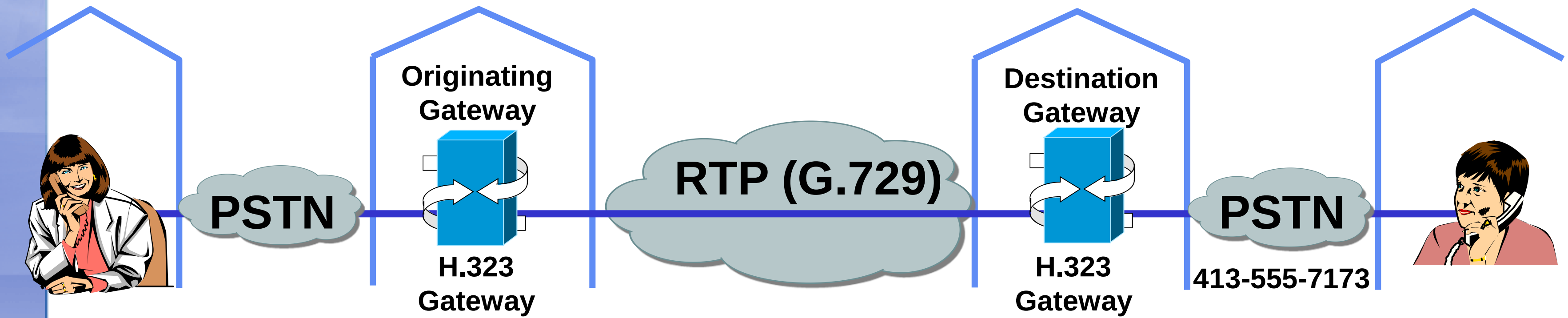
IP Telephony Service Provider Use of H.323 V2 (Continued)



- Destination Gateway responds to originating Gateway Once permission is confirmed, establishes connection:

Connect: fastStart element confirms requested audio channel(s)

IP Telephony Service Provider Use of H.323 V2 (Continued)



- Destination Gateway originates call over PSTN to destination Telephone number

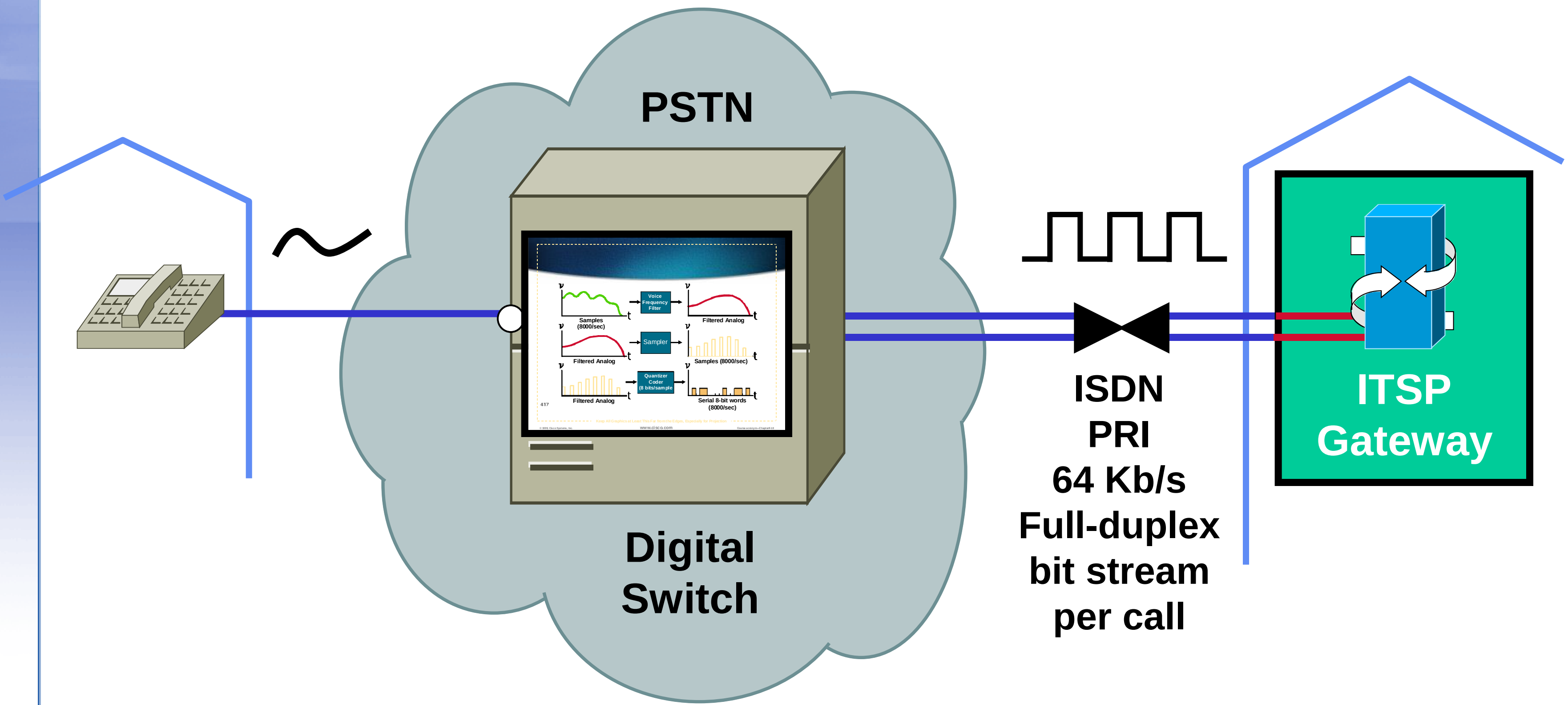
- Gateways establish internal connection between PSTN circuit and IP/H.245 logical channel for audio

RTP = Real-time Transport Protocol

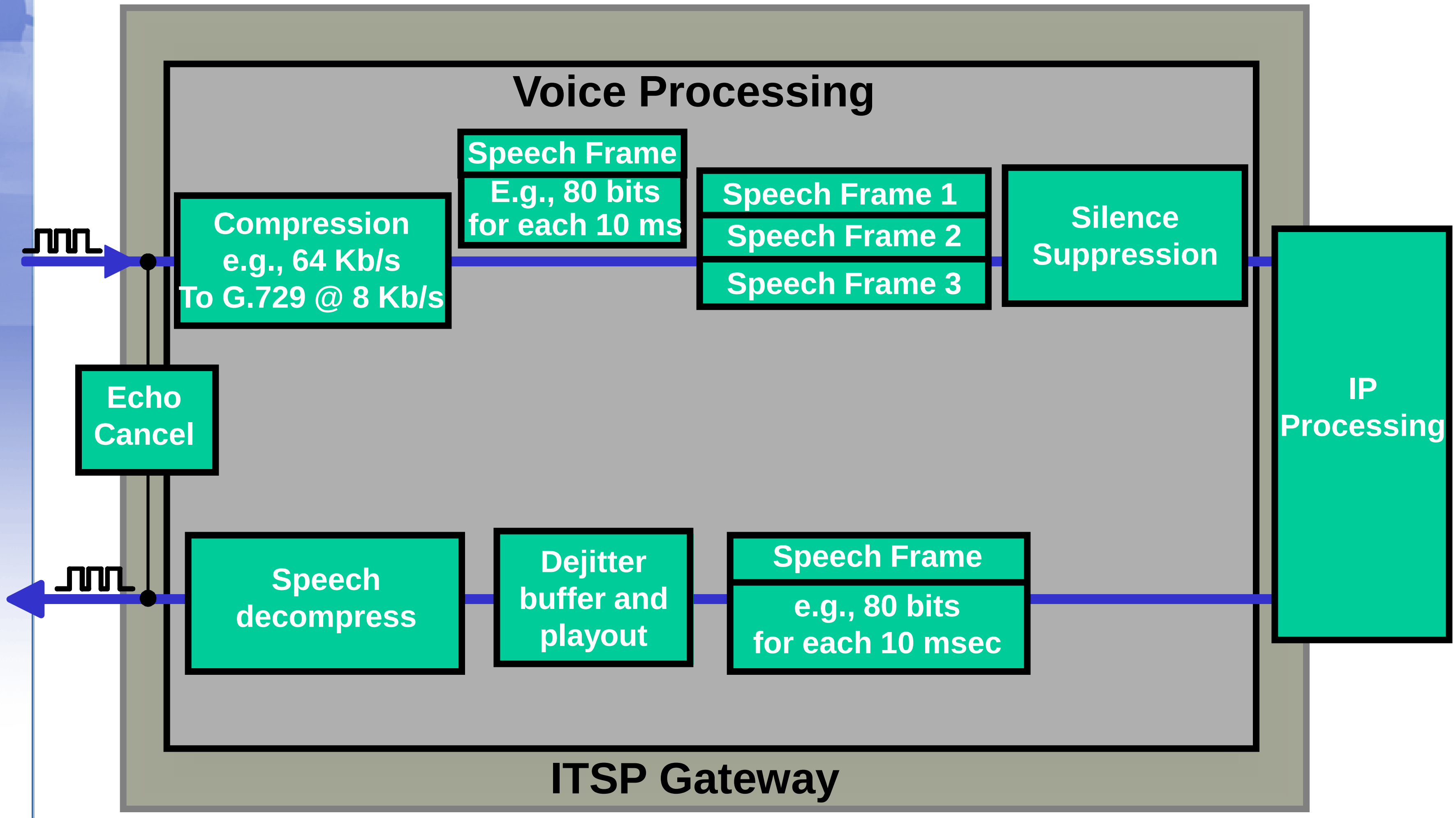
Voice Transmission Example
Phone-To-Phone Call Over VoIP Network

1. Analog voice sent from telephone set to local Central Office
2. Local Switch converts analog signal to A-law/Mu-law PCM and transmits signal as 64 Kb/s bit stream to the Gateway
3. Gateway receives 64 Kb/s bit stream
   - Compresses speech using G.72x coding
   - Silence suppression
   - Echo cancellation
4. One or more compressed speech frames accumulated
   - Transmit individual speech frames for minimum delay
   - Accumulate multiple speech frames for greater bit efficiency

Voice Transmission Example
Phone-To-Phone Call Over VoIP Network
Steps 1 and 2



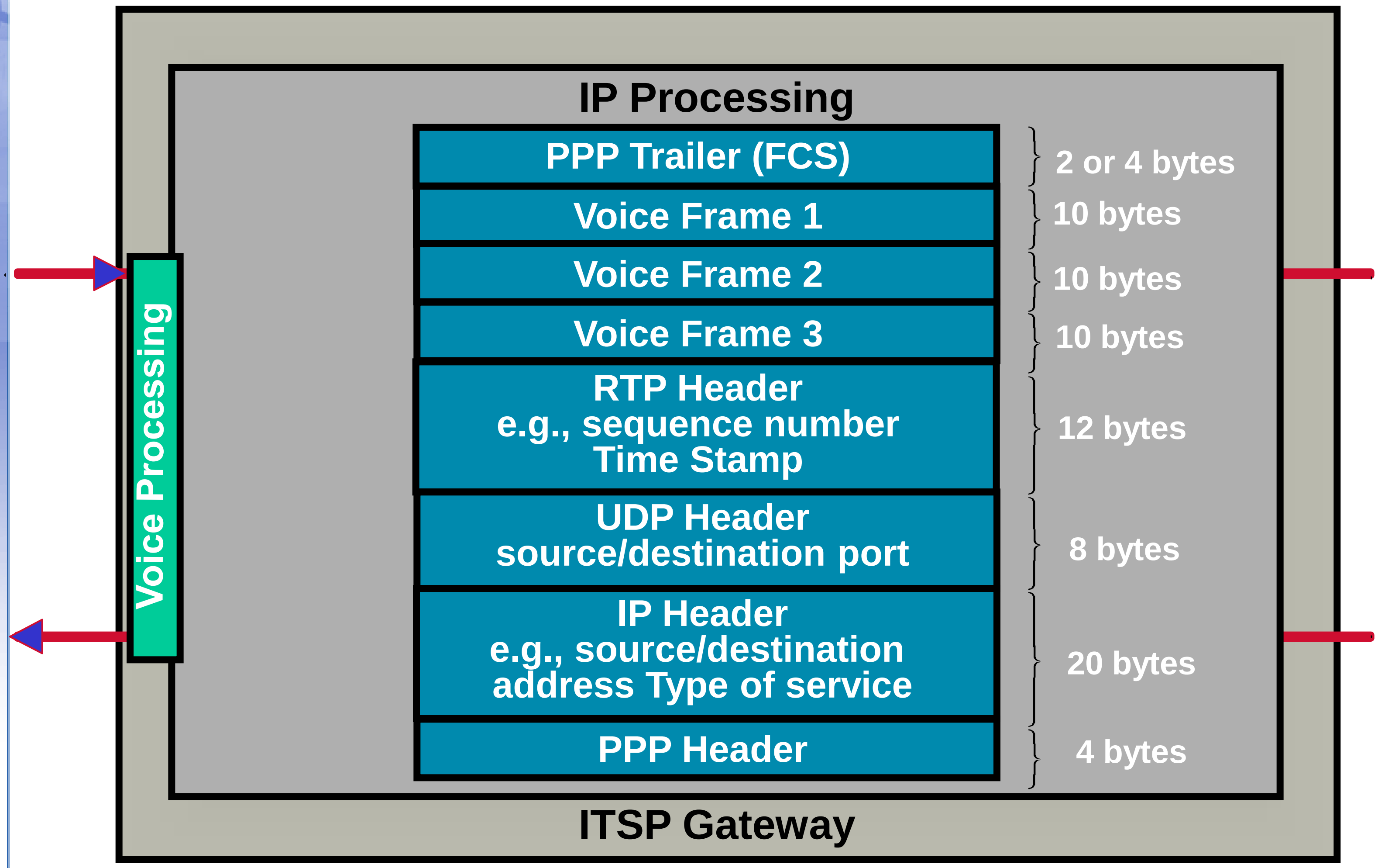
Voice Transmission Example
Phone-To Phone Call Over VoIP Network Steps 3 And 4



Voice Transmission Example Phone-To-Phone Call Over VoIP Network (continued)

5. Voice encapsulated in IP protocol stack
    - RTP for source ID, time stamp, sequence number
    - UDP to identify source/destination ports
    - IP for addressing/forwarding
    - PPP, Frame Relay or ATM at the link layer
    - Header compression may be applied on low bandwidth links
6. Voice packets sent as accumulated
    - Priority over data?
    - Connectionless transport over IP network?
    - Guaranteed maximum delay across network?
7. Voice packets arrive at destination Gateway
    - IP/UDP header stripped off
    - RTP packet delivered to voice processor
8. RTP header info used to determine sequence and timing of voice playout
9. Voice samples placed in jitter buffer

Voice Transmission Example
Phone-To Phone Call
Over VoIP Network Step 5

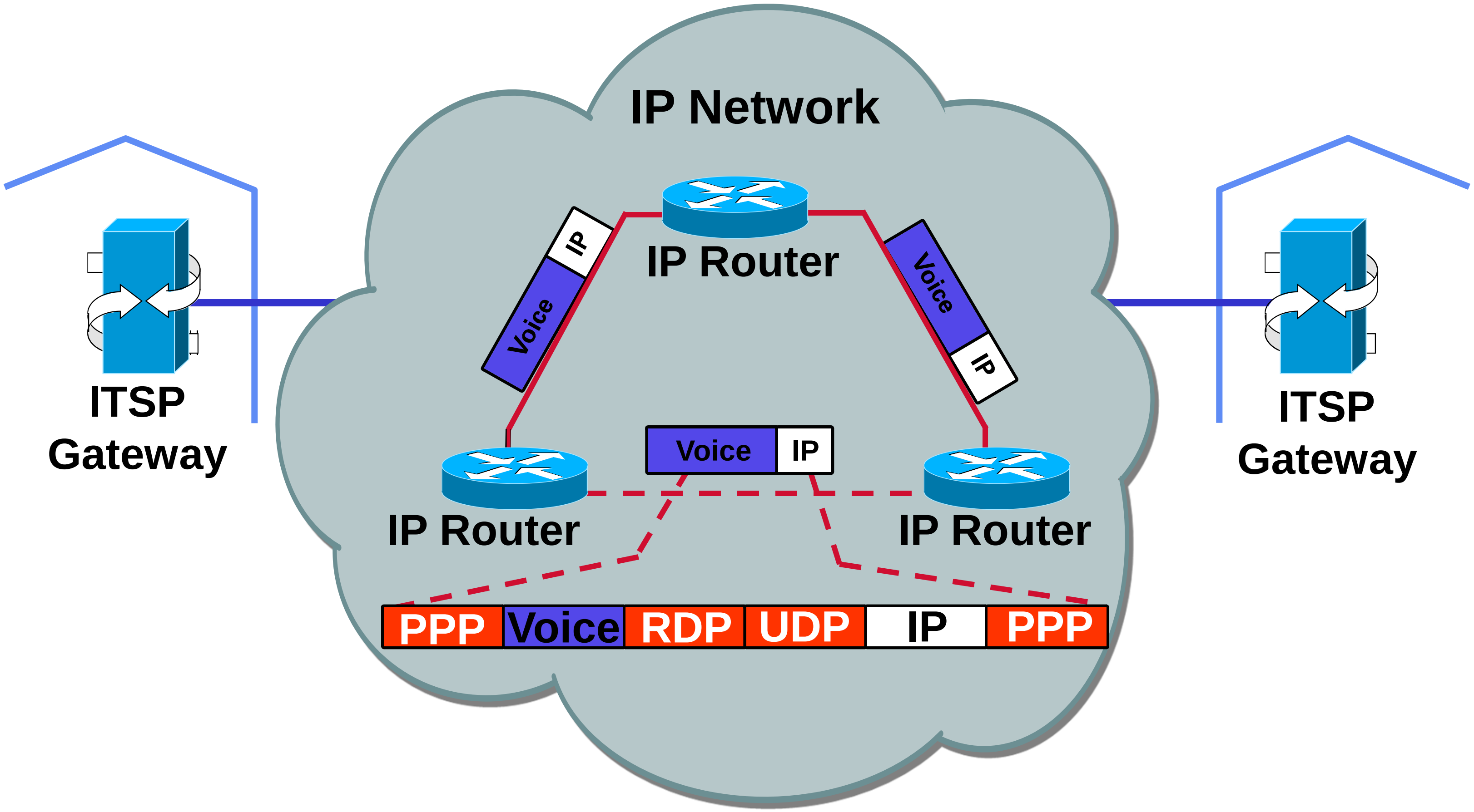


What's The Protocol Overhead?

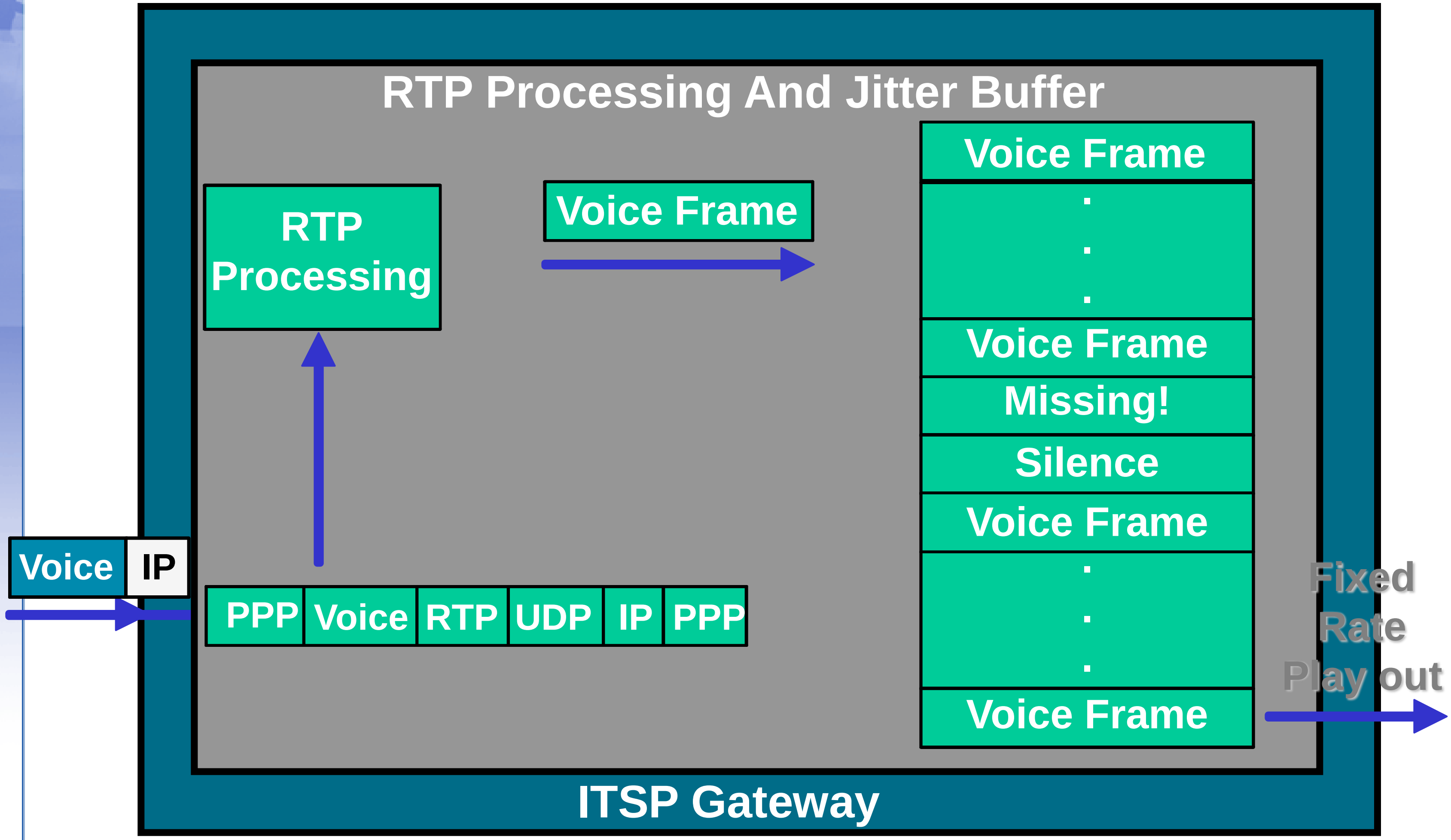
PPP Trailer (FCS)	2 or 4 bytes
Voice Frame 1	10 bytes
Voice Frame 2	10 bytes
Voice Frame 3	10 bytes
RTP Header e.g., sequence number Time Stamp	12 bytes
UDP Header source/destination port	8 bytes
IP Header e.g., source/destination address Type of service	20 bytes
PPP Header	4 bytes

- *For a 30ms voice multi-frame using G.729*
  - *46 out of 76 octets = 60% overhead*
  - *20.26 Kb/s per voice call with no silence suppression or header compression*
- *For a 60 ms voice multi-frame using G.729*
  - *46 out of 106 octets = 43% overhead*
  - *14.1 Kb/s per voice call with no silence suppression or header compression*

Voice Transmission Example Phone-To-Phone Call Over VoIP Network Step 6

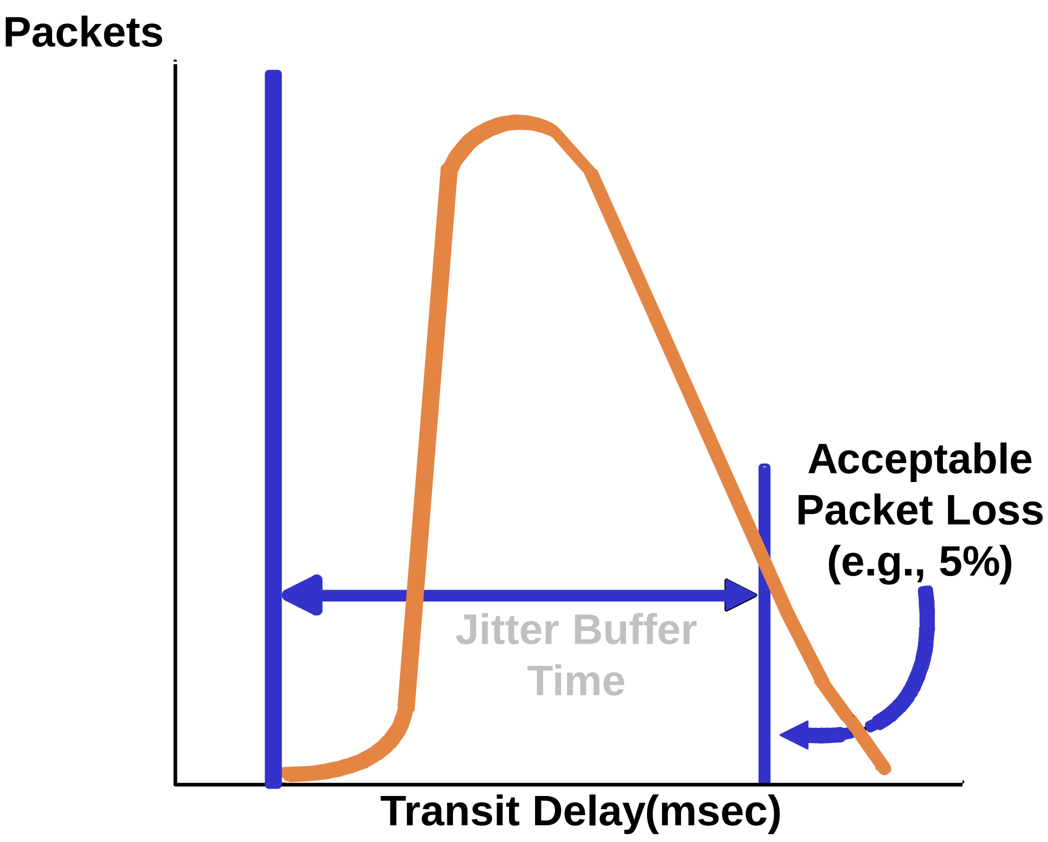


Voice Transmission Example Phone-To Phone Call Over VoIP Network Steps 7, 8, 9



How Do You Determine Jitter Buffer Size?

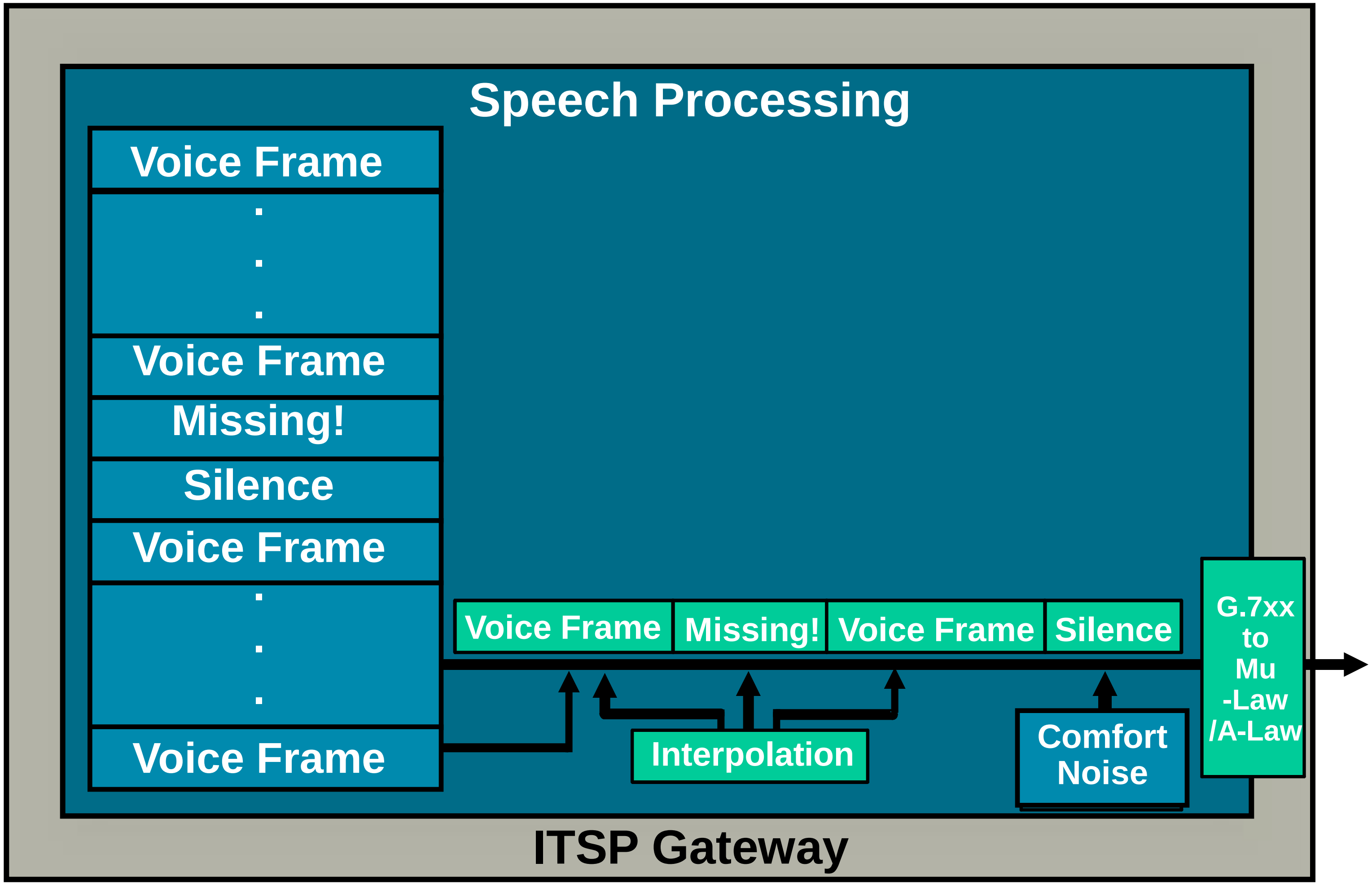
- *Estimate average transit delay*
- *Estimate transit delay distribution (e.g., how long for 95% of all packets to arrive)*
- *Determine acceptable packet loss ratio*
- *Set buffer to match acceptable loss ratio; watch performance for tuning*



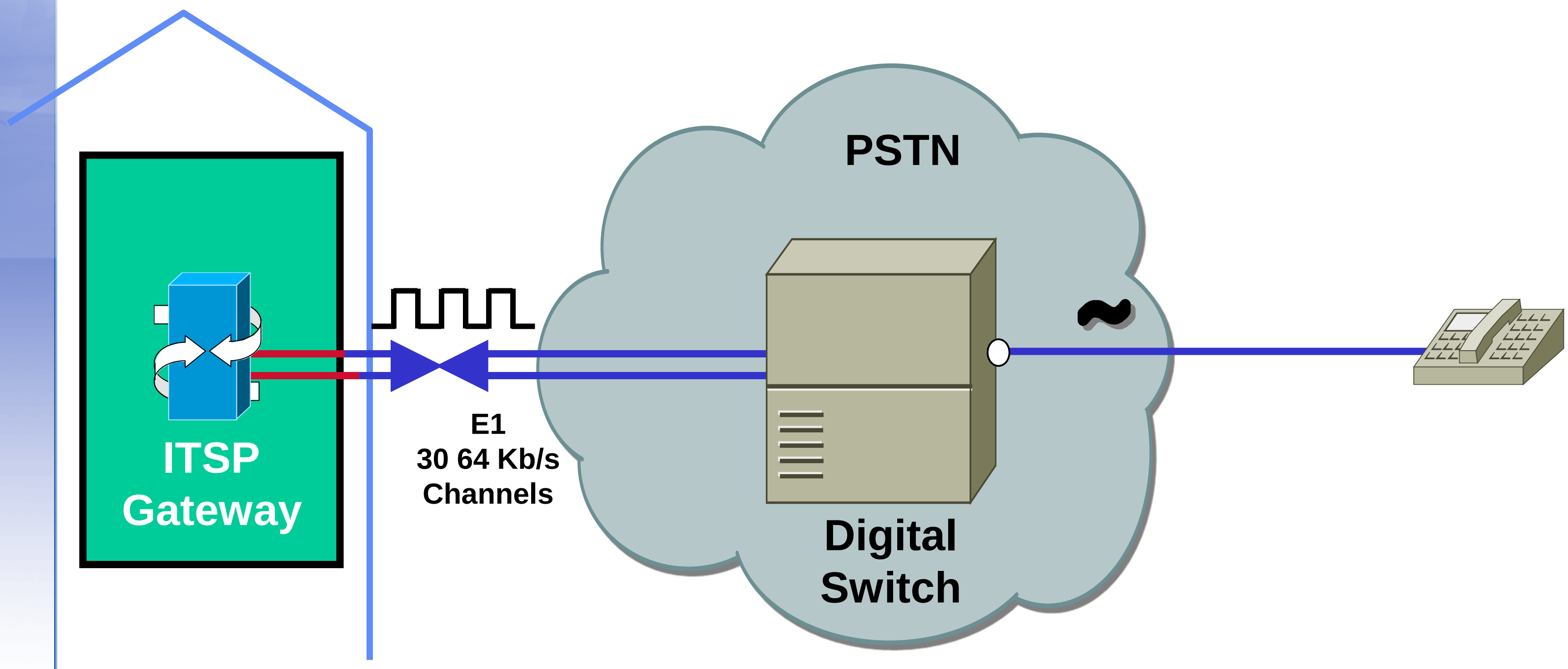
Voice Transmission Example Phone-To-Phone Call Over VoIP Network (continued)

10. Voice played out on cue
    - Interpolation, silence or repeat sample for missing speech
    - Insertion of “comfort” noise for silence
    - Conversion from compressed format to A-law/Mu-law PCM
11. Speech transmitted as 64 Kb/s bit stream to local PSTN switch
12. Speech converted to analog and sent to local subscriber by local PSTN switch

Voice Transmission Example Phone-To- Phone Call Over VoIP Network Step 10



Voice Transmission Example
Phone-To-Phone Call Over VoIP Network
Steps 11 And 12

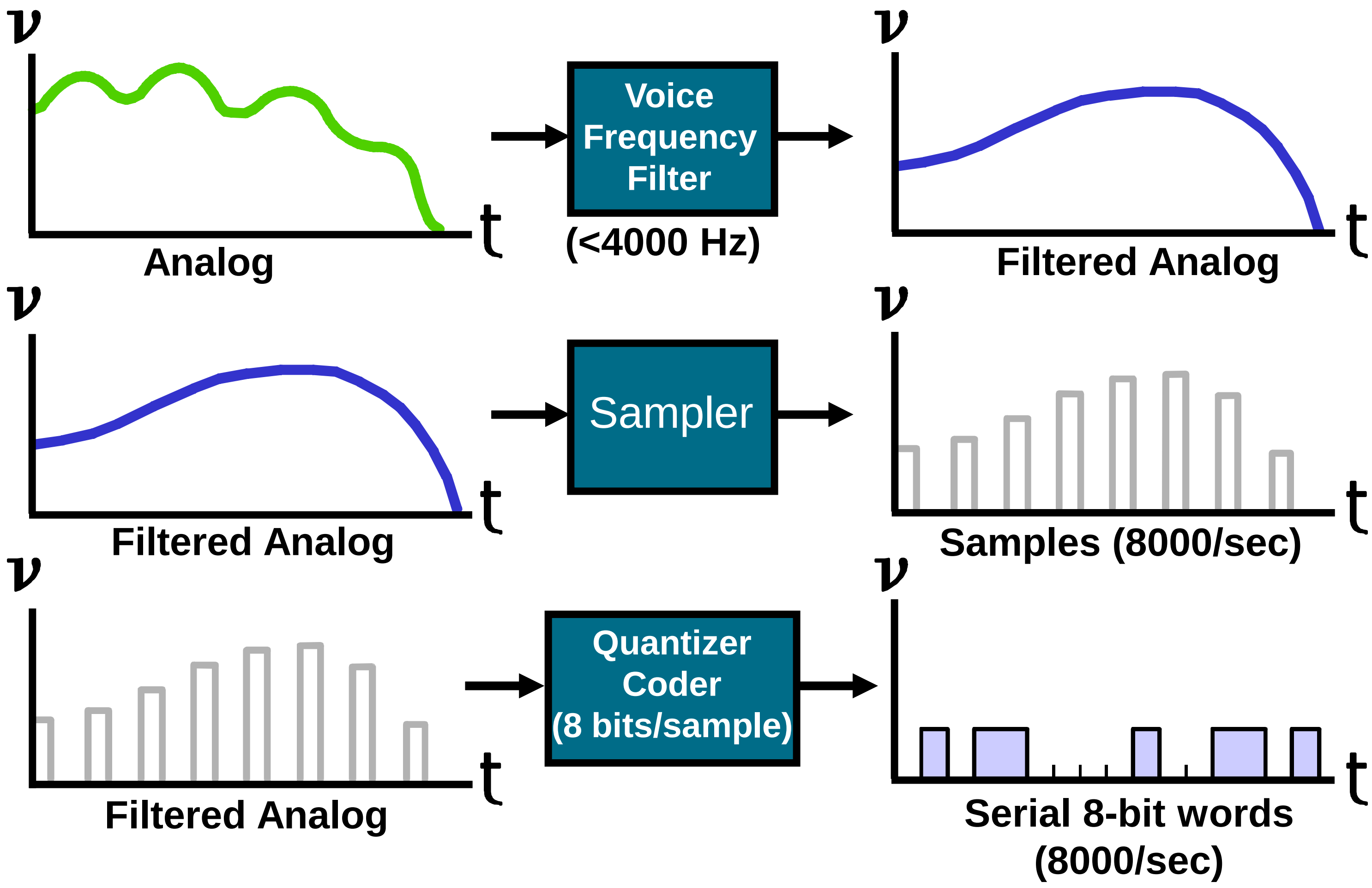


Section Summary

- Decisions on VoIP equipment and network design depend on the business purpose for the network
- Although standards are being developed, today most network equipment is not interoperable
- Connection quality can be improved by using a managed network rather then the Internet
- Call setup requires a multi-step processing between gateways, gatekeepers and the PSTN
- Speech transmission requires echo cancellation and involves compression schemes, delay management, efficiency, silence suppression, and quality

Technical Discussions

Digital Voice Transmission Pulse Code Modulation



DPI DC

ITU Standard Coding Schemes

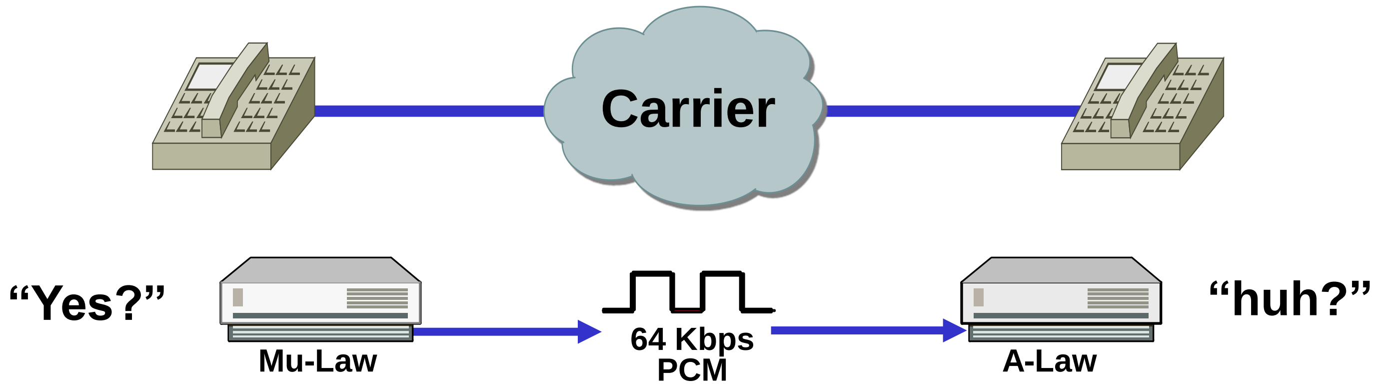
Standard	Bit Rate (Kb/s)	Frame Size (ms)	Year Finalized
G.711 PCM	64	0.125	1972
G.726, G.727 ADPCM	16, 24, 32, 40	0.125	1988/1990
G.722 Wideband coder	48, 56, 64	0.125	1988
G.728 LD-CELP	16	0.625	1992,1994
G.729 CS-ACELP	8	10	1995
G.723.1 MPC-MLQ	5.3, 6.4	30	1995
G.729 CS-ACELP Annex A	8	10	1996

Source: Richard Cox, AT&T Labs, published in IEEE Communications, September 1997

ITU = International Telecommunications Union
CELP = Various types of Linear Predictive Coders
PCM = Pulse Code Modulation

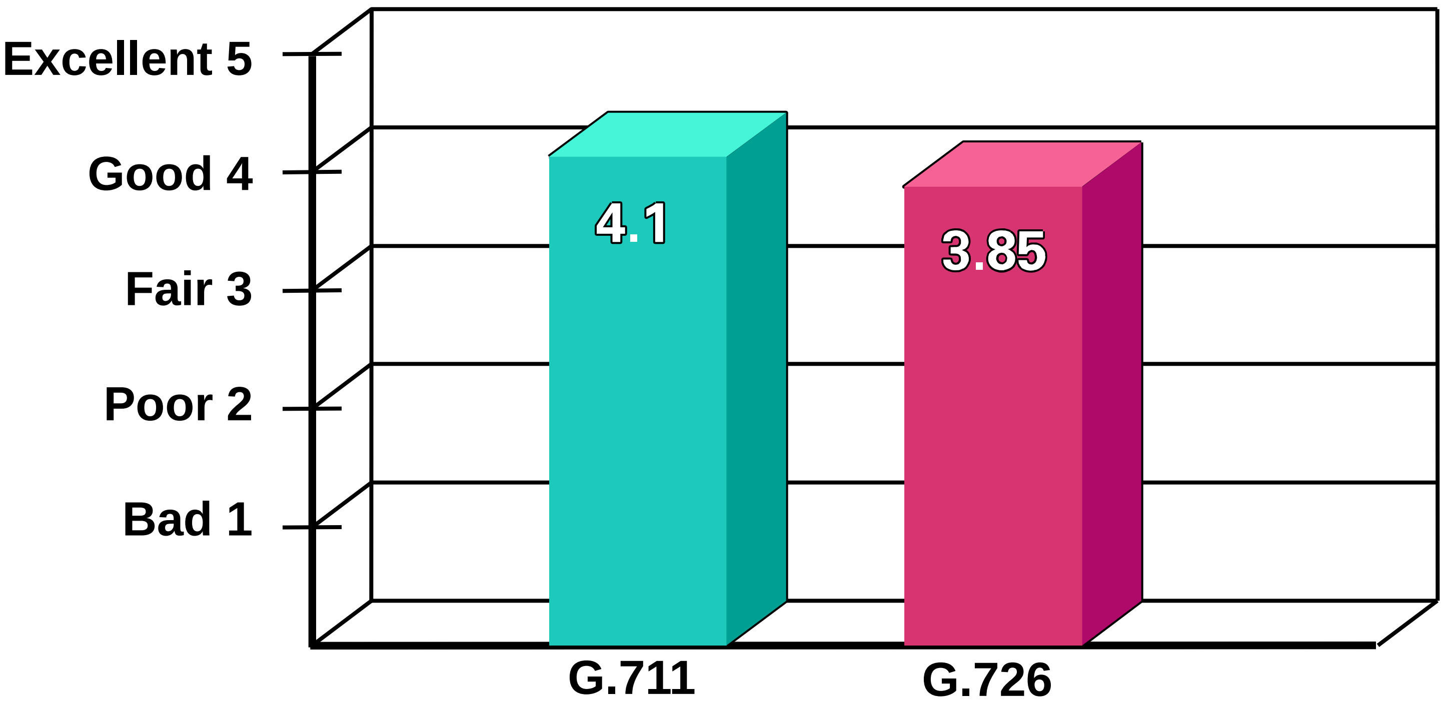
ADPCM = Adaptive Differential Pulse Code Modulation
MPC = Multi-pulse Coding
MLQ = Maximum Likelihood Quantization

Pulse Code Modulation Mu-Law/A-Law



Coded	Mu-Law	A-Law
+127	10000000	11111111
+196	10011111	11100000
+64	10111111	11000000
+32	11011111	10100000
0	11111111	10000000
0	01111111	00000000
-32	01011111	00100000
-64	00111111	01000000
-96	00011111	01100000
-126	00000001	01111110
-127	00000000	01111111

Mean Opinion Score Example



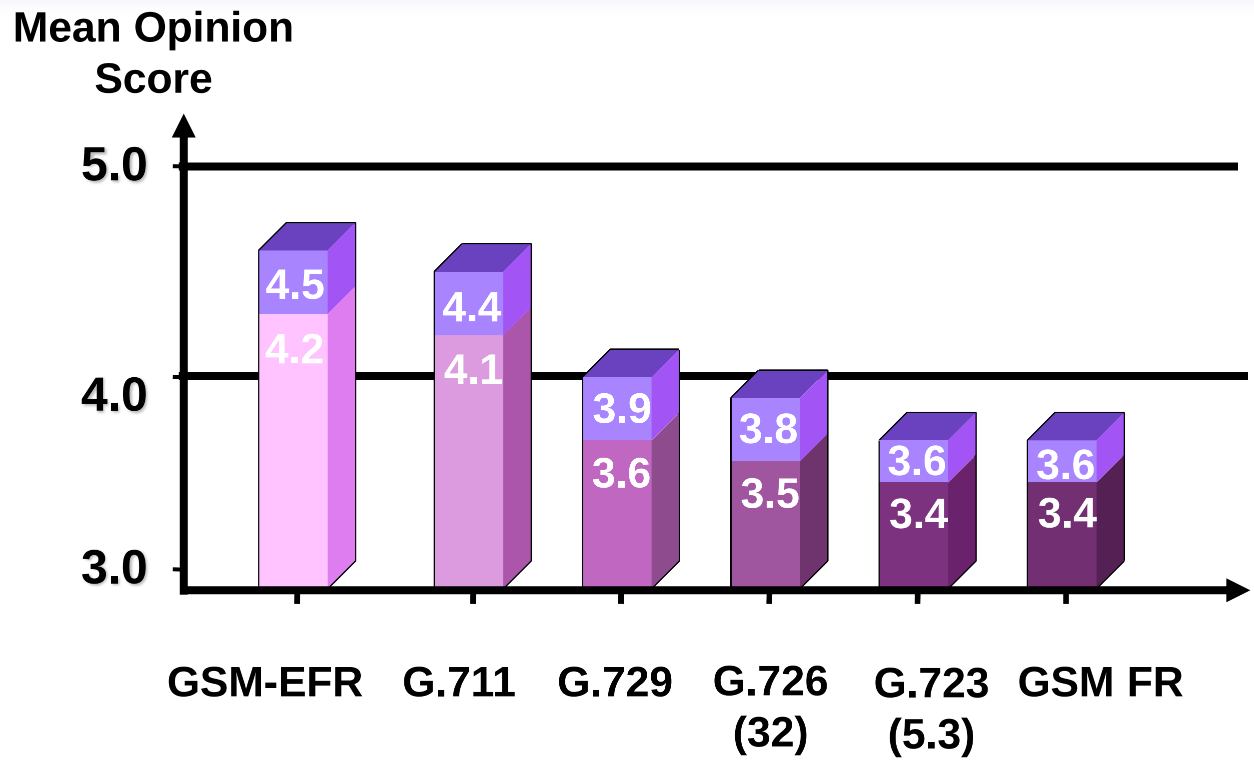
Source: M.E. Perkins/ et al, “Characterizing the Subjective Performance of the ITU-T 8 kb/s Speech Coding Algorithm” - ITU-T G.729, IEEE Communications, Sept. 1997

Comparison of G.723.1, G.729, and G.729 Annex A

	G.723.1	G.729	G.729 Annex A
Bit Rate (kb/s)	5.3 and 6.4	8	8
Total Delay (ms)	67.5	25	25
Processing Required (MIPS)	16	20	10.5
RAM Required (words)	2200	3000	2000

MIPS = Millions of Instructions per Second

One Comparison of Coder Quality



Source: ETSI/TIPHON (99) 15
Testing by Deutshe Telekom Berkom.Listening Tests, in German, with 24 subjects

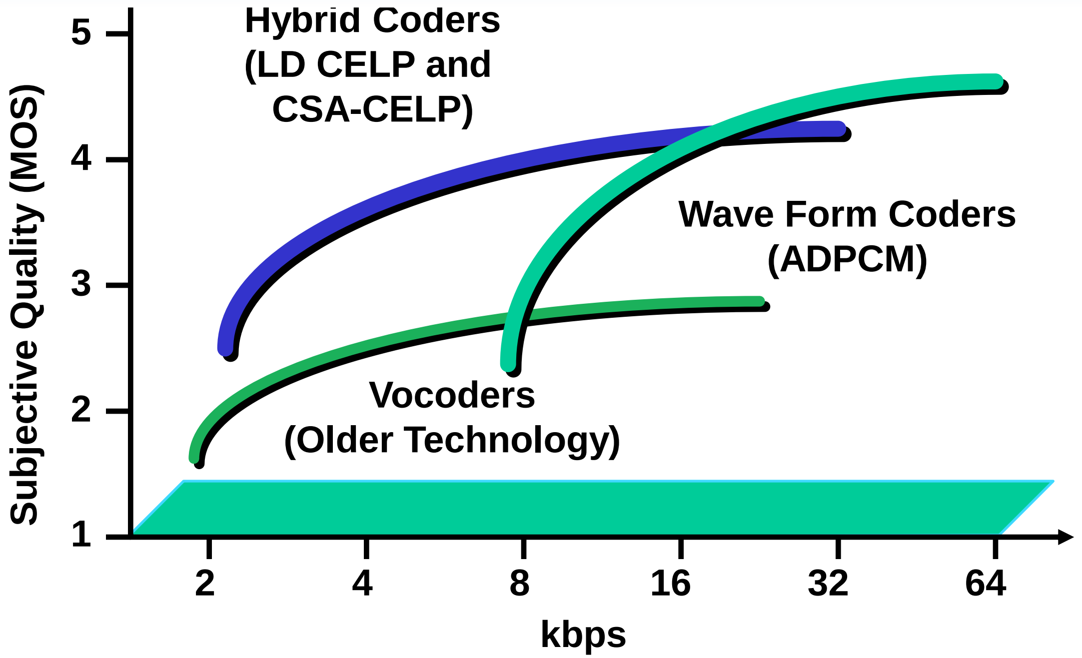
G.711 = A-law PCM "Wireline Quality"
FR = Full Rate

G.726(32) = "toll quality"
EFR = Enhanced Full Rate

Voice Quality Guidelines

Compression Method	MOS Value	Delay (ms)
PCM (G.711)	4.1	0.75
32 K ADPCM (G.726)	3.85	1
16 K LD CELP (G.728)	3.61	3–5
8 K CS-ACELP (G.729)	3.92	10
8 K CS-ACELP (G.729a)	3.9	10

Voice Quality Guidelines



Score	Quality	Description of Impairment
5	Excellent	Imperceptible
4	Good	Just perceptible, not annoying
3	Fair	Perceptible and slightly annoying
2	Poor	Annoying but not objectionable
1	Bad	Very annoying and objectionable

Source: A.M. Kondoz, *Digital Speech Coding for Low Bit-Rate Communications Systems*, 1995

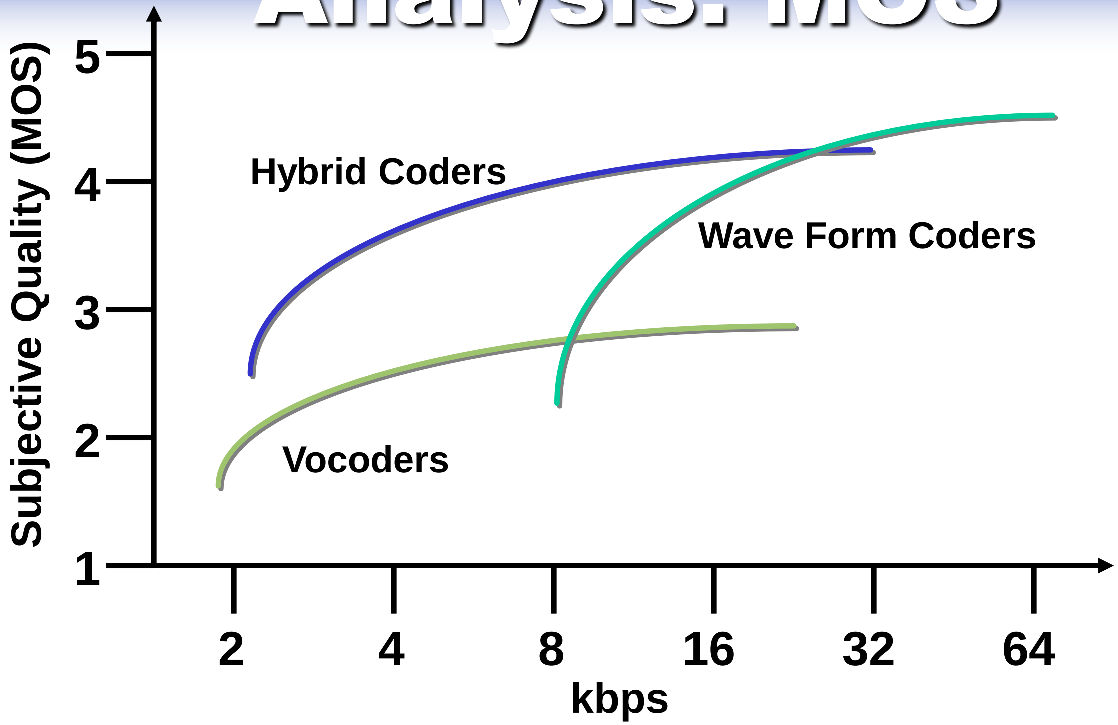
MOS Rating of Digital Voice

Codec		Bit Rate	MIPS	Comp. Delay (ms)	Framing Size	MOS
G.711	PCM	64	0.34	0.75	0.125	4.1
G.726	ADPCM	32	13	1	0.125	3.85
G.728	LD CELP	16	33	3-5	0.625	3.61
G.729	CS-ACELP	8	20	10	10	3.92
G.729a	CS-ACELP	8	10.5	10	10	3.9
G.723.1	MPMLQ	6.3	16	30	30	3.90
G.723.1	ACELP	5.3	16	30	30	3.8

MOS Under Varying Conditions

Example: G.729	MOS Rating
Average speech level	3.92
Low input level	3.54
Two tandem codings	3.46
Three tandem codings	2.68
5% bit error rate	3.24
5% frame error rate	3.02

Subjective Impairment Analysis: MOS



Score	Quality	Description of Impairment
5	Excellent	Imperceptible
4	Good	Just perceptible, not annoying
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1	Bad	Very annoying and objectionable

Product Line Overview—Data/Voice Integration

	Cisco AS5300	Cisco MC3810	Cisco 3600	Cisco 2600
Cisco IOS LAN-to-LAN Routing	Yes	Yes	Yes	Yes
Voice over IP	Yes	Yes	Yes	Yes
Voice over Frame Relay	No	Yes	Yes	Yes
Voice over ATM	No	Yes	Yes	Yes
ATM WAN Access	No	Yes	Yes	Yes
Integrated Dial	Yes	No	Yes	Yes
LAN Media	E/FE	E	E, TR, FE	E, TR, FE
WAN Media	PRI	Serial, BRI, ATM	PRI, MBRI, Serial, ATM	PRI, MBRI, Serial, ATM
Maximum Analog/ Digital Voice Density	0/120	6/24	24/180	4/30

Pricing Structure

- *One-time cost for h/w & s/w*
- *fees charged for gateway service*
  - *pre-charged account*
  - *pay-as-you-use account (digital cash)*
- *Dial-up : $19.95/month unlimited use*
- *Direct connection*
  - *monthly flat-fee based on line rate(64,256,2Mbps)*
  - *line utilization*

market

- *International VoIP calls could cost 1/5th of normal rates*
- *500,000 IP telephony users at the end of 1995*
- *15% of all voice calls on IP/Internet by 2000*
- *10M users & $500M in VoIP product sales in 1999*
- *US VoIP service will grow from $30M in 1998 to $2B in 2004 [Forester research]*
  $2B in 2001 & $16B in 2004 [frost & sullivan]

Summary

- *VoIP products & services are being rolled-out*
- *Ideal for computer based communications*
- *IP needs QoS for acceptable quality*
- *A number of working group at IETF are working on it*
- *H.323 provides interoperability*

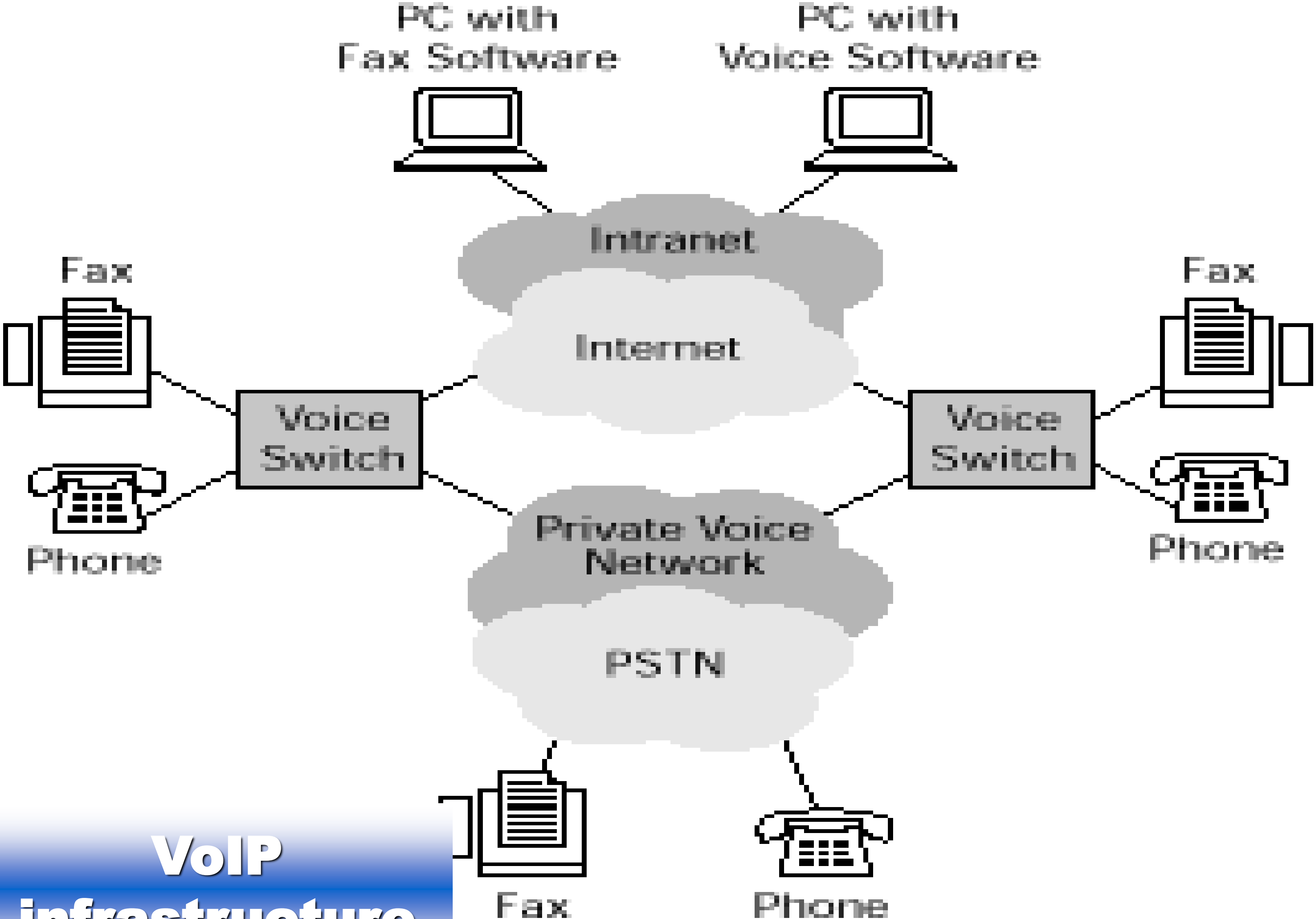
conclusion

- *What dpi can do?*
  - *PC-to-PC, Special Card?*
  - *Pc-to-Phone, which ITSP?*
  - *Intranet VoIP,which brand? (cisco)*
  - *phone-to-phone, TCI?*

Q&A

سؤال

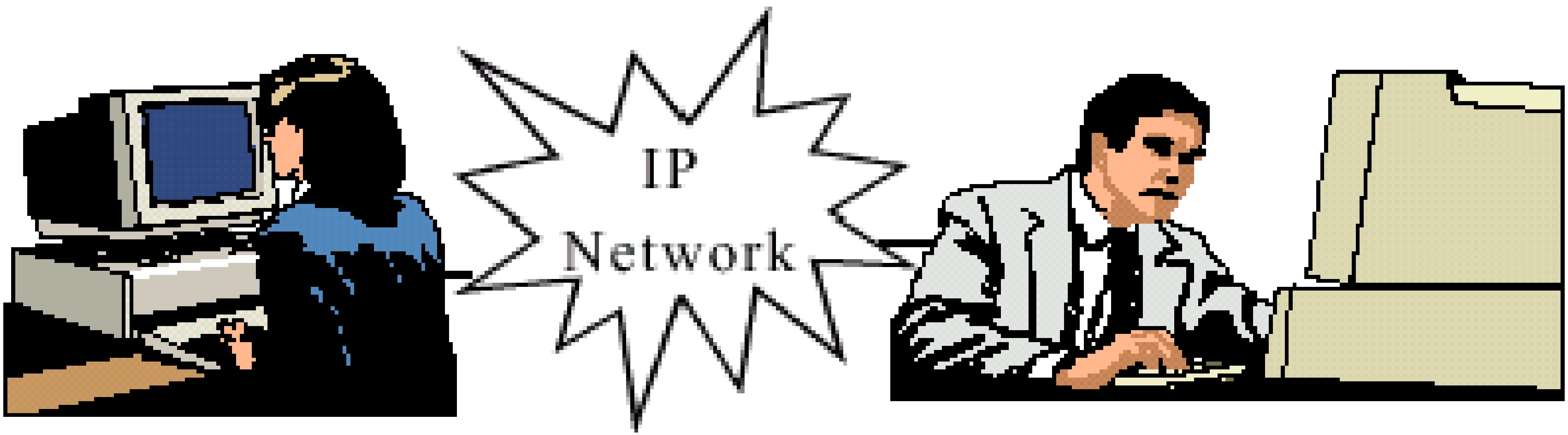
جواب



VoIP infrastructure

Scenario 1: PC to PC

- *Transmission of digitized voice conversations over IP or the Internet by individual PC users.*
  - *Use of sound card, a microphone and appropriate software.*
  - *Using a phone attached to PC through a special card*

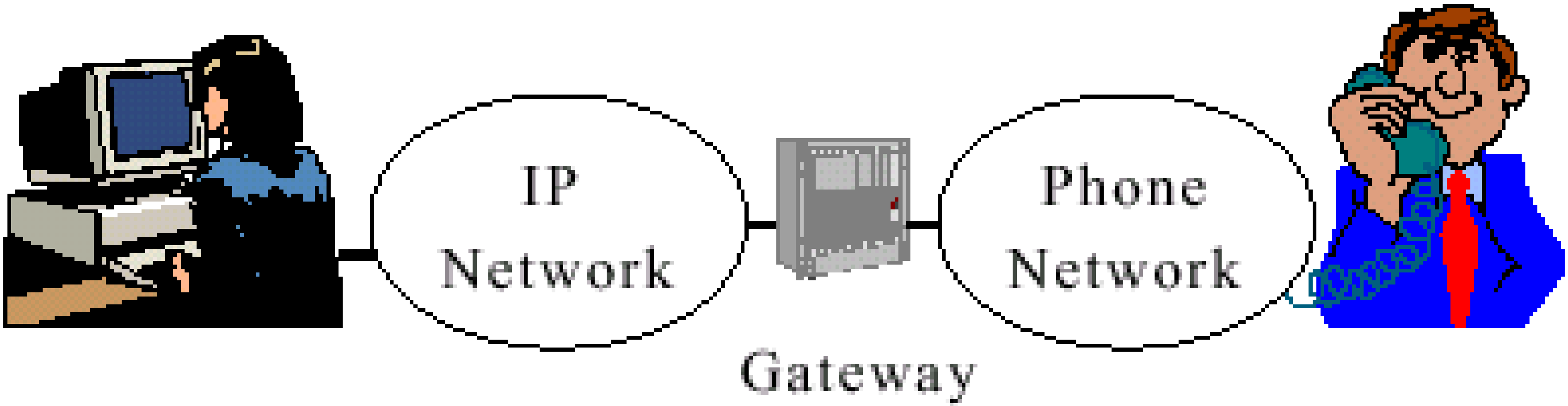


Scenario 1: PC to PC

- *CuSeeMe*
- *LinkTel*
- *VocalTec's Internet phone*
- *Microsoft's NetMeeting*
- *Yahoo Messenger*
- *PalTalk*
- *A lot more…*

Scenario 2: PC to Phone

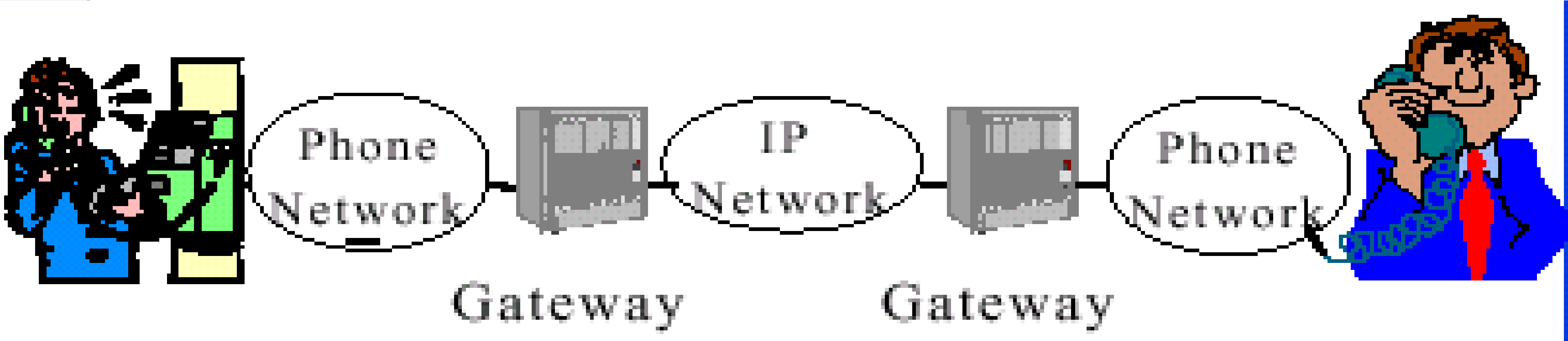
- *Needs a gateway that connect IP network to phone network (Router or PBX)*
- *Users call through the ITSP (Internet Telephony Service Provider)*



Software	Website	Software needed	Rate	PC To PC	PC To Phone
DialPad	www.dialpad.com	No	Call to United States isfree	○	○
DeltaThree	www.deltathree.com	Yes	Very Cheap	×	○
FreeTlakClub	www.zipcom.com.tw	No	Absolutely free globally	○	×
HotTelephone	www.hottelephone.com	No	Call to 20 countries are free	×	○
Iphone	www.iphone.com	Yes	Very Cheap	○	○
MediaRing	www.mediaring.com	Yes	Very Cheap	○	○
Myfreeld	www.myfreeld.com	No	Call to USA and Canada are free	×	○
Net2Phone	www.net2phone.com	Yes	Very Cheap	○	○

Scenario 3: Phone to Phone

- *Needs more gateways that connect IP network to phone network*
- *The IP network could be a dedicated intranet or the Internet*
- *The phone networks could be intra-company PBXs or the carrier switches*



Sample Service (ITSP=GSP)

- *IDT corporation : Net2Phone, Carrier2Phone, Phone2Phone services*
- *Global Exchange Carrier offers international calls using VocalTec InternetPhone s/w & gateways*
- *ITXC, GRIC, Qwest, AOL, AT&T, USA Global link, Delta3, MCI, sprint, KDD/japan, Dacom/korea, Deutche telecom, France telecom, telecom Filand, New Zeland telecom, Bell atlantic, AT&T/japan,World Com, iBasis*

Providers & ITSPs

- *Arbinet, AT&T, CollCall, DeltaThree, DotCom, GRIC, GTE, iBasis, Inter-Tel, IPVoice, ITXC, …, Vocaltec*
- *DialPad, DigitCom, DSG, Excite, Gecco, Go2Call, Golden Line, iCall,*