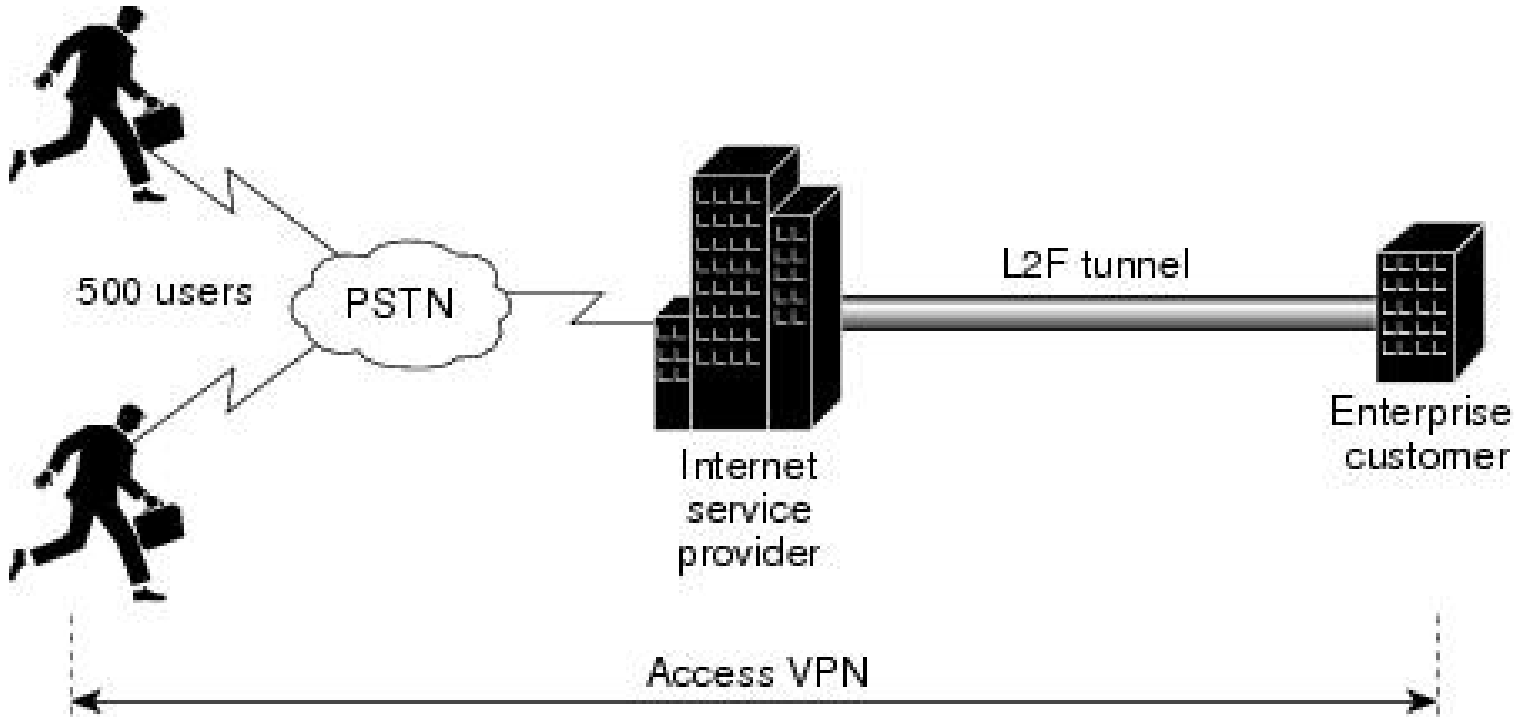
Layer 2 Forwarding (L2F)

- Разработен от Cisco - RFC 2341
- Предвиден за PPP (Dial-up) пакети, може да се използва и за други



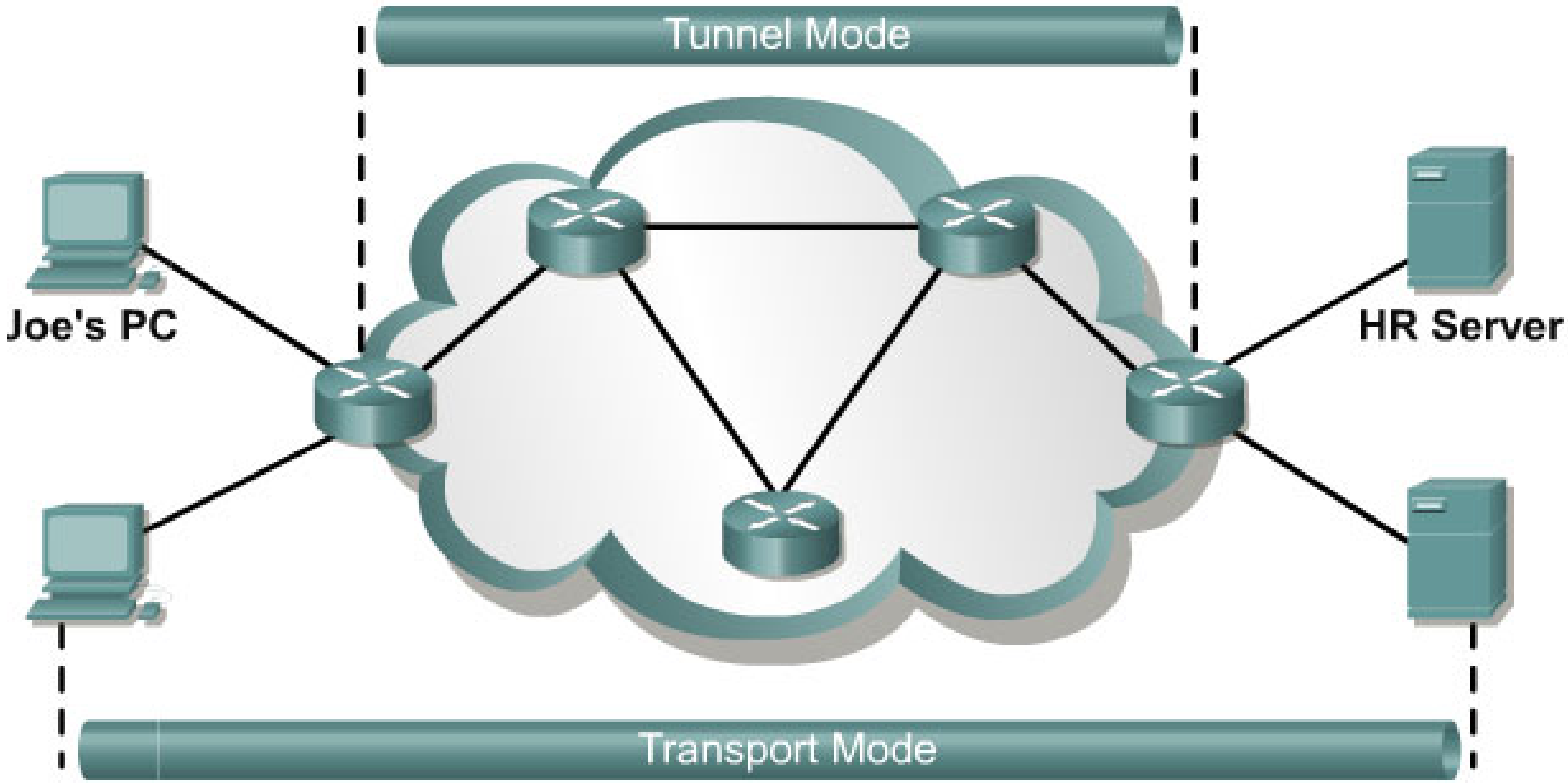
Layer 2 Tunneling Protocol - L2TP

- L2F = Layer 2 Forwarding (CISCO)
- L2TP = L2F + PPTP
- Събира добрите страни на L2F и PPTP
- Позволява изпращане на PPP кадри през не-IP (Frame relay, ATM) мрежи (PPTP работи само върху IP)
- Позволява много (с различен QoS) тунели между едни и същи точки.
- Поддържа flow control

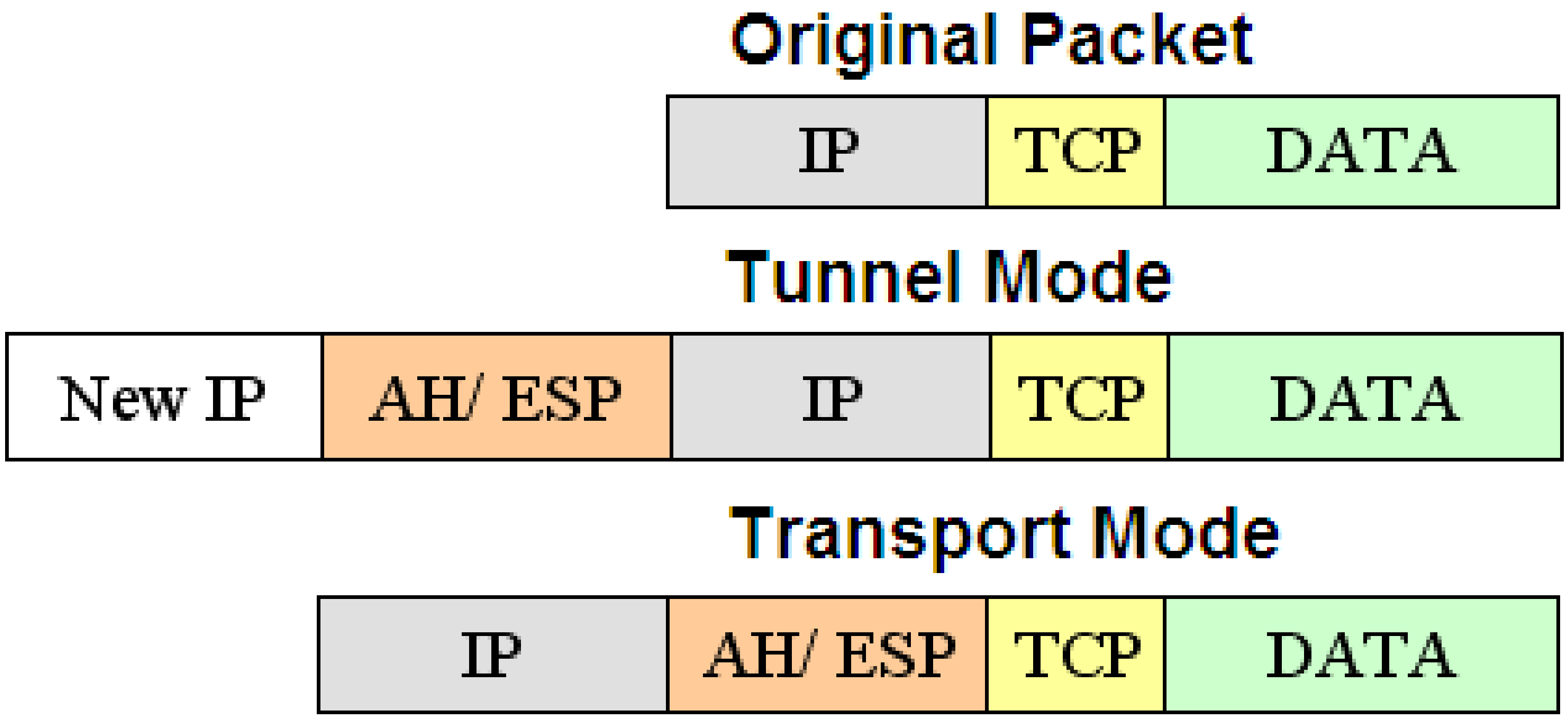
IPSec (Secure IP)

- Серия препоръки на IETF
- Разделя автентичността и поверителността
- Authentication Header (AH) се грижи за цялостта и автентичността на данните
- Encapsulating Security Protocol (ESP) осигурява цялост и поверителност
- Могат да се използват самостоятелно или заедно

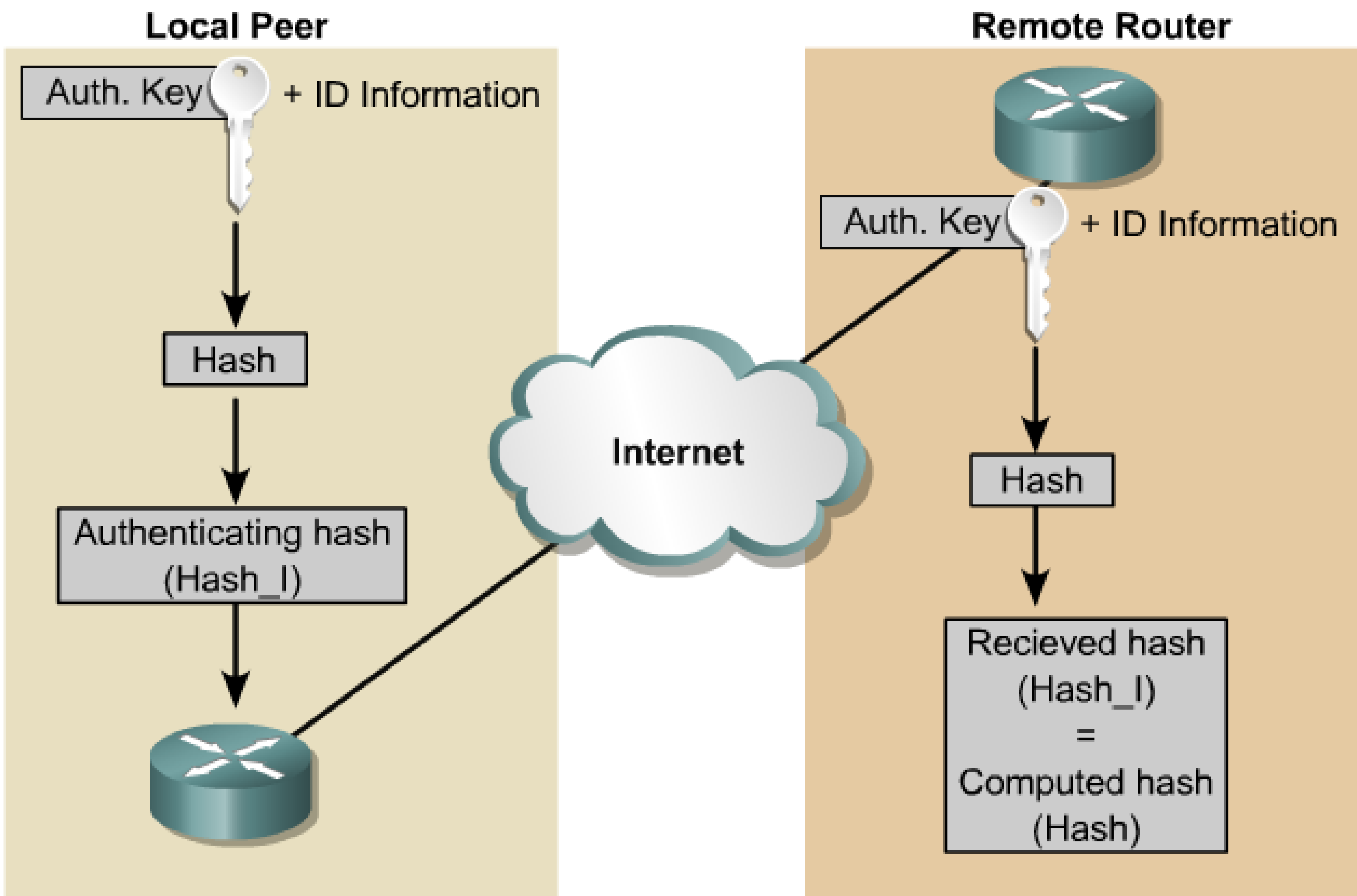
Режими на работа



IPSec режими на работа



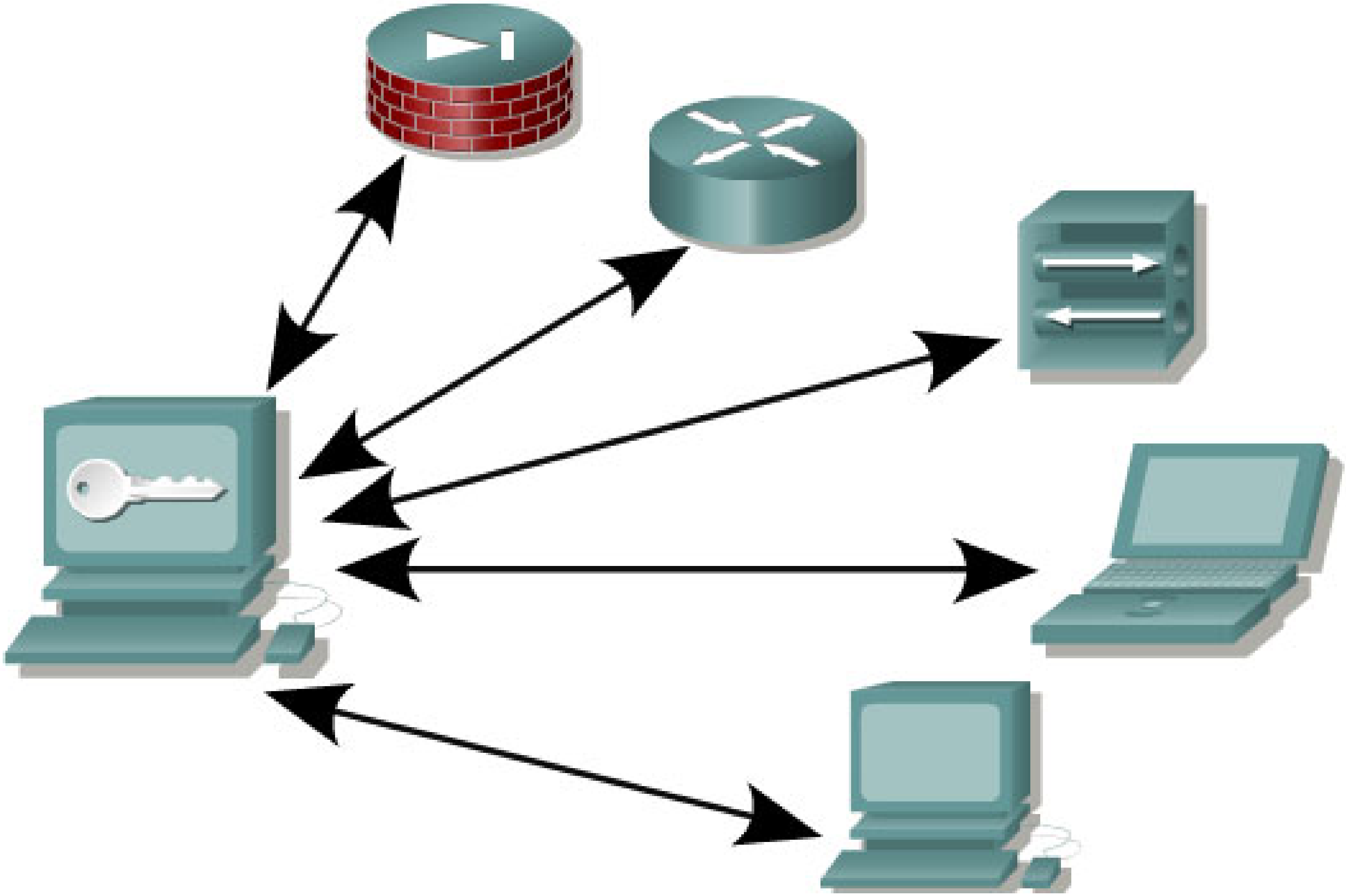
Споделени ключове



Цифрови сертификати

- **Internet Key Exchange (IKE)**
- **Public-Key Cryptography Standard #7 (PKCS #7) – RSA Data Security**
- **Public-Key Cryptography Standard #10 (PKCS#10) – RSA Data Security**
- **RSA – Rivest, Shamir, Adleman**
- **X.509v3 certificates**

Certificate Authority сървър

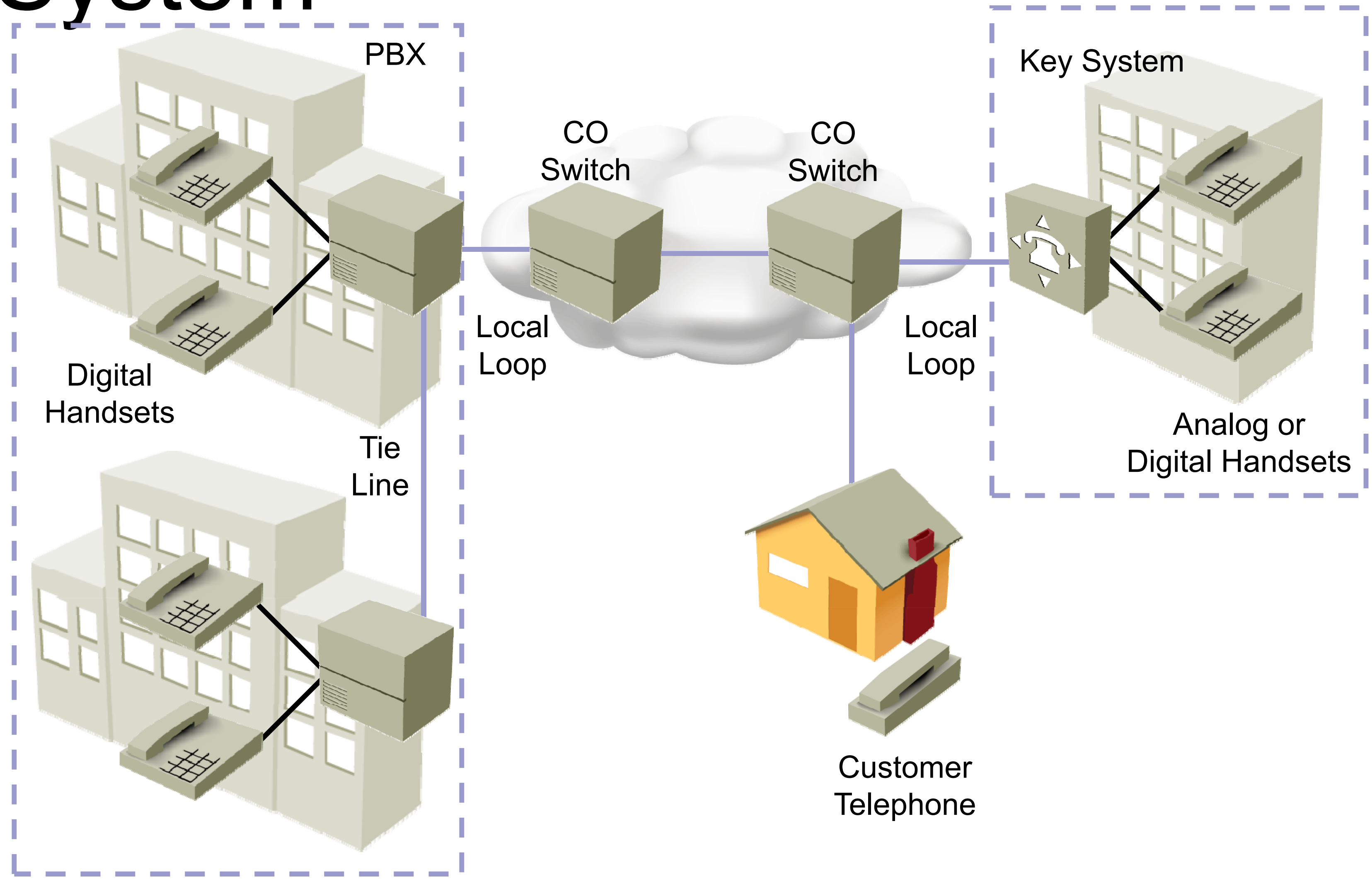


https://youtu.be/bnV-_BN9OkE - PPTP/L2TP/SSTP/OpenVPN
https://youtu.be/doSW8d2iLFM - Общ преглед на IPSec
https://youtu.be/tuDVWQOG0C0 - Какво е IPSec
https://youtu.be/rwu8__GG_rw – Authentication Header (AH) vs Encapsulating Security Payload (ESP)
https://youtu.be/qXLD2UHq2vk - Въведение в цифровите сертификати

Тема : IP Телефония

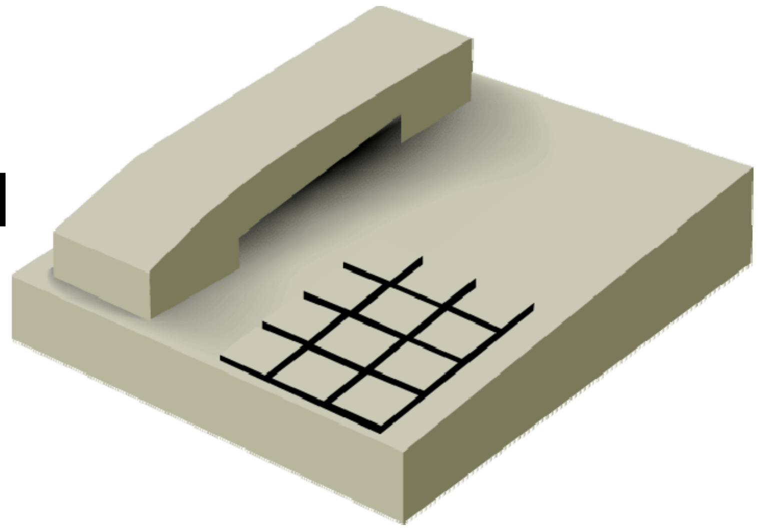
(IP Telephony)

Traditional Business Phone System

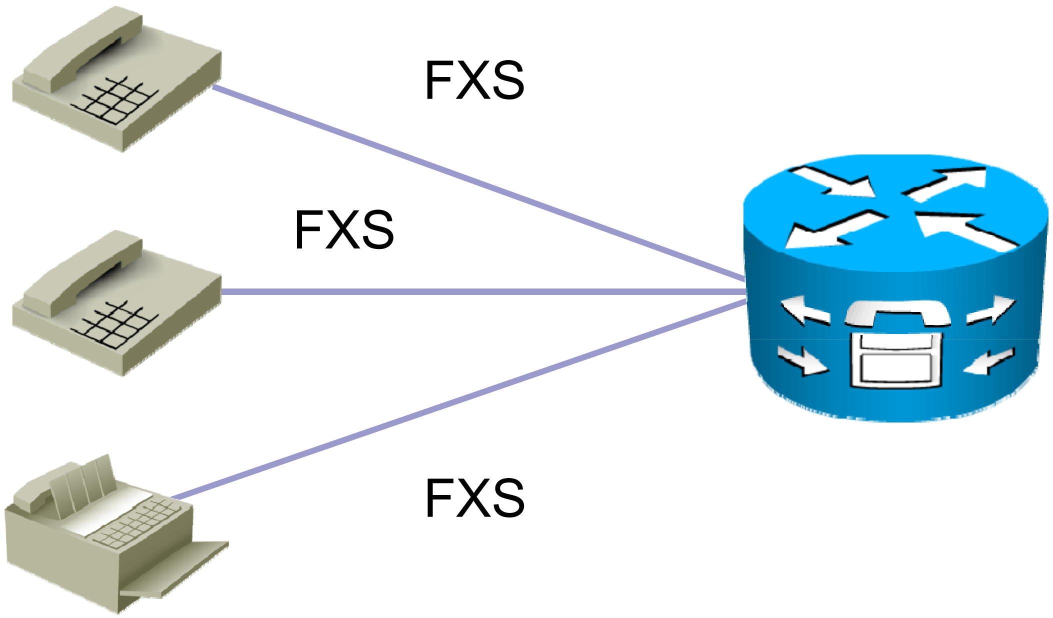


Components of an Analog Telephone

- Receiver
- Transmitter
- Two-wire/four-wire hybrid
- Dialer (DTMF or pulse)
- Switch hook
- Ringer

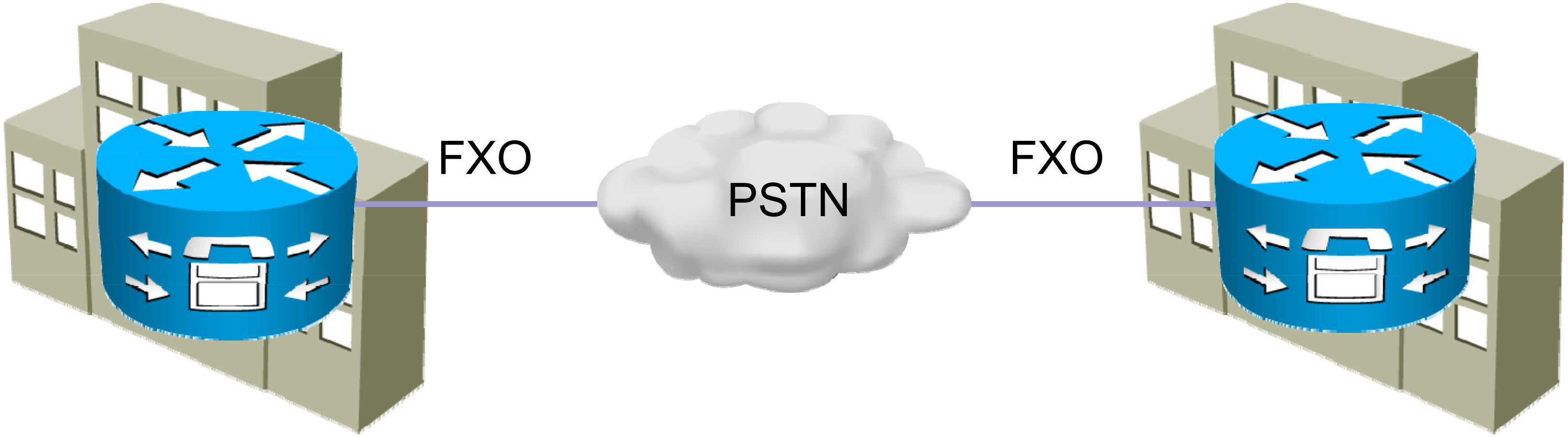


FXS Interface



- Connects directly to analog phones or faxes
- Provisions local service
- Emulates the CO to the attached devices
- Provides power, call progress tones, and dial tone

FXO Interface

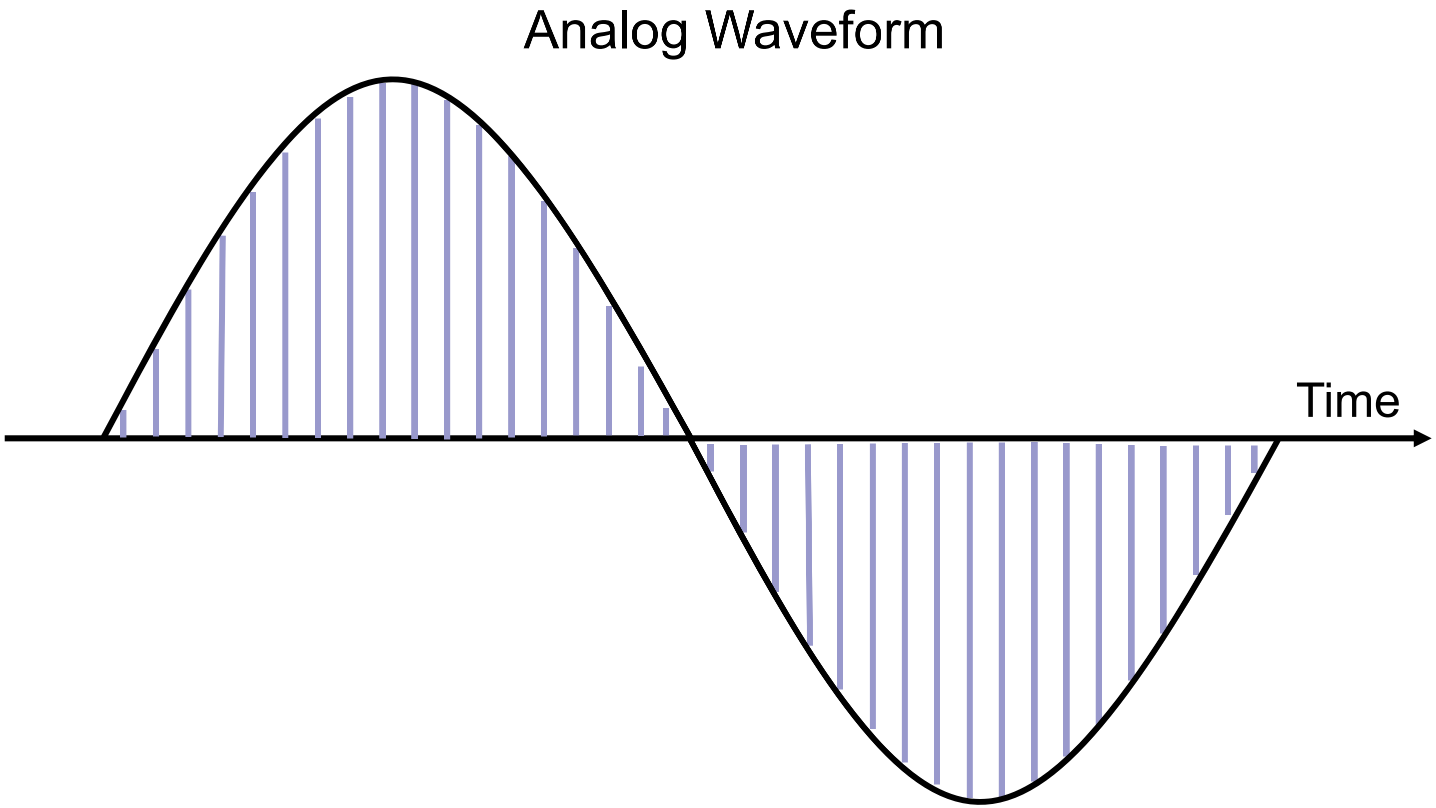


- Connects directly to office equipment
- Used to make and receive calls from the PSTN
- Can be used to connect through the PSTN to another site
- Answers inbound calls

Digitizing Analog Signals

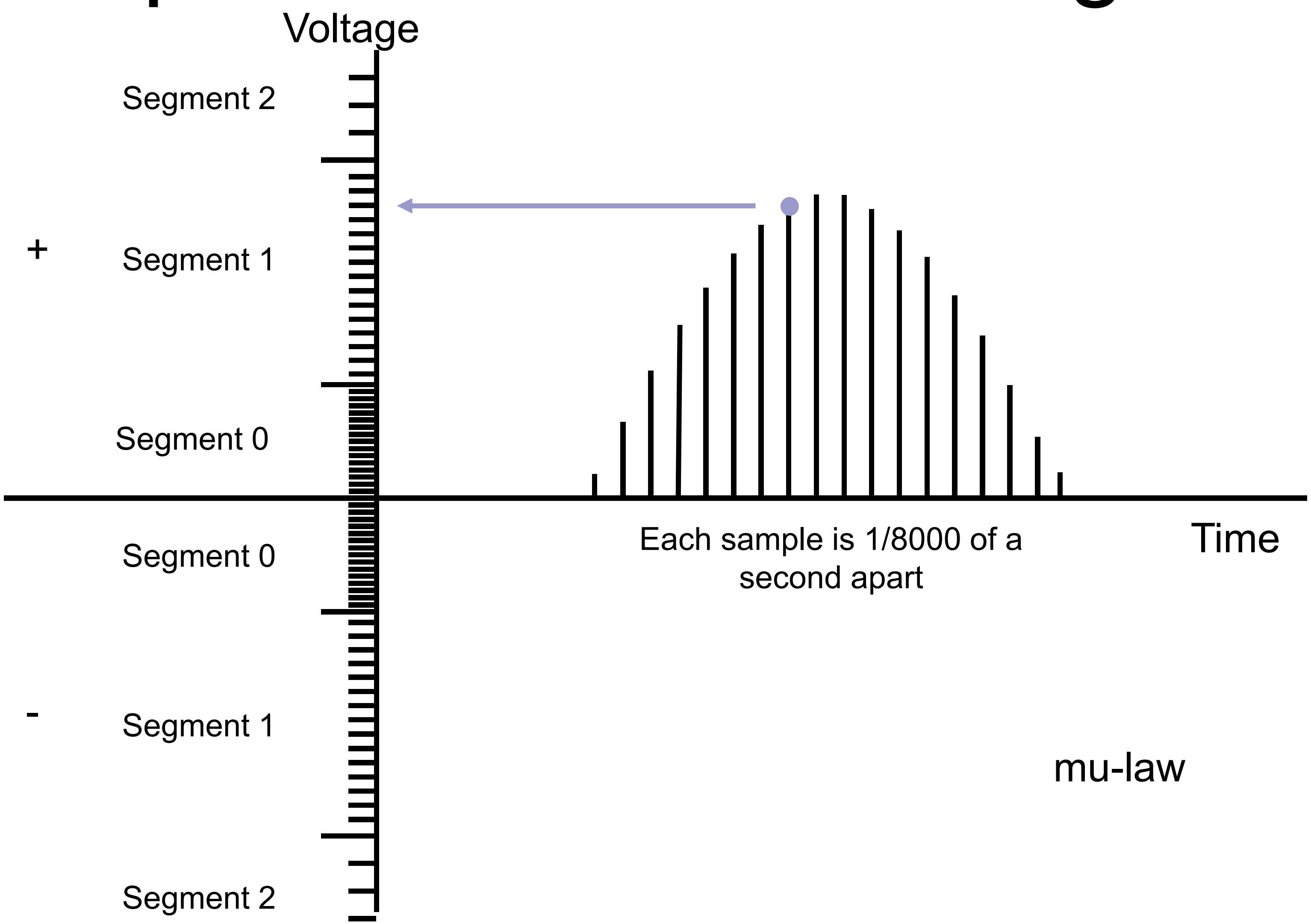
1. Sample the analog signal regularly.
2. Quantize the sample.
3. Encode the value into a binary expression.
4. Compress the samples to reduce bandwidth (optional).

Step 1—Sample the Signal

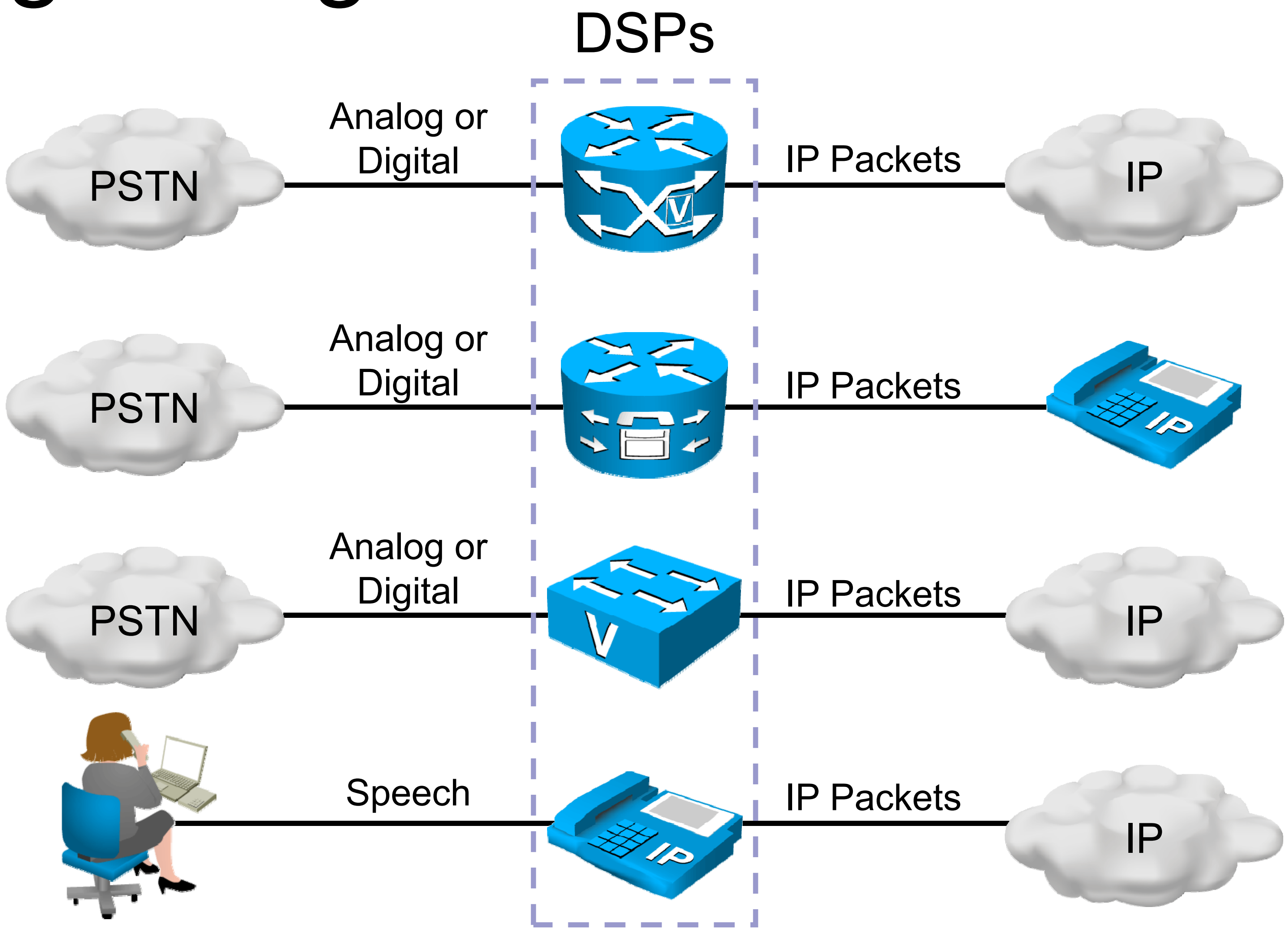


Each sample is 1/8000 of a second apart.

Step 2—Quantize the Signal



Digital Signal Processors



Digital Signal Processors (Cont.)

- The DSP chip performs the sampling, quantization, encoding, and optional compression step of digitization.
- It is used in both directions to convert from a traditional analog or digital voice signal to VoIP; or from VoIP to a traditional analog or digital voice signal.
- The number of simultaneous calls a chip can handle depends on the type of DSP and the codec being used.

https://youtu.be/2x3Ie6VZ_sg – Въведение във VoIP