

Basic Telephony

With a view to VoFR and VoIP



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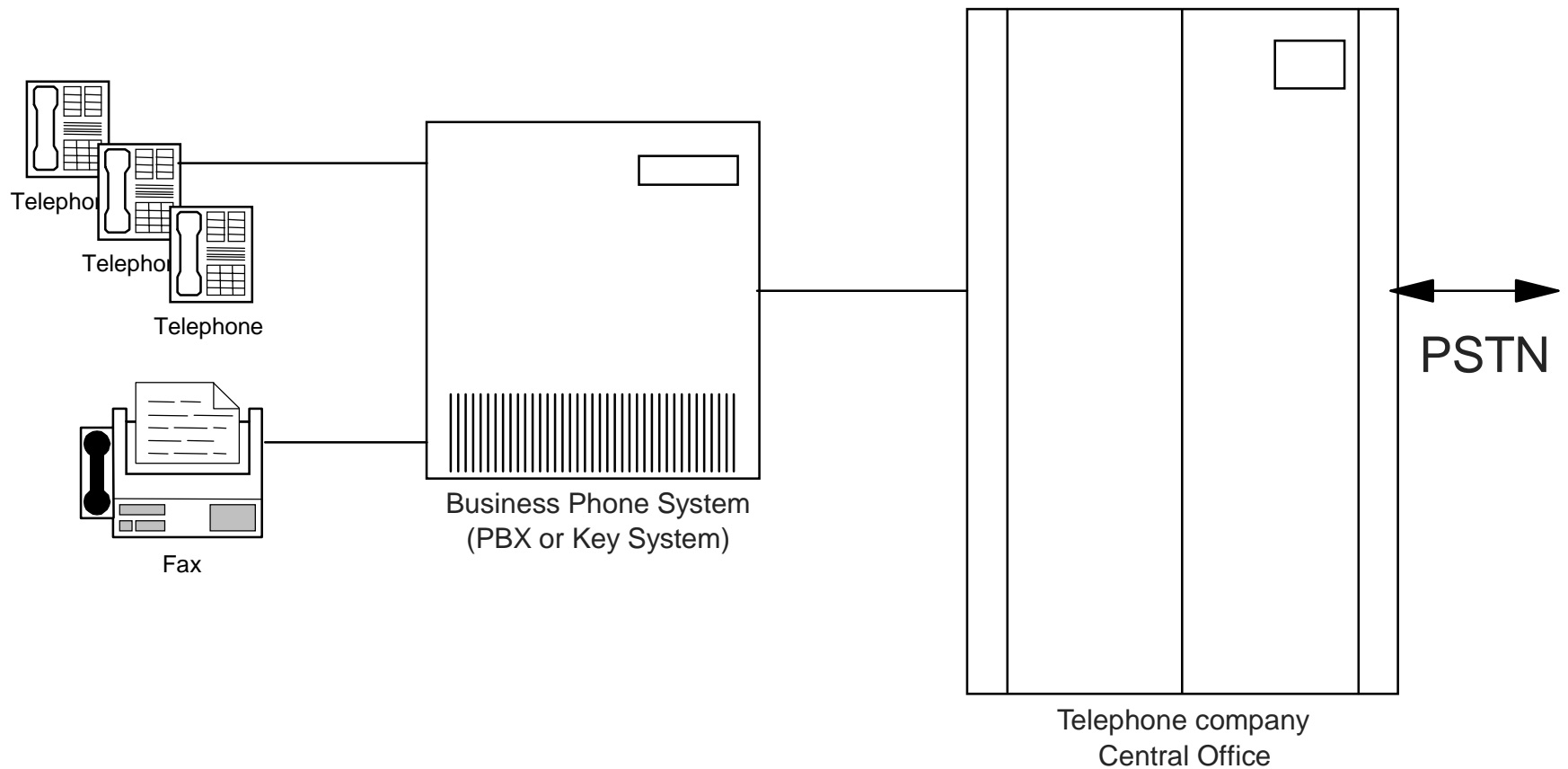
Agenda

- Business telephone networks
- Signaling
- Foreign exchange
 - FXO
 - FXS
- Signaling examples
 - Loop Start
 - E & M
- Network signaling
- Packet switch connections
- Voice Transport
 - Analog voice
 - Digital voice
 - Compression
 - Silence suppression
 - Delay
 - Jitter
 - Echo
 - Fax



Telephone Networks

The Business environment



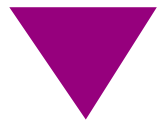


Telephone Networks

Some definitions

Public Switched Telephone Network (PSTN) - The global, publicly accessible telephone network. What we used to call "the telephone company".

Central Office (CO) - a publicly accessible telephone switch owned by a telephone company. Typically large, 10k - 100k lines. A CO serves subscribers. CO are interconnected by trunks and other switches to form the PSTN. Virtually all are stored program controlled (SPC) with digital switch fabrics.



Telephone Networks

Some definitions

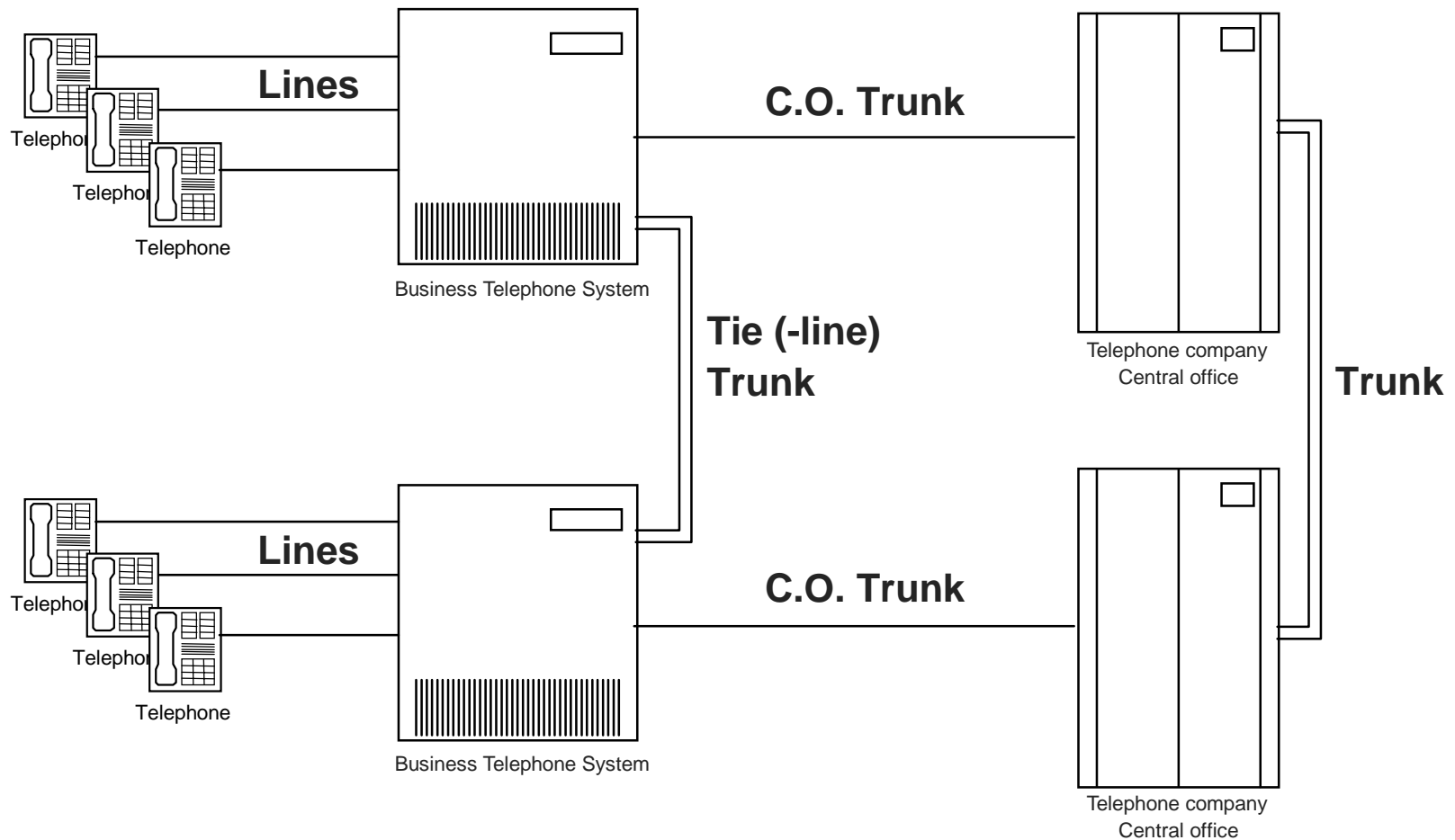
Key Telephone System - a manually controlled small business telephone system in which selection of outgoing lines is by pressing buttons (keys) on the individual telephones. Most modern KS are actually small PBXs. The prototypical KS was the 1A2 by Western Electric. Very small, typically 2 to 10 telephones.

Private Branch Exchange (PBX) - an automatic business telephone system which permits calls between extensions without accessing the PSTN. Selection of outgoing lines is by dialing codes. Usually provides convenience features not available on public telephone switches. Typically 4 to 5000 lines.

PBX were made in every technology; fortunately, virtually all non-digital PBX are gone.



Telephone Networks





Telephone Networks

A **line** is a single circuit that connects a telephone to a switch

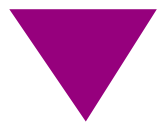
- Analog line connects a station set, modem, fax
- Digital line connects high-function proprietary telephones.
- OPX lines connect distant stations as though they were local.
- KTS lines are usually called *extensions*.

A **trunk** is one or more circuits that interconnect switches.

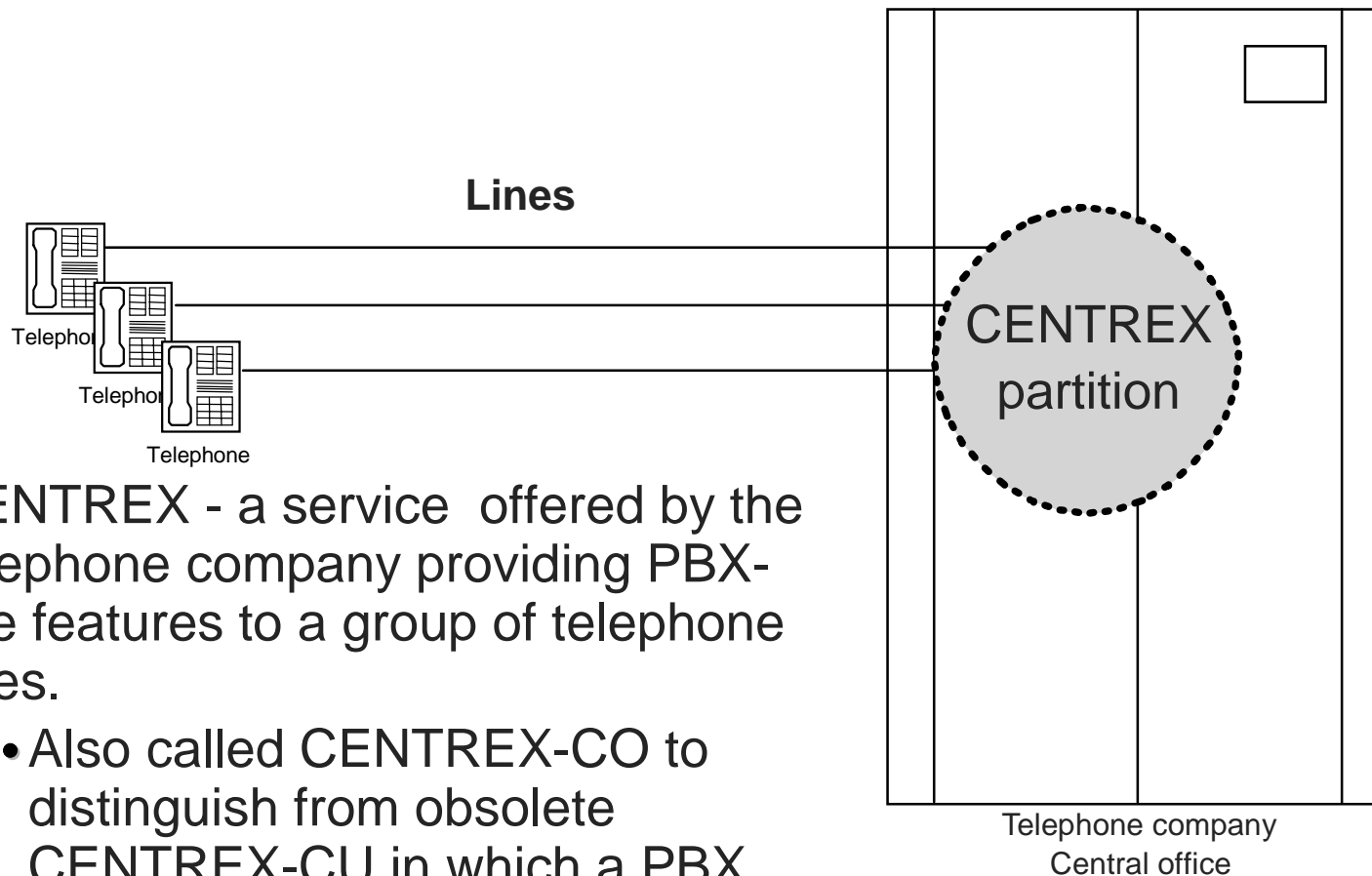
- Central Office trunks connect the PBX to the local public network switch.
 - They appear as lines on the CO switch but trunks on the PBX.
 - On KTS trunks are called lines
- Direct Inward dialing trunks are CO trunks which allow direct dialing to a PBX line.
- Tie trunks connect two PBXs. They are often called Tie Lines.

A **tandem trunk** is a trunk which is switched in an intermediate PBX on its way to the terminating PBX.

A **trunk group** is a collection of trunks with similar destination and use.

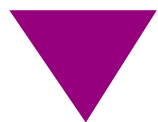


Telephone Networks

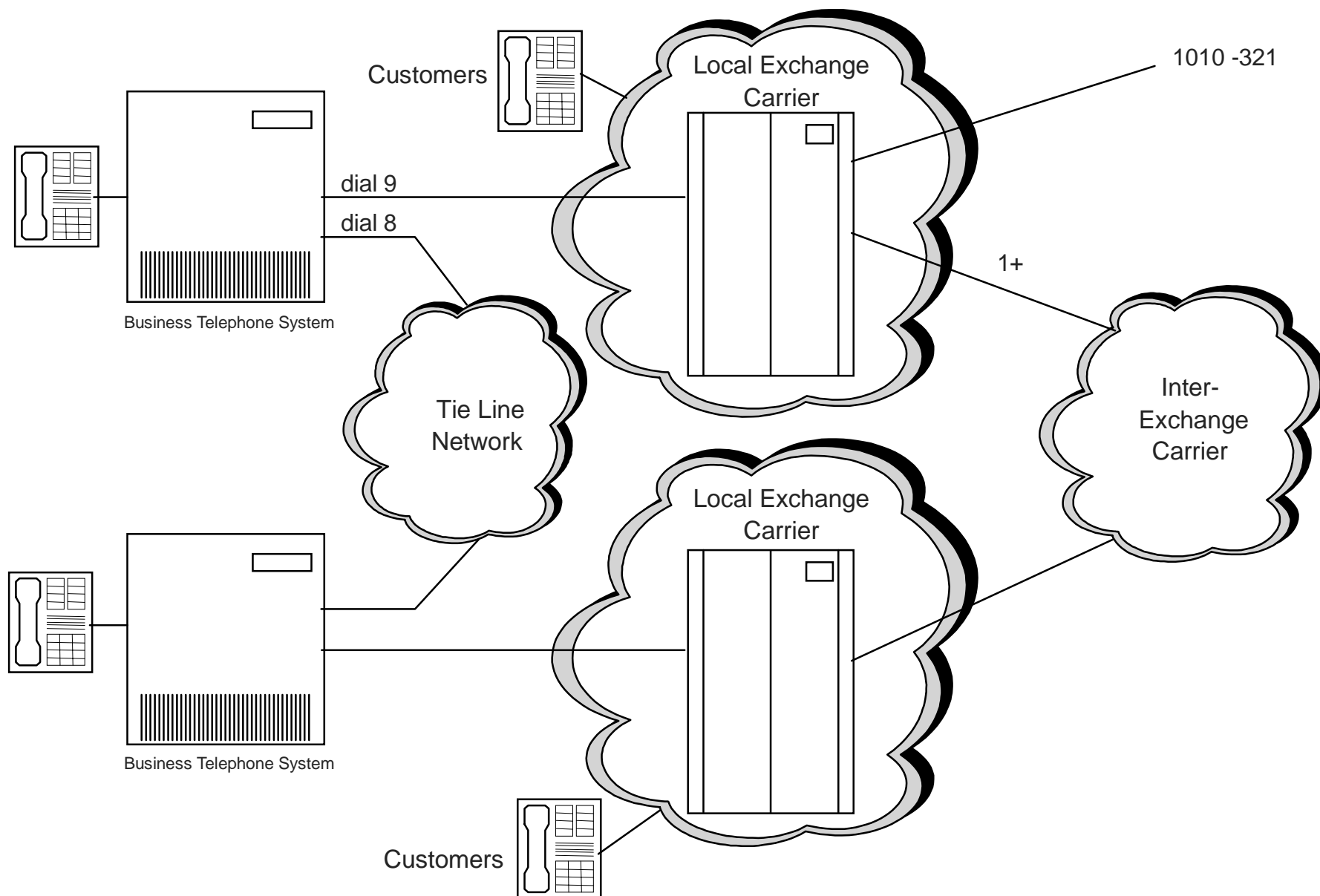


CENTREX - a service offered by the telephone company providing PBX-like features to a group of telephone lines.

- Also called CENTREX-CO to distinguish from obsolete CENTREX-CU in which a PBX was leased to the customer.

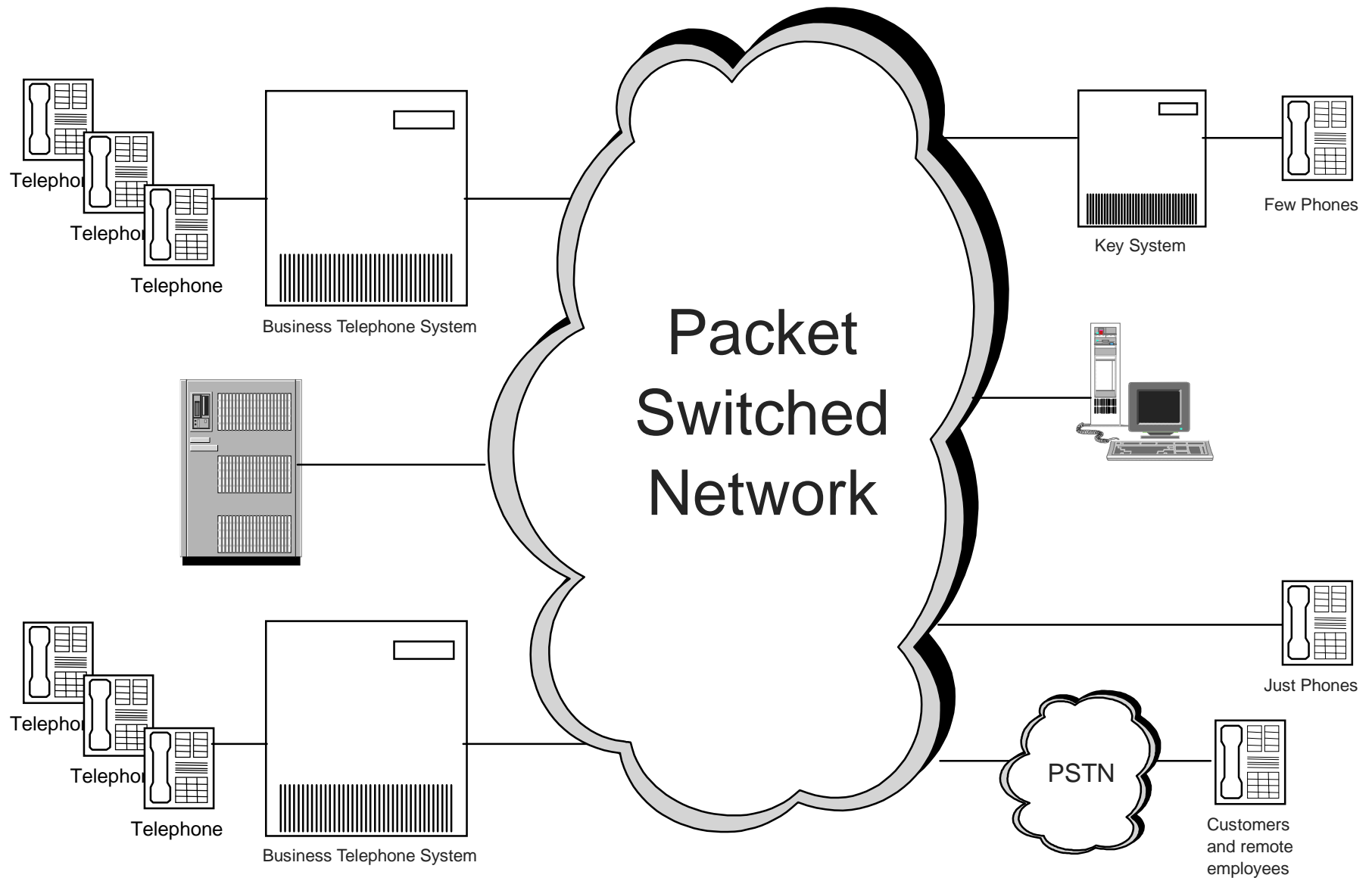


Telephone Networks





Telephone Networks



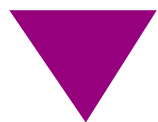


Numbering Plans

Numbering plans insure that every telephone and service in a network has a unique destination address.

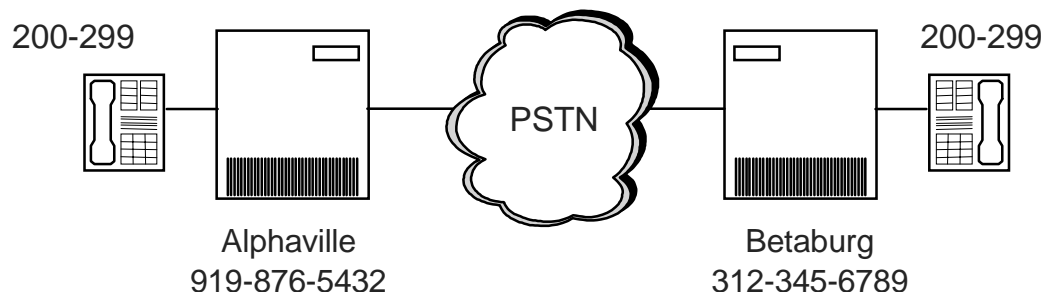
Every network needs a numbering plan

- Global numbering plan defined by E.164
 - Region Codes, Country Codes, City Codes, Lines
- Private networks usually use unique plans - PNP
 - IBM tieline network is a good example of a complex PNP

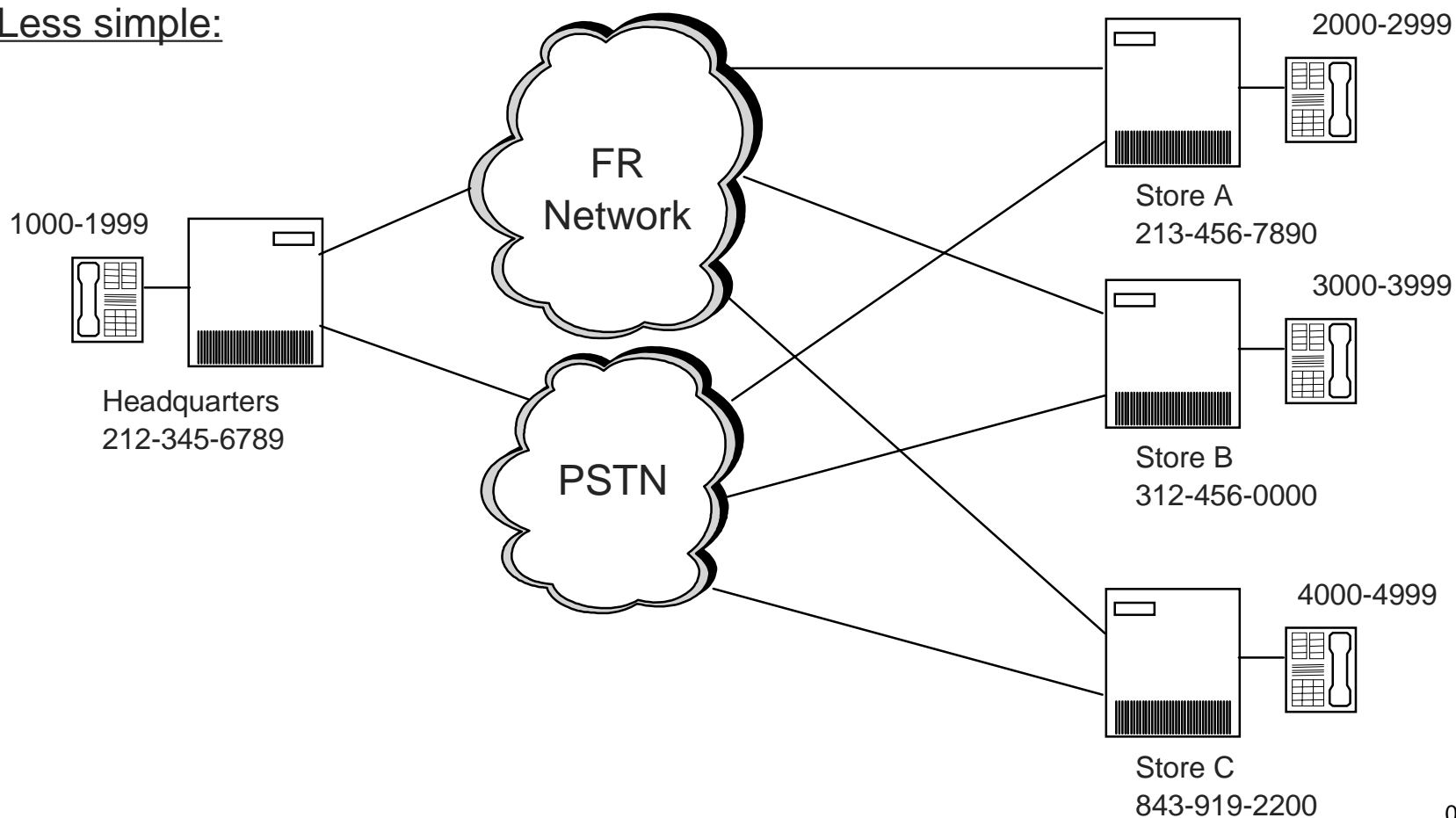


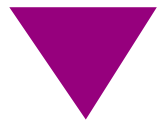
Numbering Plans

Simple:



Less simple:





Translation

Anything that switches telephone calls based on a numerical input must do translation.

- A PBX translates a number into a physical line or a route.
- A 2212 translates a number into a DLCI and subchannel.

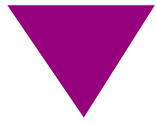
Translation is accomplished through tables in the switches.

- Results in the selection of a route for outgoing calls
- Results in the selection of a physical line for incoming calls

Translation tables also define:

- What address digits to send out
 - may be fewer *or more* than what was dialed
- What to do if route is unusable

Who will write the translation tables is an important part of installation planning



Signaling overview

Signalling is the exchange of control information:

- Between subscribers and switches
- Between switches

Signalling includes Supervision, Addressing, and Alerting

Supervision describes the condition of a line or trunk:

- Minimum of two states:
 - On-hook - idle
 - Off-hook - active, also called seized

Addressing transfers the called number to and through the network.

Alerting lets a subscriber know that action is required.

- Usually it means "pick up the 'phone"



Line Signaling

Common Line Signaling methods:

Supervision:

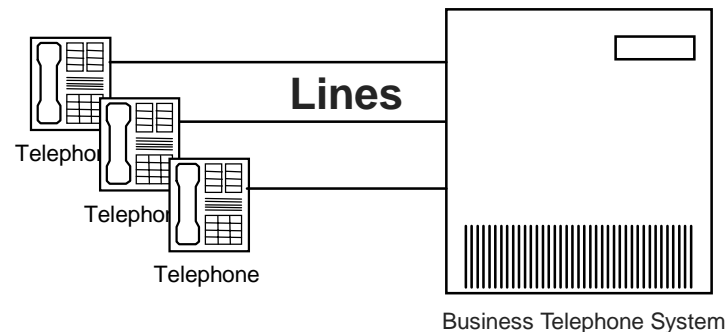
- Loop
- Proprietary digital

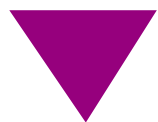
Addressing:

- Dial Pulse - rare
- DTMF tones
- Proprietary digital

Alerting:

- AC ringing - 105v 20Hz common
- Proprietary digital





CO Trunk Signaling

Common CO Trunk Signaling methods:

Supervision:

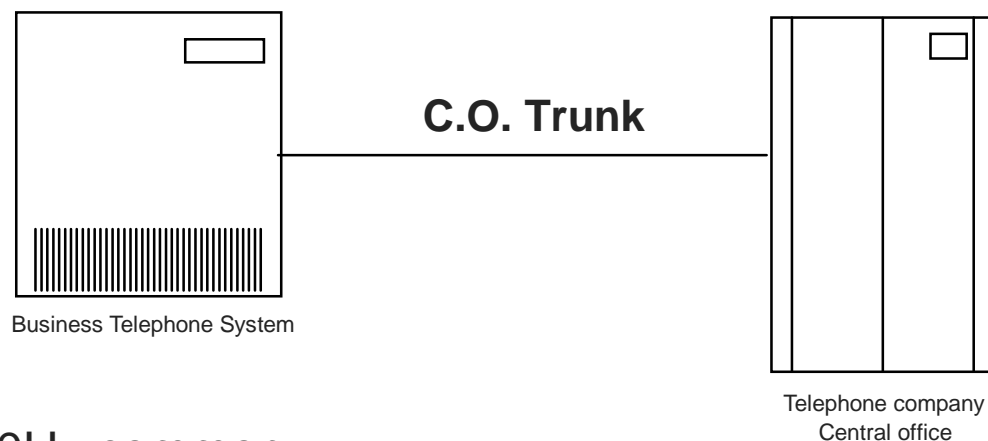
- Loop
- Ground Start

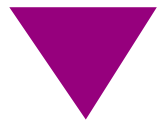
Addressing:

- Dial Pulse - rare
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Alerting:

- AC ringing - 105v 20Hz common





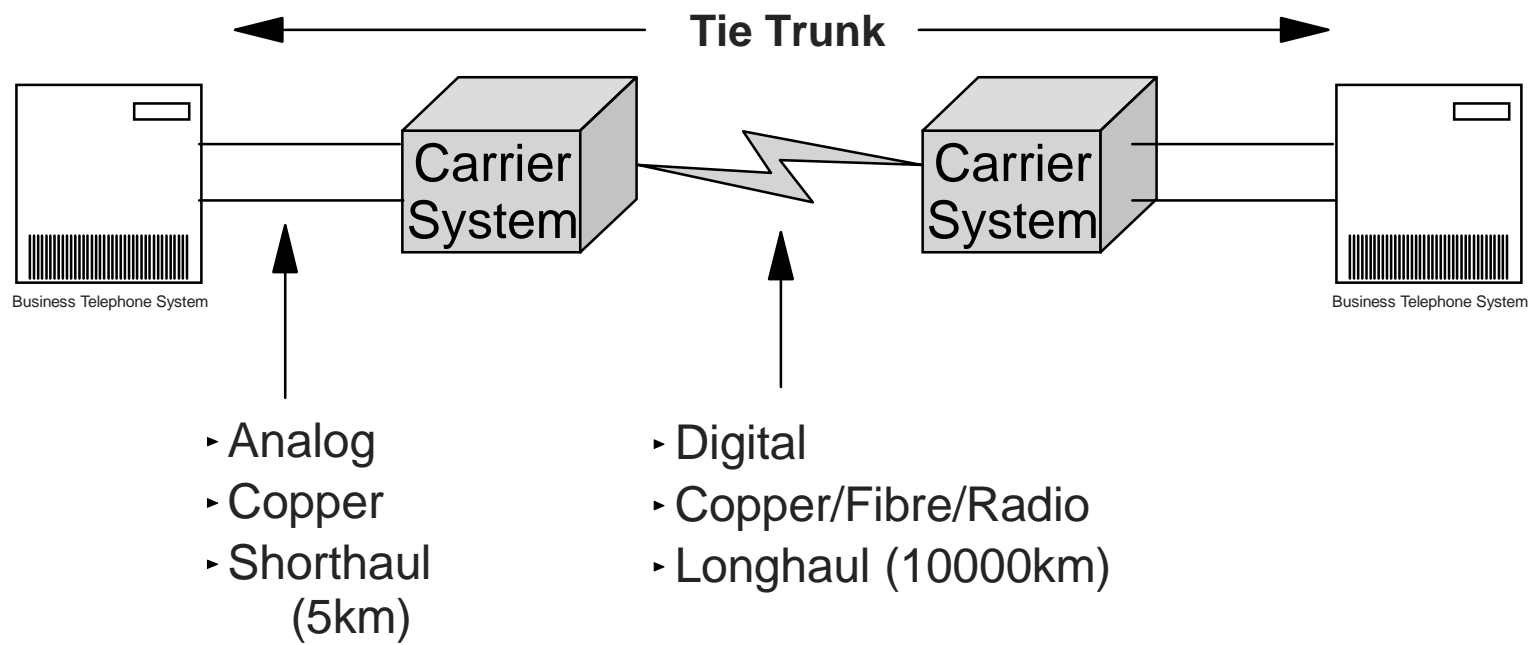
Trunk Signaling

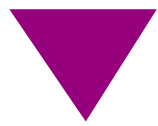
Signaling between switches

Analog Methods

E & M Signalling

- Short Distance method
 - Analog signals are distance limited
- Requires a Carrier System
 - T1 and E1 most common





Trunk Signaling

Signaling between switches

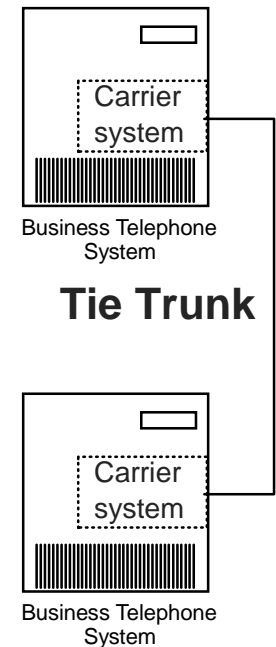
Digital Methods (T1 and E1)

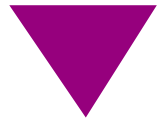
Channel Associated Signaling (CAS):

- The signaling is carried in the voice channel, or in a place permanently associated with the voice channel.
- CAS Supervision:
 - E&M
- CAS Addressing:
 - DTMF tones

Common Channel Signaling (CCS):

- The signaling is in a separate channel from the voice, and may be in a separate physical facility.
- CCS implies that the signaling is in the form of messages.
- Proprietary CCS:
 - Cornet, DMI-BOS, various "ISDN"s
- Standard-based CCS:
 - SS#7, Q.Sig, DPNSS



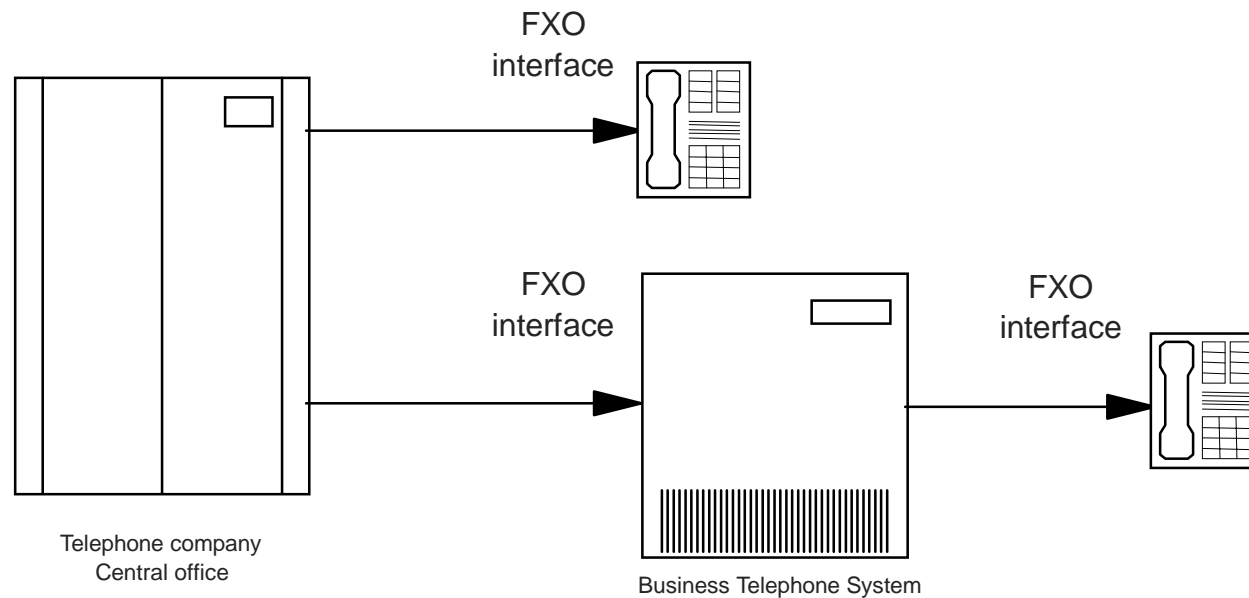


FXO and FXS

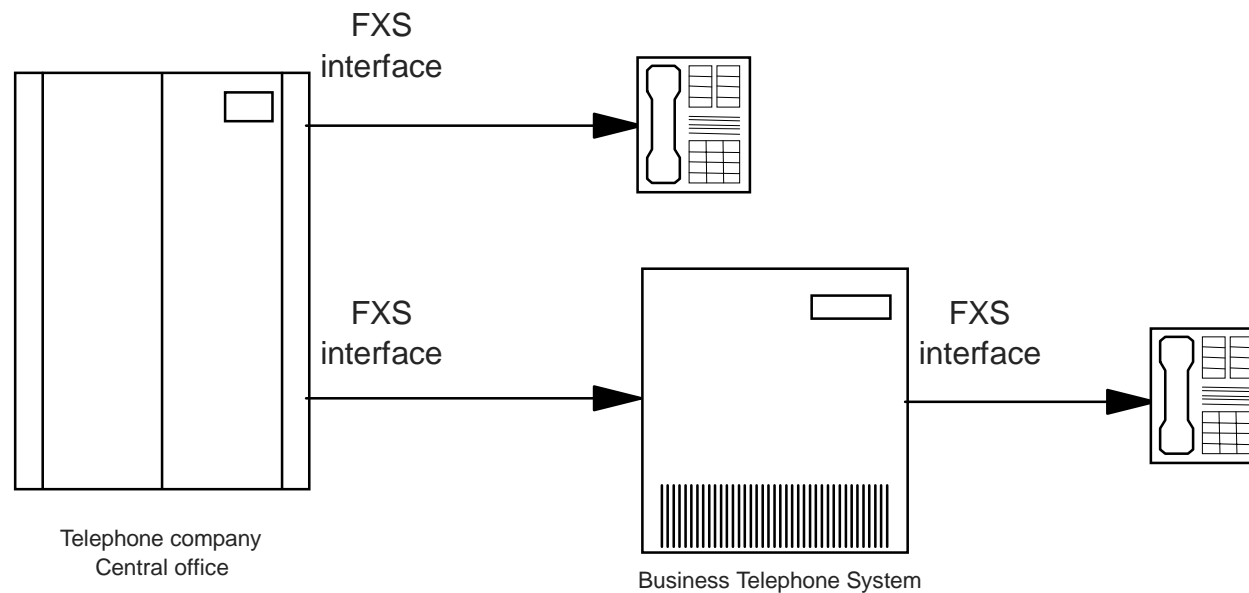
- The terminology defines the relationship between devices in a loop-start network
 - It identifies device functions within the network
 - It enables telecom people to quickly identify and integrate devices into the network
- FXO connects to FXS
- An example of an FXO device is a standard telephone
- An example of an FXS device is the line side of a telephone central office.

FXO

- Foreign Exchange Office
 - Receives battery
 - Receives dial tone
 - Receives ringing



- Foreign Exchange Station
 - Supplies battery
 - Supplies dial tone
 - Supplies ringing

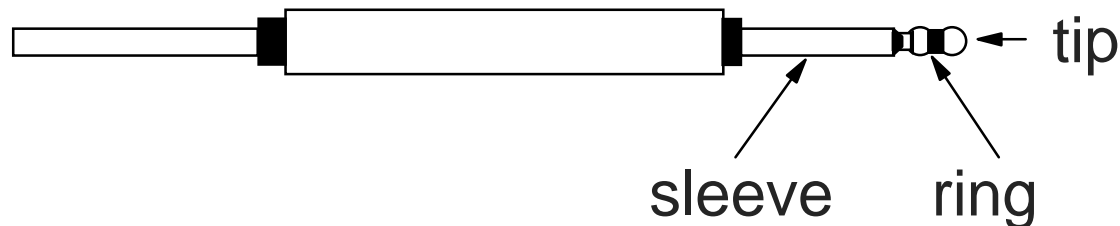


▼ Tip & Ring

The names of voice wires

Always implemented as a twisted pair of copper wires

Named for physical appearance of telephone plugs in manual switchboards



In 4-wire circuits the second voice pair is Tip1 and Ring1.

- T & R go into the cloud; T1 & R1 come out of the cloud.

When DC is present Tip is normally more positive and Ring more negative.

- The opposite condition is called *Reverse Battery*.

When wiring circuits always connect Tip to Tip (or Tip1) and Ring to Ring (or Ring 1), even when making loopbacks or crossovers.

Sometimes called P1 and P2 (in UK), or + and -

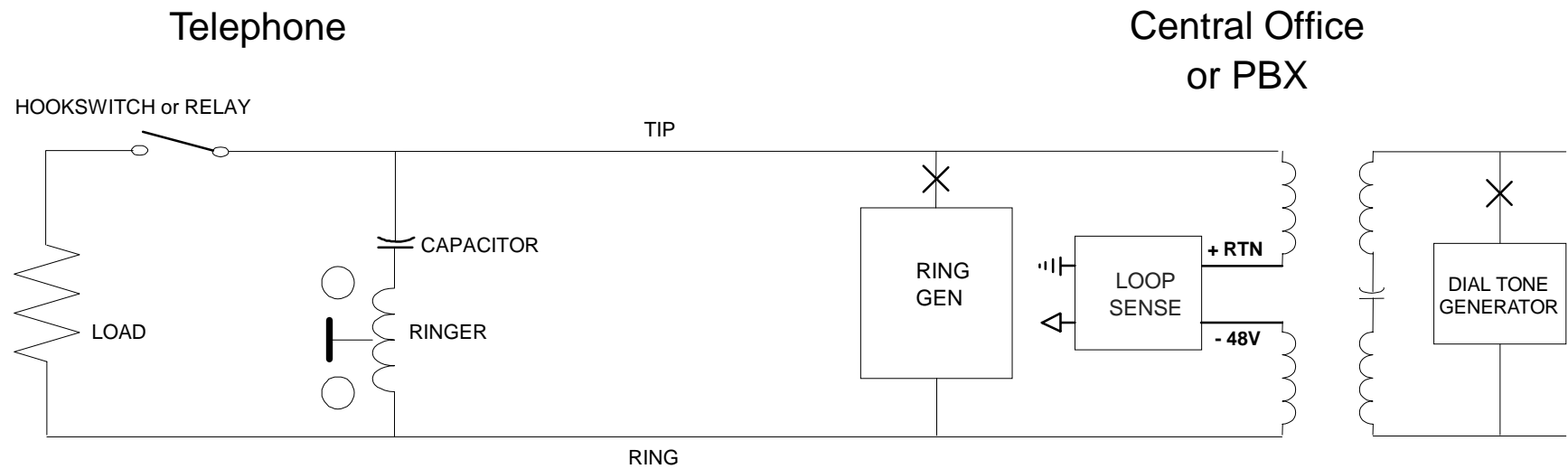


Loop Start Signaling

- Simple signaling scheme
 - Used between telephone and central office
 - Used between PBX and central office
- One side closes a DC current loop which is detected at the other side
- Two wires
 - Tip
 - Ring
- Reverse signaling is by Ring Voltage
 - 90 volts
 - 20 Hertz



Loop Start Signaling





Loop Start Signaling

Call initiated by FXO

Typically the FXO is an analog telephone.

- Idle - handset "On Hook"
 - Hookswitch is open
 - No current flows
- Start - handset "Off Hook"
 - Hookswitch closes the DC loop
 - Current flows from CO battery through the loop and telephone, back to CO ground.
 - CO senses the loop current, provides dial tone, connects a register to receive digits.
- Disconnect - handset returns "On hook"
 - Hookswitch opens loop
 - CO detects open loop, disconnects the call thru the network

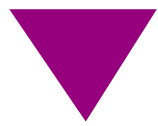


Loop Start Signaling

Call initiated by FXS

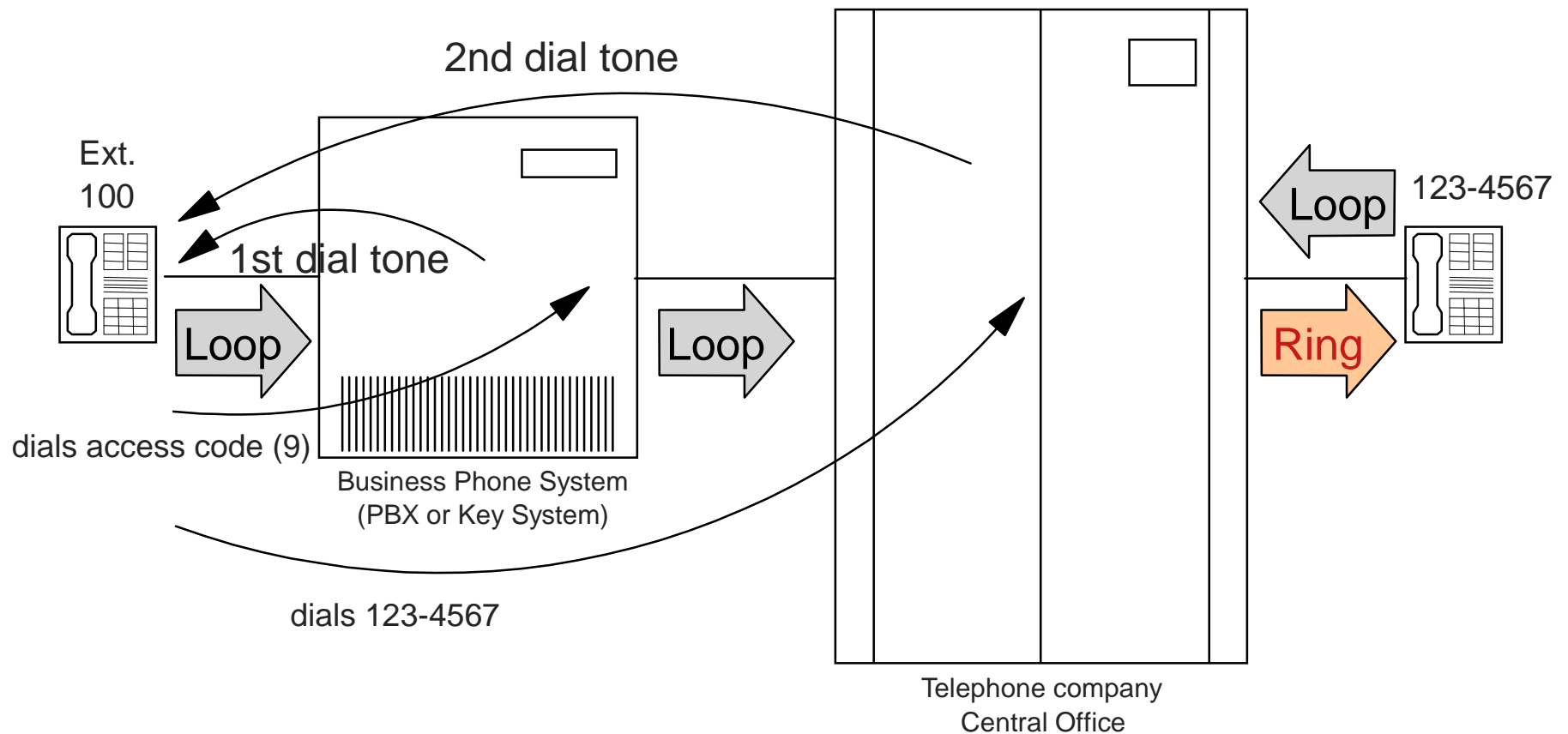
Typically the FXS is a CO or PBX.

- Idle - handset "On Hook"
 - Hookswitch is open
 - No DC current flows
- Alerting - handset still "On Hook"
 - CO applies ring voltage to the loop
 - AC current passes through capacitor and powers ringer.
- Start - handset "Off Hook"
 - Hookswitch closes the DC loop
 - Current flows from CO battery through the loop and telephone, back to CO ground.
 - CO senses the loop current, disconnects ring voltage, and connects the incoming call voice path.
- Disconnect - handset returns "On hook"
 - Hookswitch opens loop
 - CO detects open loop, disconnects the call thru the network



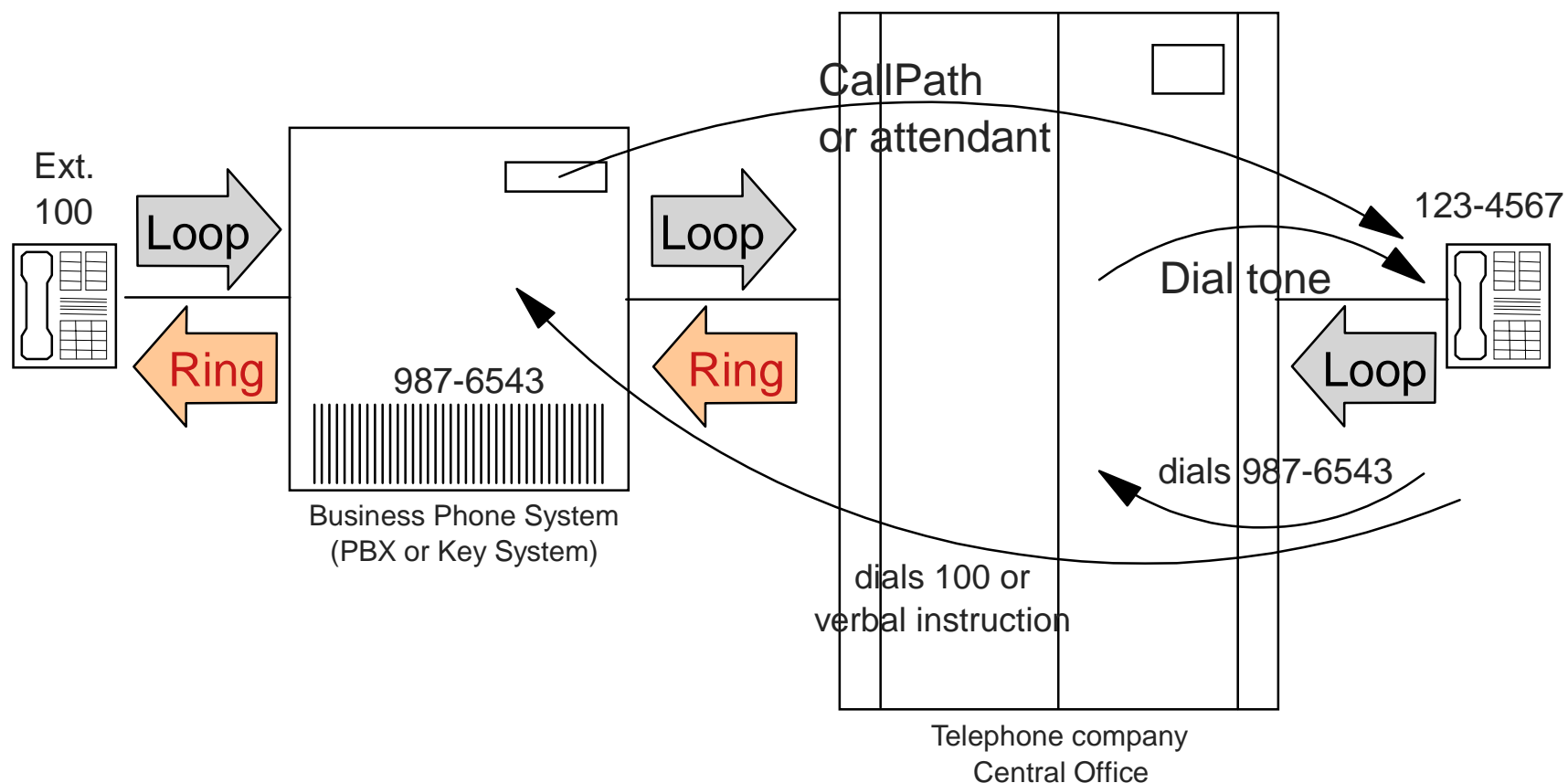
Loop Start Signaling

Typical loop start network with a PBX - PBX initiated call



▼ Loop Start Signaling

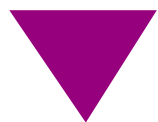
Typical loop start network with a PBX - PSTN initiated call





E&M Signaling

- Used for trunk - to - trunk connections
- Separate wires for speech and supervision
 - 2W E&M uses one pair for both directions of speech transmission (T,R).
 - 4W E&M uses two speech pairs, one for each direction of transmission (T,R,T1,R1).
 - Supervision uses 2 or 4 additional wires (E,M,SB,SG)
 - Five "types" of supervision.
 - Type 1 through 5, sometimes I through V
 - Types 1,2,5 most common

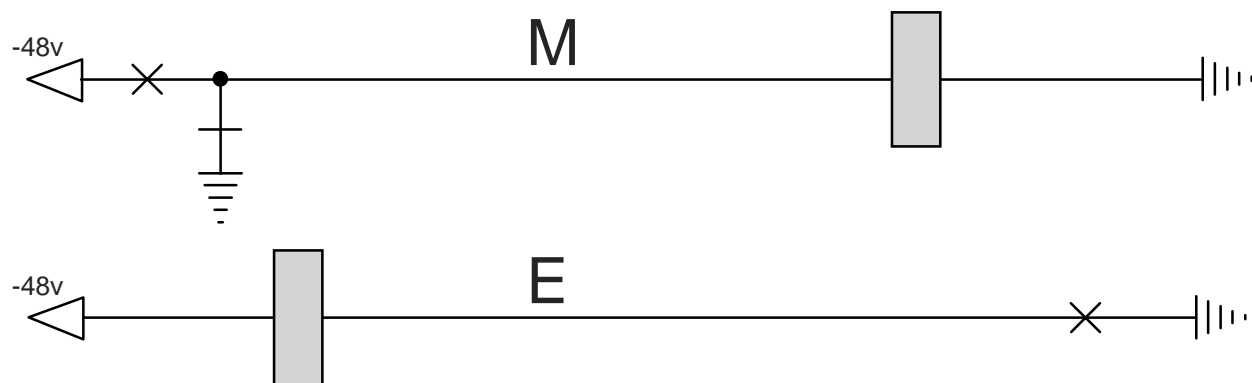


E & M Signaling

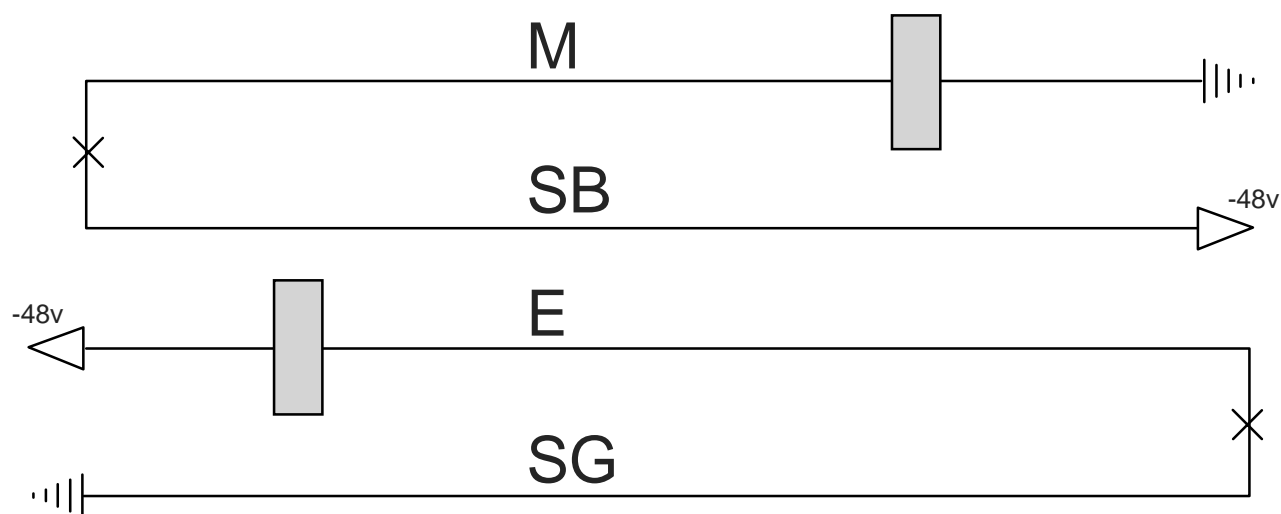
Switch Side (PBX)

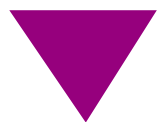
Network Side (2212/9783)

Type
1



Type
2



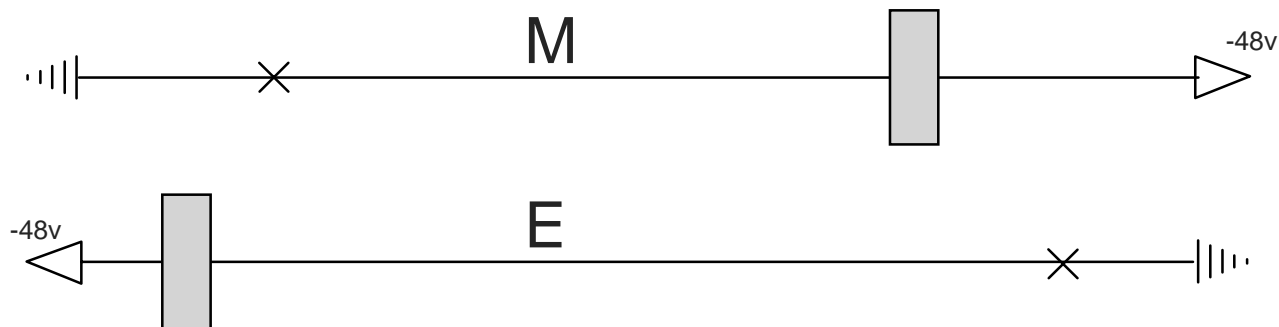


E & M Signaling

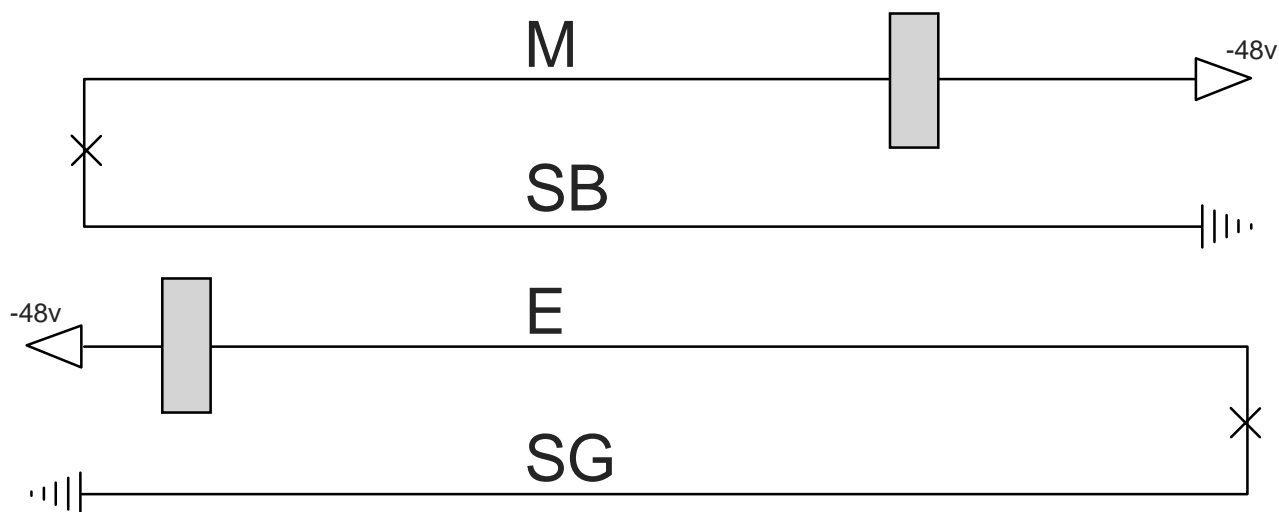
Switch Side (PBX)

Network Side (2212/9783)

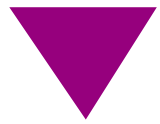
Type
5



Type
4

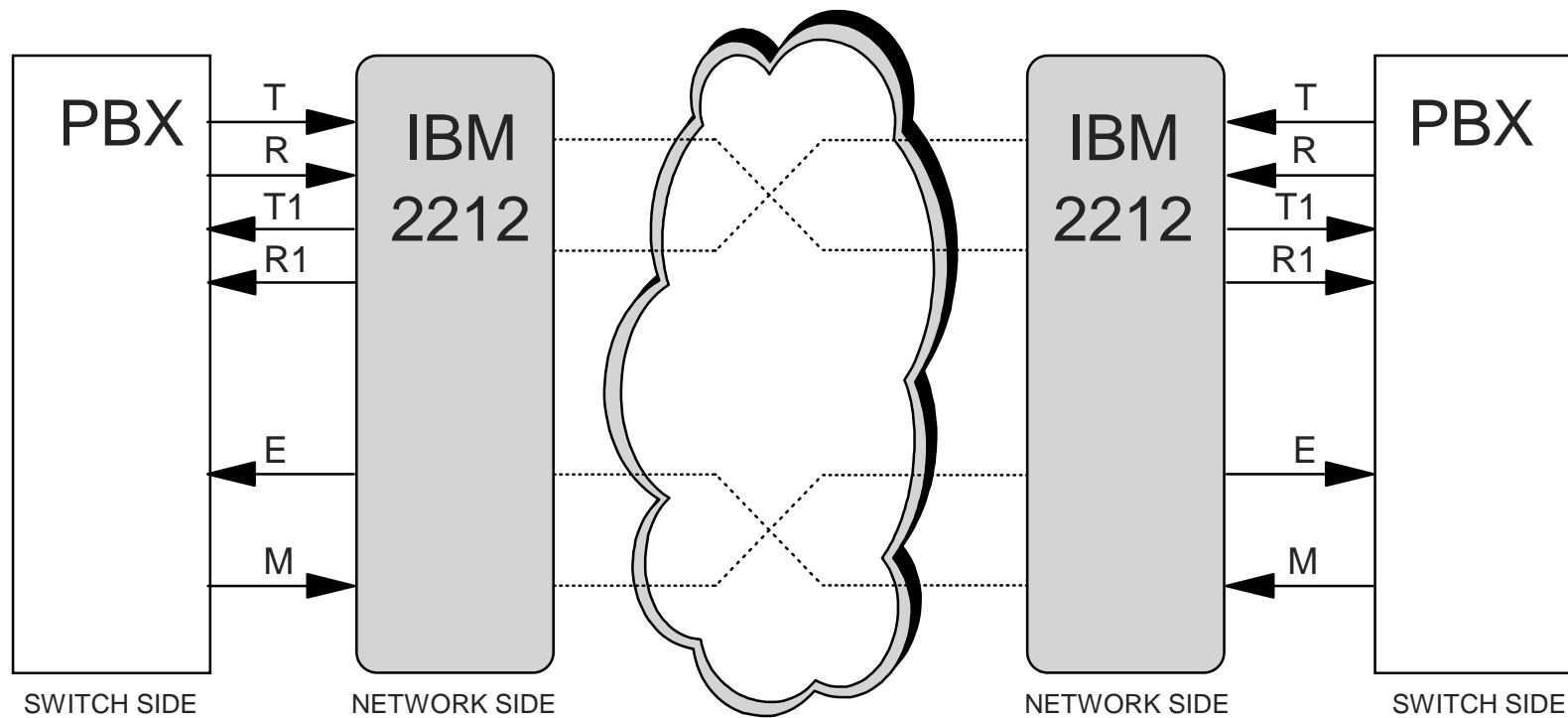


Not supported
by IBM 2212,
supported by
IBM 9783



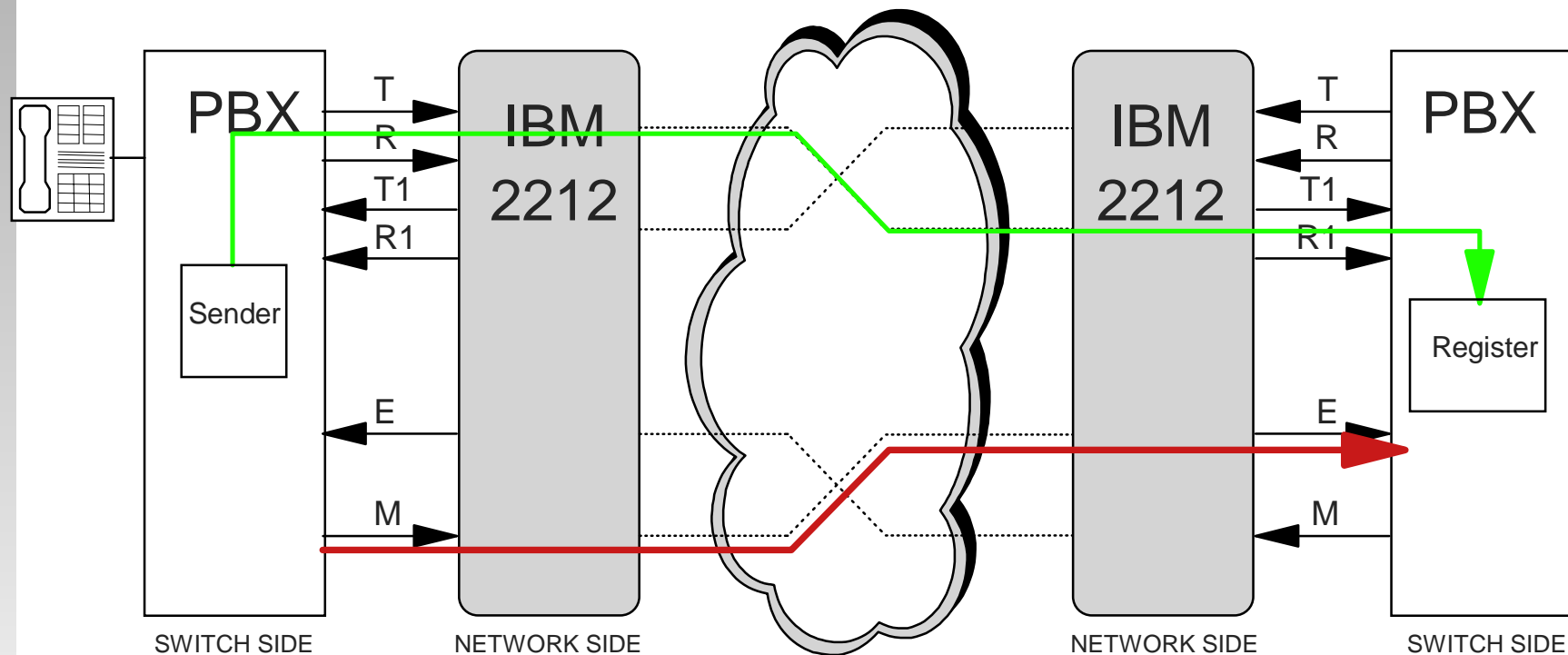
E & M Signaling

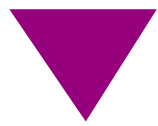
4-Wire E & M



E & M Signaling

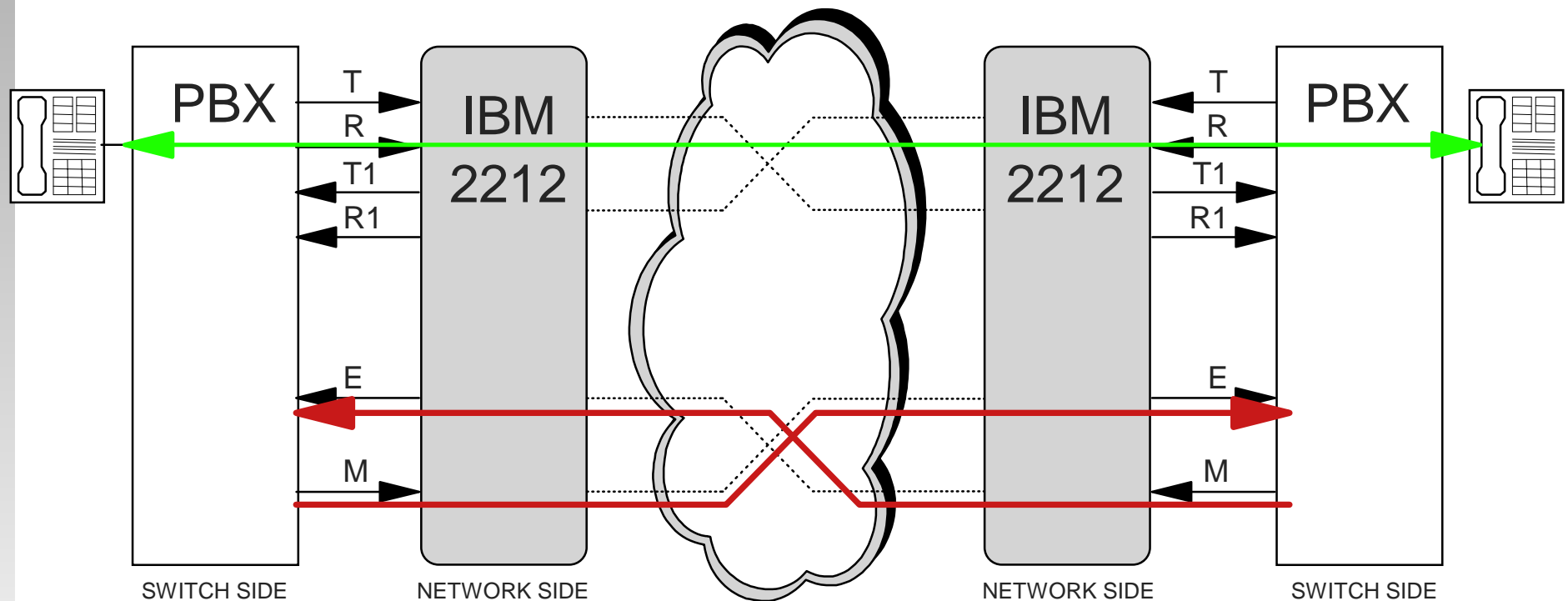
Immediate Start - Sieze & Send





E&M Signalling

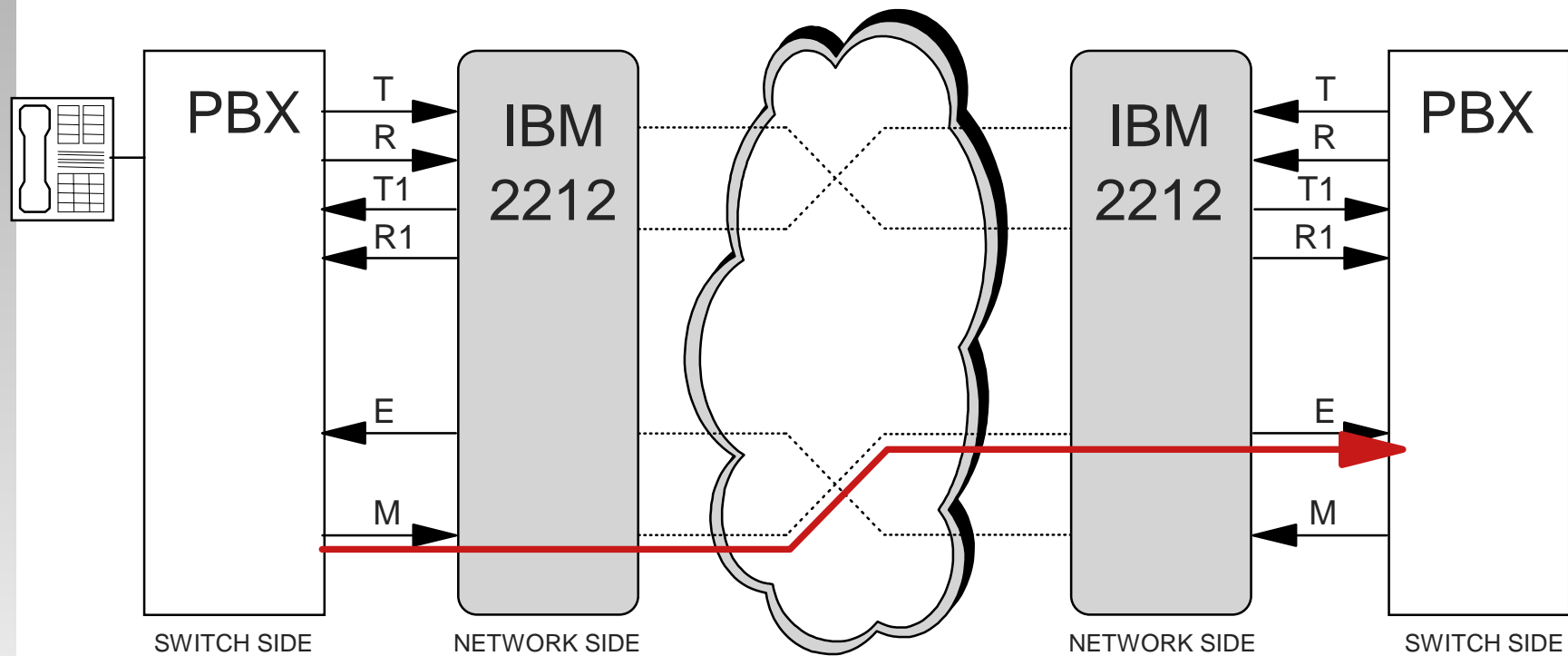
Immediate Start - Answer Supervision

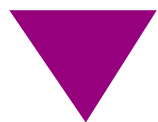




E & M Signaling

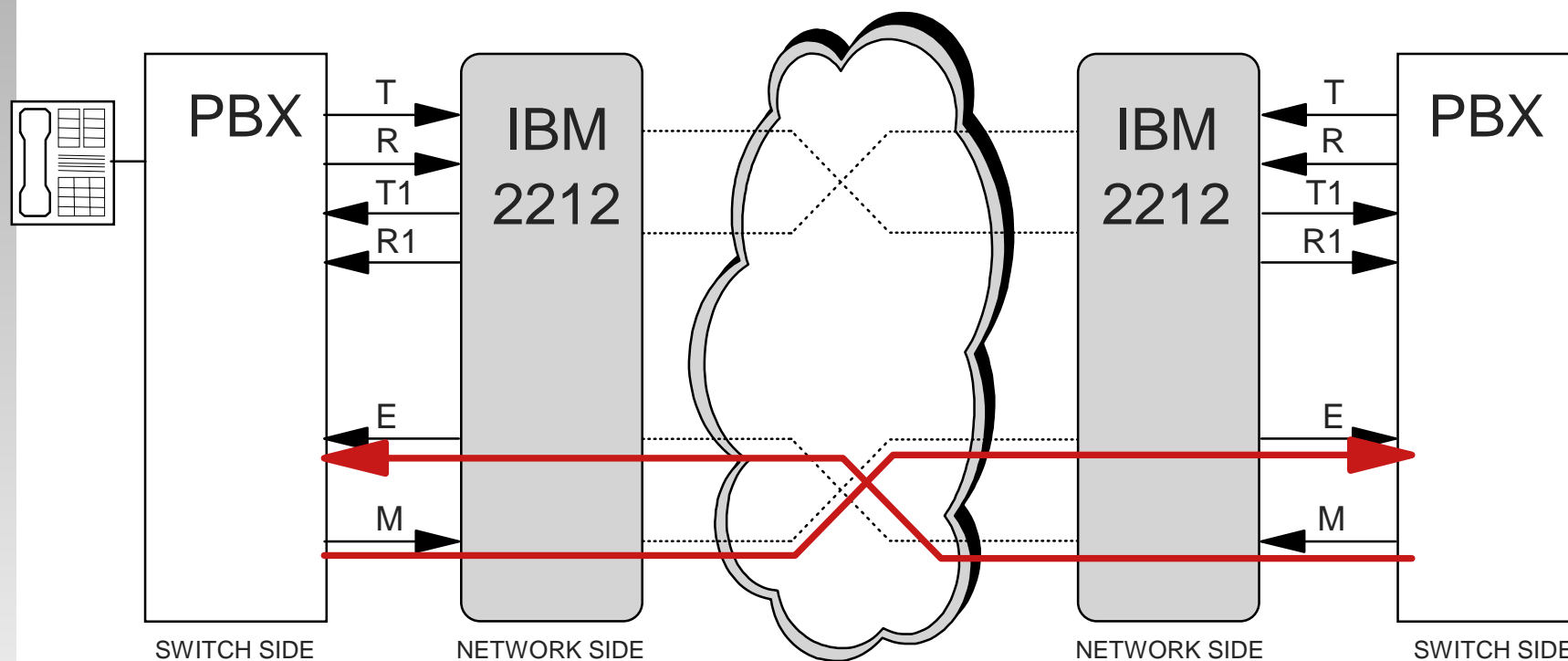
Wink Start - Sieze

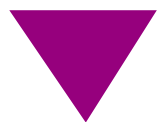




E & M Signaling

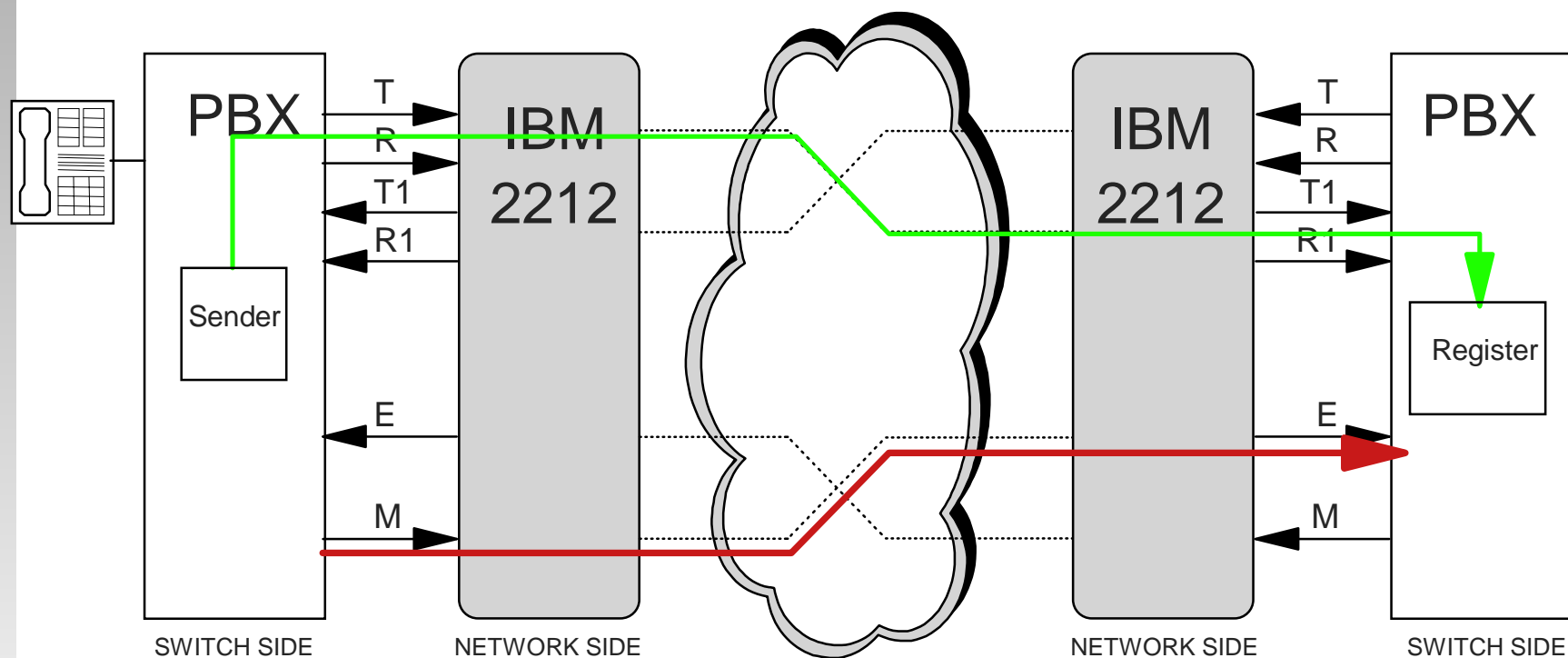
Wink Start - Acknowledge (Stop dial)

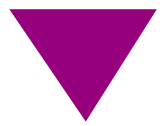




E & M Signaling

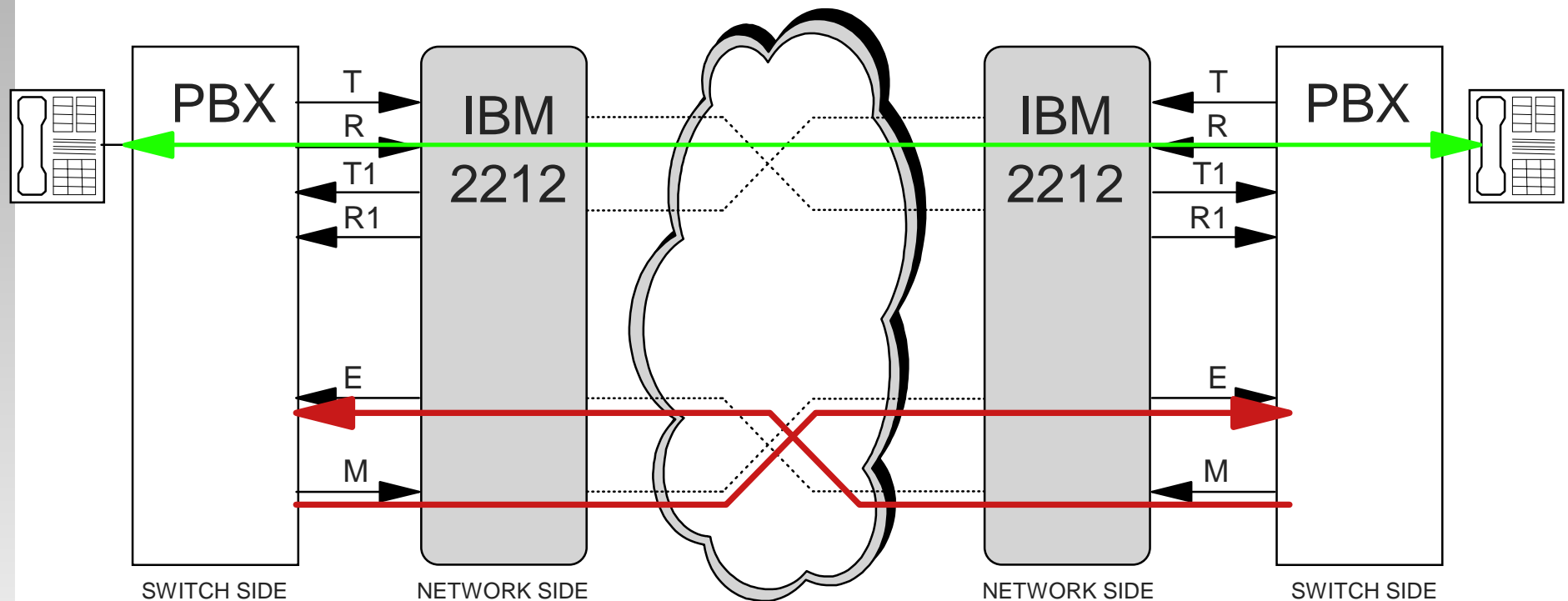
Wink Start - End of wink (Start dial)





E & M Signaling

Wink Start - Answer Supervision



▼ E & M Signaling

Wink start call setup

Originating switch -

- Receives number from originating subscriber
- Selects route based on translation
- Changes M lead to "off hook" to seize terminating switch

Terminating switch -

- Acknowledges seizure with M "off hook"
- Connects register to receive digits
- Changes M back to "on hook"



Originating switch -

- Sends address tones to terminating switch
- Completes voice path to allow progress tones to originator

If called subscriber answers terminating switch changes M to "off hook"



E & M Signaling

Call disconnect

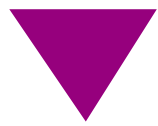
With call in progress, both M leads are "off hook"

If either party disconnects, the local switch will

- Break the local connection
- Drop the M lead (go on hook)

The distant switch will see its E lead go on hook, and disconnect its subscriber.

When both M leads are on hook, the trunk is available for a new call after a timeout.

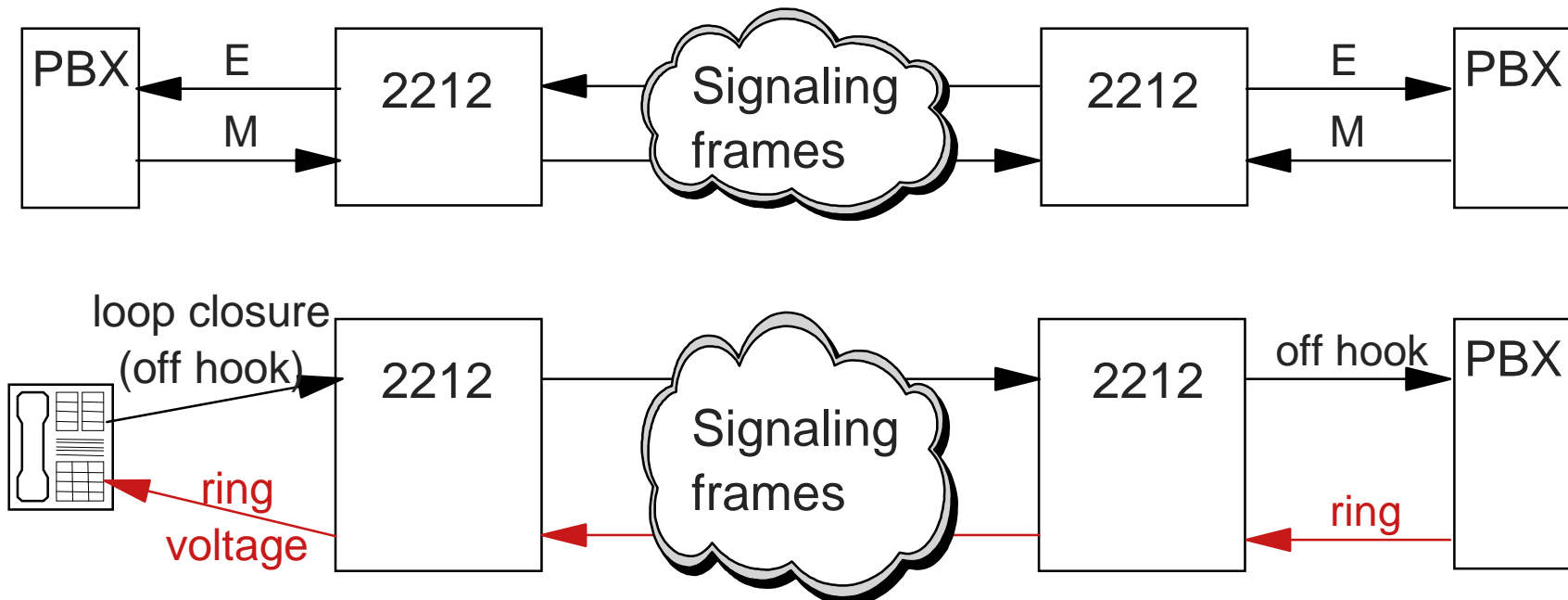


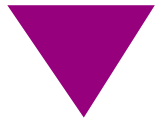
Network Signaling

Loop and DC signaling will not pass through the network cloud. They must be converted to a form compatible with the network.

- Channel Associated Signaling - bit streams
- VoFR - messages in dedicated (sub) DLCIs
- Common Channel Signaling - Q.931 messages

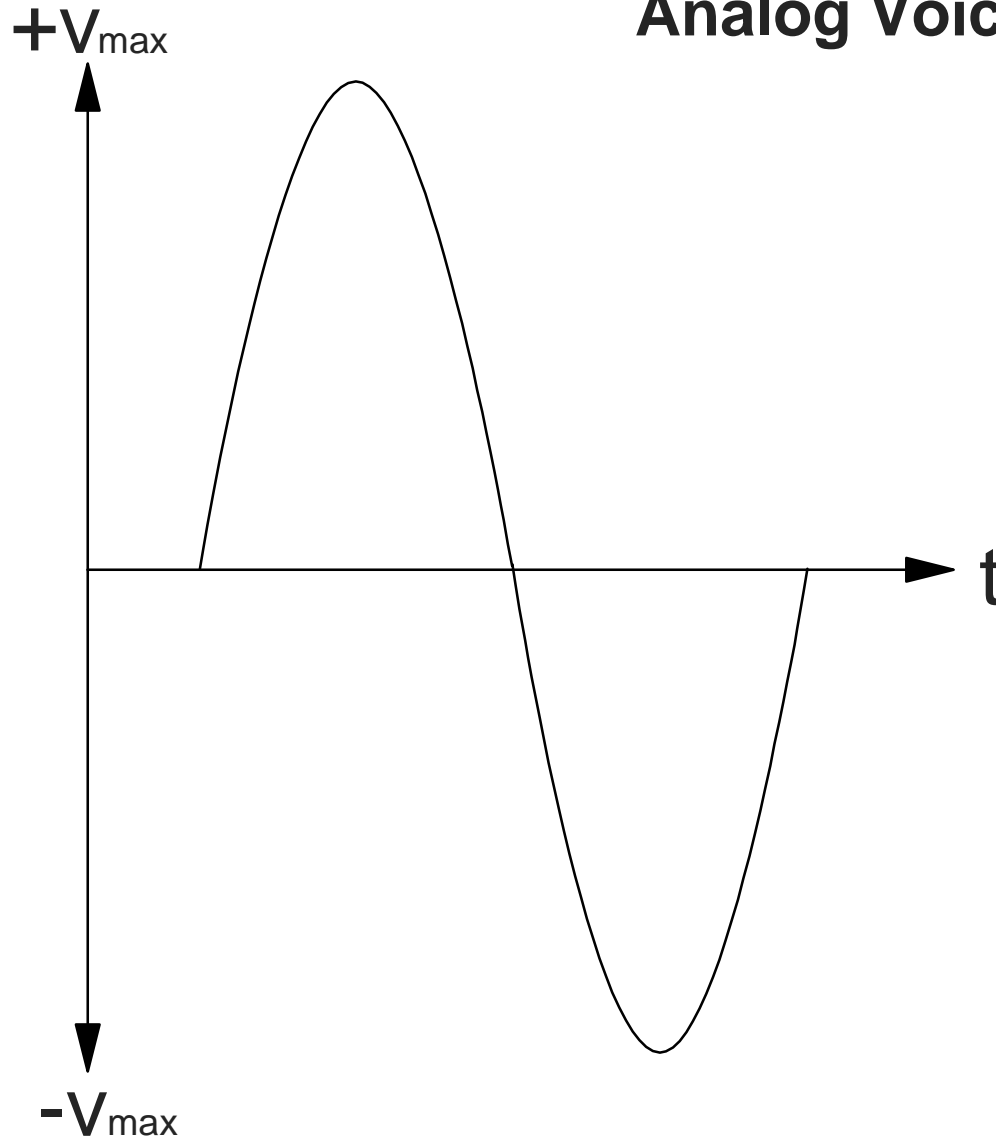
What happens to the analog supervision when put on a frame relay trunk?





Voice

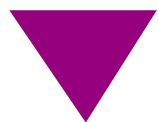
Analog Voice



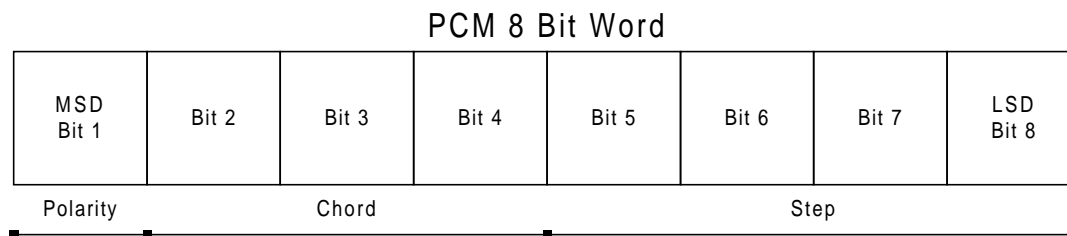
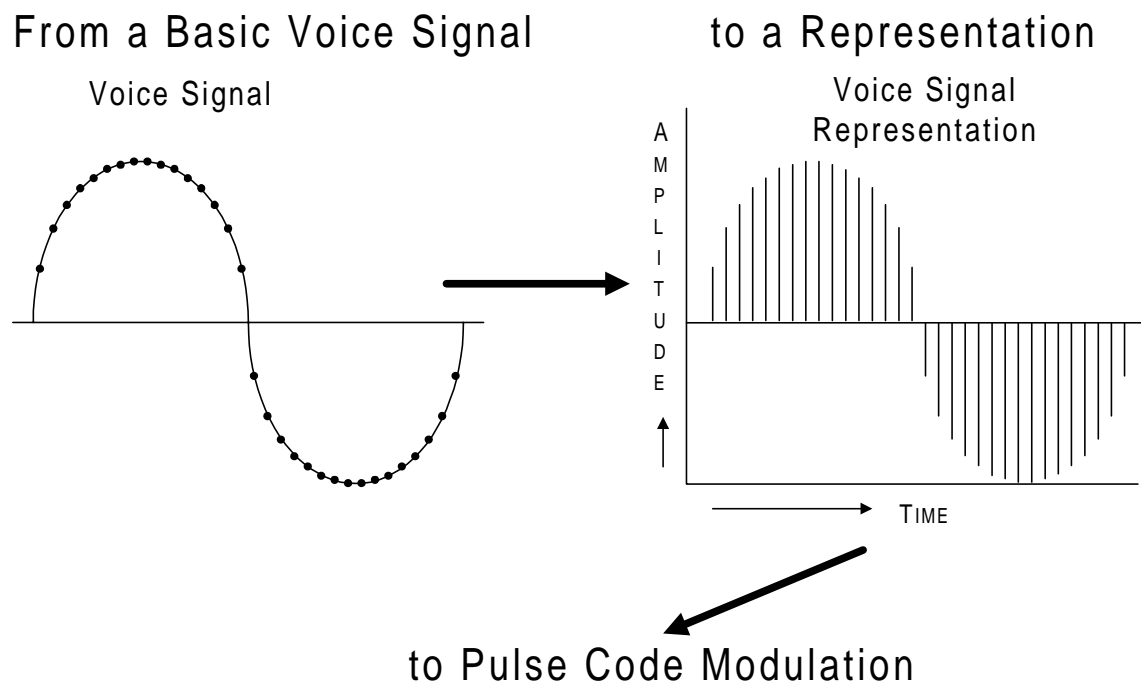
"Voice" begins as:
Analog speech in air
Analog tones in modem

Voice signals are:
Periodic, non-sinusoidal
Band limited (300-3400 Hz)
Low power ($\ll 1$ mW)
Low voltage (< 1 V)

For transport over packet networks, analog voice must be digitized.



Digitizing Voice

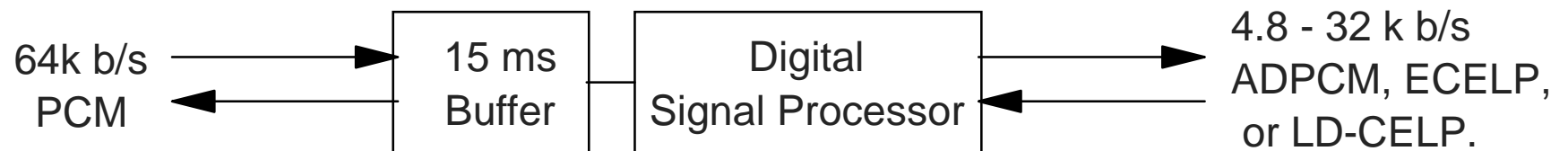


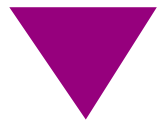
Standard PCM is 8000 samples/second, 8 bits/sample = 64k bit/sec



Compressing Voice

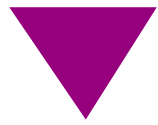
- Voice compression reduces the network bandwidth needed to transport voice.
- Allows voice on low speed (<64k) lines
- Two common techniques
 - Waveform coders (ADPCM) try to copy the input
 - Source coders (CELP) try to deliver the sound of speech
- Compression affects
 - Delay
 - Quality
 - Transport of voiceband data (modems, FAX)





Compressing Voice

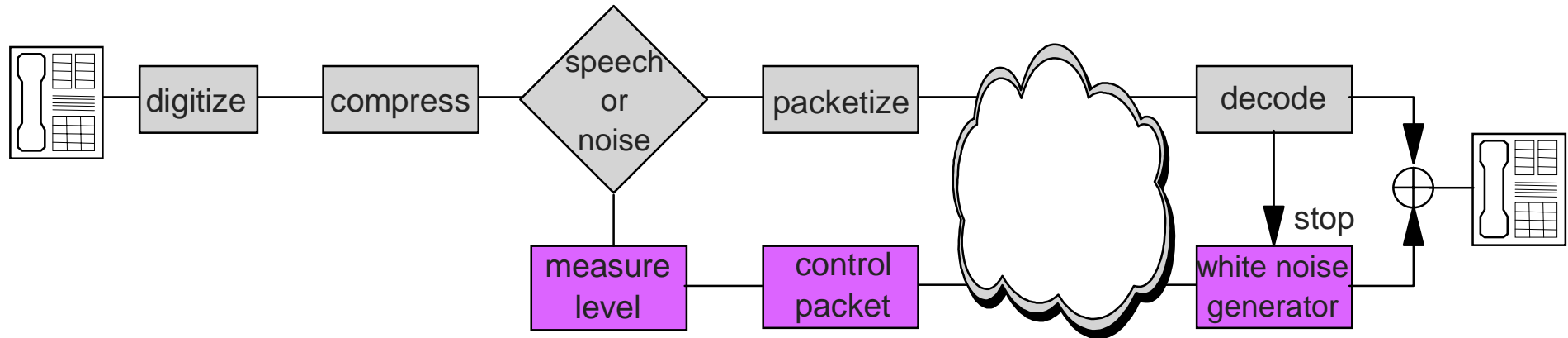
- IBM 2212 and 9783 Voice cards use onboard Digital Signal Processor for Voice Compression. Available algorithms are:
 - "Standards Suite"
 - G.726 ADPCM at 32 Kbps
 - G.728 LD-CELP at 16 Kbps
 - G.729 CS-CELP at 8 Kbps
 - " Proprietary Suite"
 - E CELP at 4.8 Kbps
 - E CELP at 7.47 Kbps
 - E CELP at 9.6 Kbps
 - G.726 ADPCM at 32 Kbps
- When configuring, you select the suite and the rate. If two routers are configured differently, they will use a common rate or 32 Kbps.



Removing silence

Voice Activity Detection (VAD)

Silence removal is a natural function of a packet voice network.

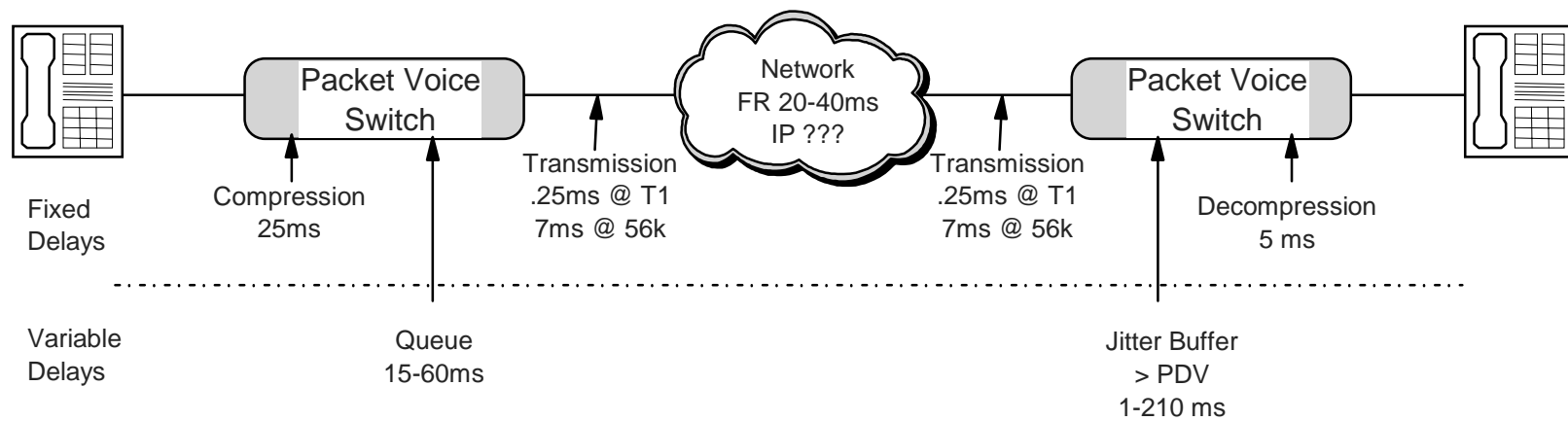


Time averaged bandwidth saving is about 50%

To be safe, estimate 33% saving

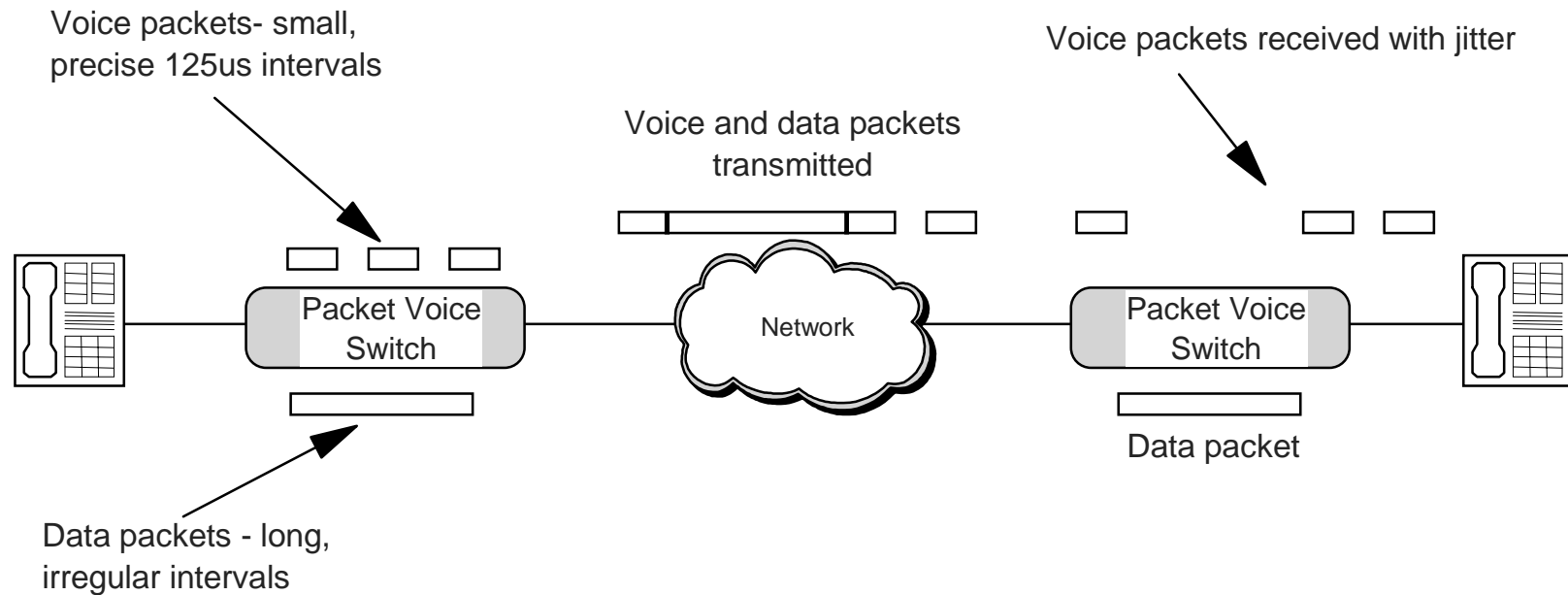
Network Delay

High quality service - up to 150 msec one-way
Fair quality service - up to 400 msec one-way





Jitter



Jitter avoidance

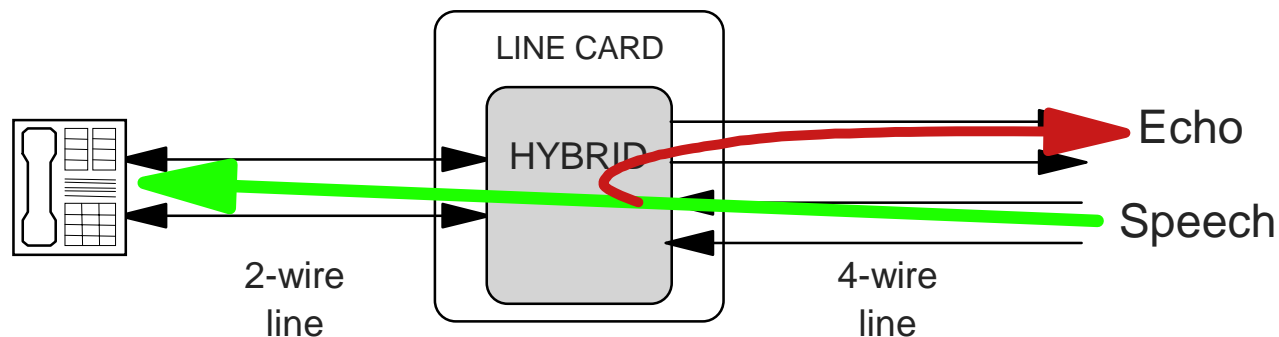
- Small data packets
- Data packet segmentation/reassembly - FRF.12
- Voice packet priority
- Dynamic playout buffer size
- Missing packet algorithms

▼ Echo

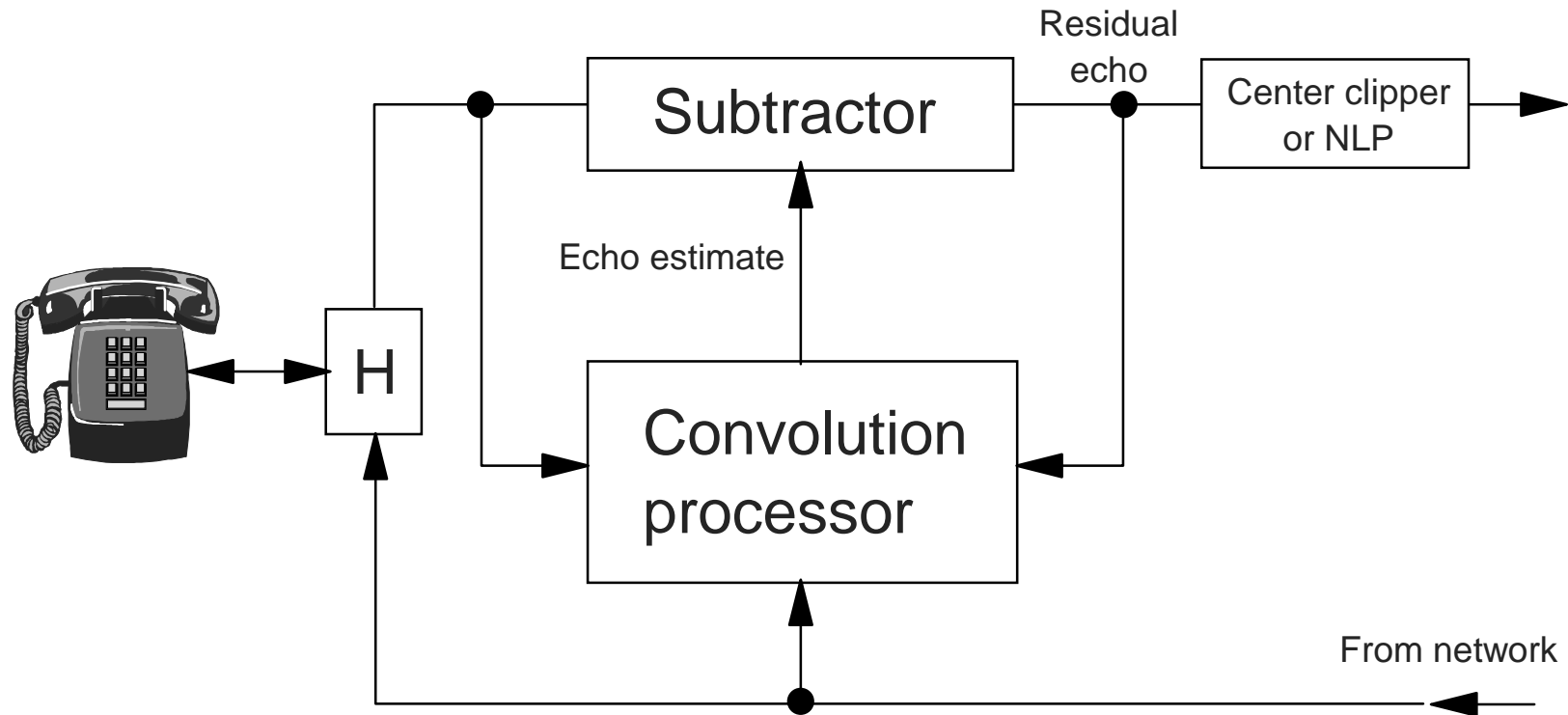
Talker Echo is the return of voice energy to the talker's ear.

- Echo is objectionable if the round-trip delay of the circuit exceeds 40 ms.
- Therefore echo is a consideration for ALL VoFR and VoIP networks

In telephony the principal source of echo is hybrids.



▼ Echo canceller

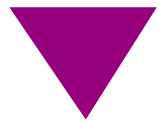


Two-stage echo canceller

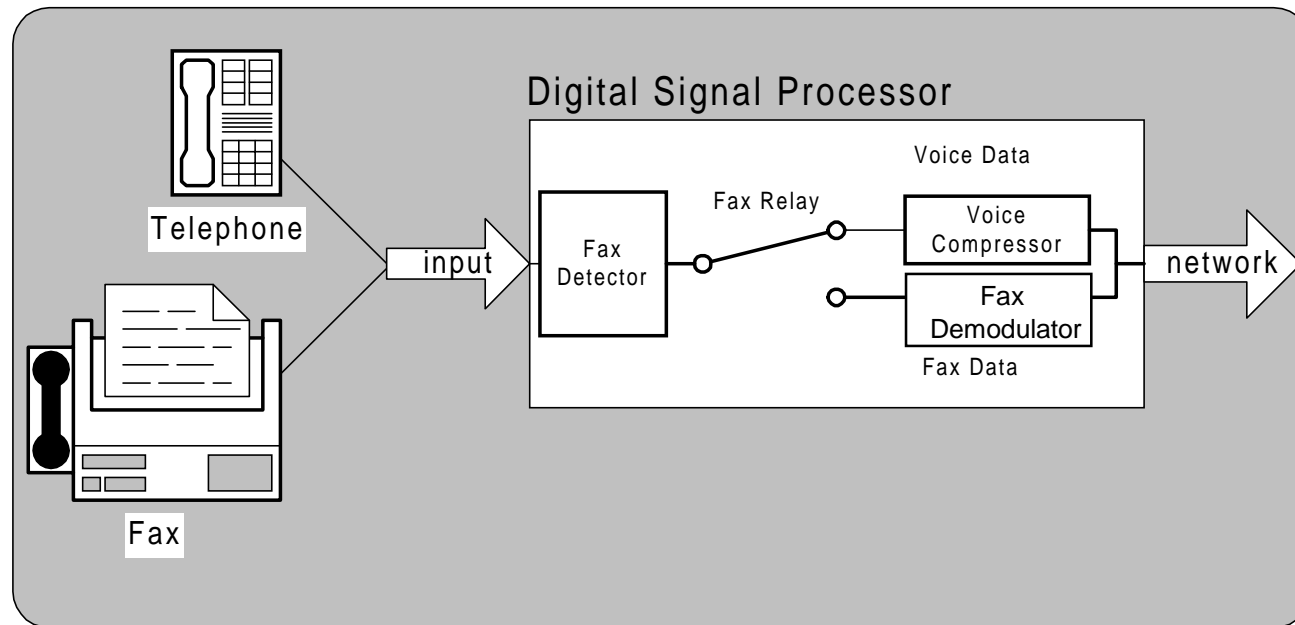
- Estimates echo and subtracts it from returning signal
- Deletes remaining low-level speech energy

Echo cancellers may interfere with some data equipment

- Cancellor should shut down upon detecting 2100Hz tone ("fax tone").



Fax Relay



2212 and 9783 change DSP algorithm

- Demodulate FAX to data rate
- Re-modulate at far end of network

Negotiate both fax machines to 9.6k or lower