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VoIP Troubleshooting From the Headend

Ruth Cloonan

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Agenda

- VoIP Defined
- PacketCable™ VoIP Architecture
- Basic Tools for Troubleshooting VoIP
- Advanced VoIP Troubleshooting Tools
- Conclusions

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What is VoIP?

- **VoIP** in and of itself is **NOT** a "service" – it is a term used to describe the application of IP based digital technology to transmit voice over a digital network.

- **VoIP**, in the DOCSIS access loop, can provide the same customer experience as traditional telephone services

- **VoIP** technology can be transparent to the user, with the same telephony features and reliability of the traditional telephone network – AND MORE!

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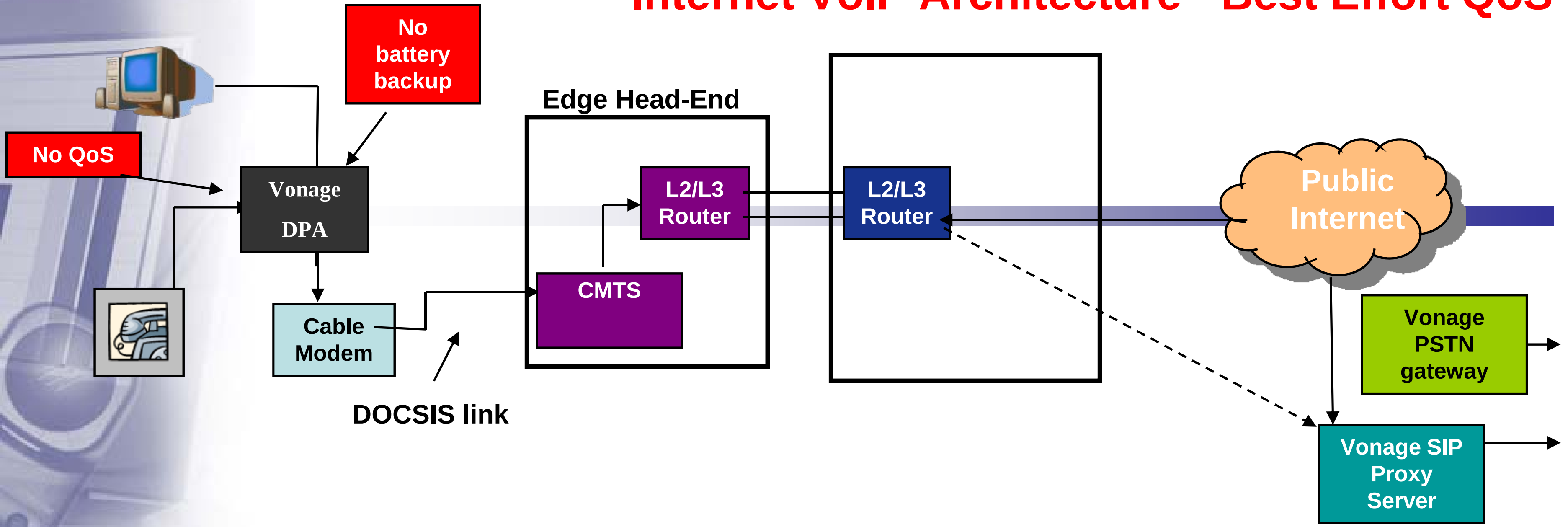
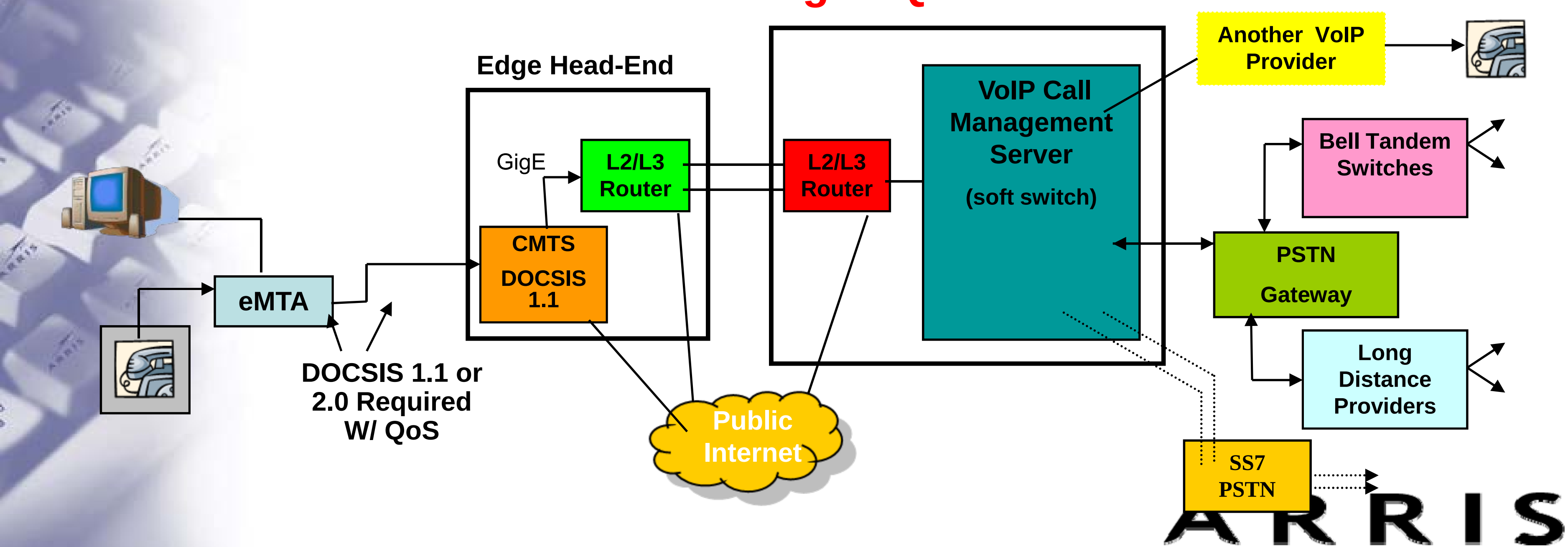
Types of VoIP over Cable

- **Internet VoIP**
  - Voice is delivered over the public Internet
  - Based on Session Initiated Protocol (SIP)
  - Uses Analog Telephony Adapter (ATA) or computer connected to Cable Modem
  - Best-effort: No Quality of Service (QoS), not generally suitable for primary line service
- **PacketCable™ VoIP**
  - Standards based architecture defined by CableLabs®
  - Voice is delivered over managed IP network
  - NCS/MGCP with centralized call management
  - Uses Embedded Multimedia Terminal Adapter (EMTA)
  - QoS protocol to ensure primary line voice quality

Internet VoIP Architecture - Best Effort QoS



PacketCable VoIP Architecture- Managed QoS



How does “PacketCable VoIP” differ from “Internet VoIP”?

- PacketCable offers Priority Quality-of-Service (QoS)
- PacketCable is designed to be more secure & reliable
- PacketCable supports E-911, Operator Assistance, Directory Assistance, & CALEA
- PacketCable allows the use of conventional telephones
  - Interfaces with analog CPE and existing building wiring.
- PacketCable offers Battery backup
- PacketCable qualifies as a “primary line” product
- PacketCable offers the same features offered by the ILEC (TDM switch)… see next slide

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PacketCable Call Features Supported

- Automatic Number Assignment Confirmation
- DTMF Dialing
- Direct Distance Dialing
- Critical Interdigital Timing for Dialing Plan
- International DDD (IDDD) Local Billing Control
- Residence Distinctive Alerting Service
- Free Terminating Service
- Code Restriction & Diversion
- Toll Restricted Service
- CLASSSM : Calling Number Delivery
- CLASSSM : Customer Originated Trace
- CLASSSM : Anonymous Call Rejection
- CLASSSM : Calling Number Delivery Blocking
- CLASSSM : Calling Identity Delivery & Suppression
- CLASSSM : Calling Name Delivery Blocking
- CLASSSM : Calling Name Delivery
- CLASSSM : Calling Identity Delivery on Call Waiting
- Speed Calling 8
- Speed Calling 30
- Call Waiting
- Cancel Call Waiting (*70)
- Call Waiting Deluxe
- Access to Telecommunications Relay Service (TDD)
- Intercept
- Routing for blank/changed/etc. ph#s
- Customer-Changeable Speed Calling
- VIP Alert (Distinctive Ringing)
- Visual Message Waiting Indicator (FSK)
- Message Waiting Tone (stutter dial tone)
- Conference Calling - Six-Way Station Controlled
- Call Hold, Call Pick-up, Toll Free Calling
- E911
- Customer Call Back (Automatic Recall) (*69)
- Three-Way Calling
- Call Forwarding Variable
- Call Forwarding Busy Line
- Call Forwarding - Don't Answer - All Calls
- Service Provider Originated Trace
- Courtesy Ring Generation
- Multiple Directory Numbers on a Line
- Customer Access Treatment (CAT) code restrictions
- Single-Digit Dialing
- Line Number Portability
- Remote Activation of Call Forwarding (RACF)
- Outside Calling Area Alerting (OCAA)

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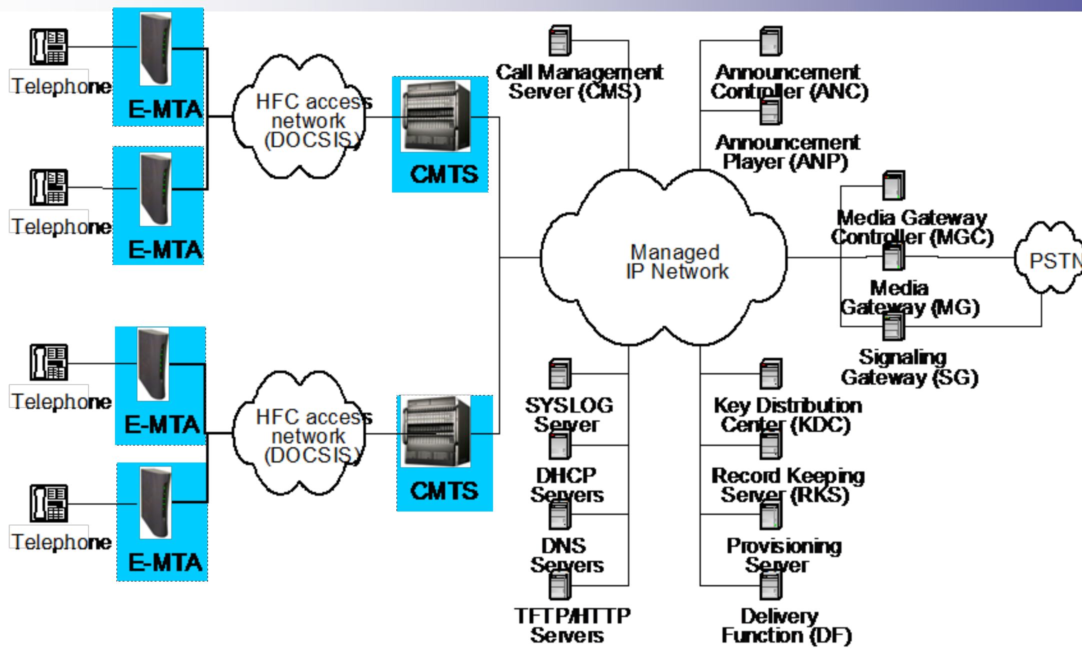
PacketCable 1.x System Overview: System Diagram (EMTA, CMTS)



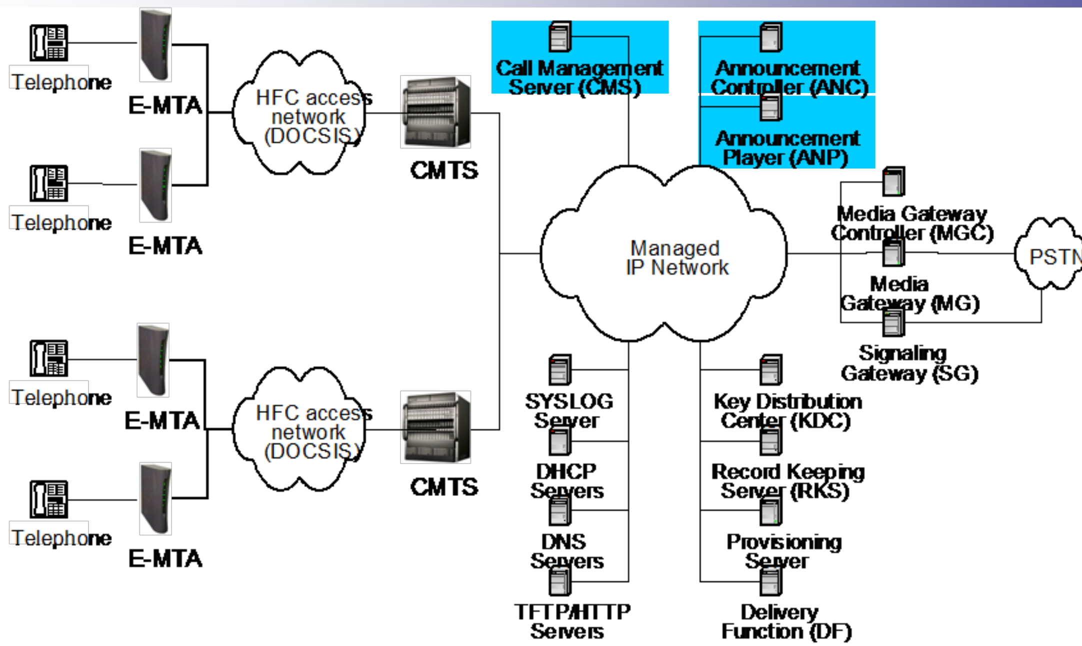
Embedded Multimedia Terminal Adapter (E-MTA)

Single device containing a DOCSIS cable modem and a device that provides one or more POTS interfaces.

Cable Modem Termination System (CMTS)

Provides connectivity between DOCSIS network and PacketCable devices; also performs call authorization, bandwidth allocation, and call trace functions

PacketCable 1.x System Overview: System Diagram (CMS, ANC, ANP)



Call Management Server (CMS)

Provides call control and signaling related services for the MTA, CMTS, and PSTN gateways; typically performs both Call Agent (handles call state) and Gate-Controller (provides authorization) functionality

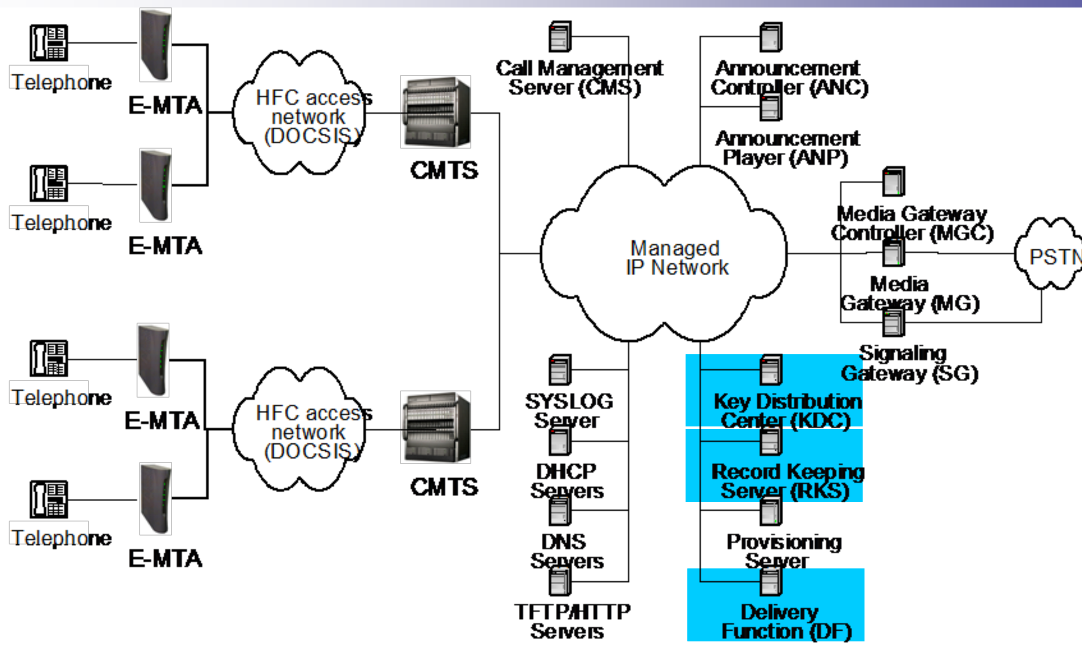
Announcement Controller (ANC)

Initiates and manages all announcement services that are provided by the announcement player

Announcement Player (ANP)

Delivers the appropriate announcement(s) to the MTA under control of the announcement controller

PacketCable 1.x System Overview: System Diagram (KDC, RKS, DF)



Key Distribution Center (KDC)

Performs security key negotiations for MTA and Provisioning Server in the PacketCable network

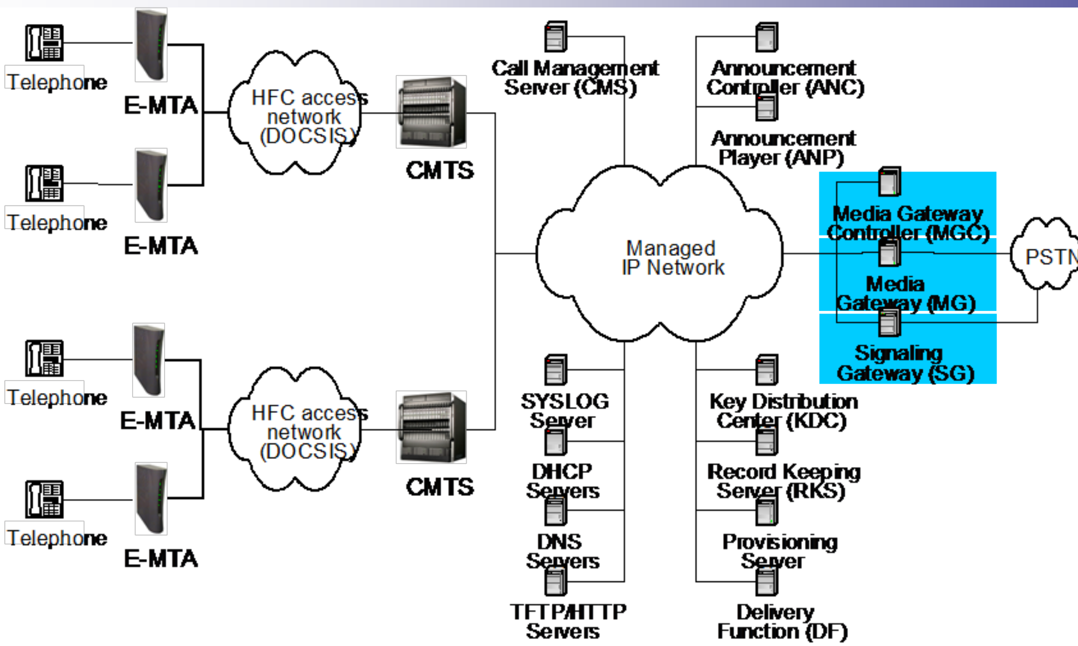
Record Keeping Server (RKS)

Collection point for all PacketCable Event Messages; may create Call Detail Records for billing interfaces

Delivery Function (DF)

Aggregation point for electronic surveillance; delivers reasonably available call-identifying information and call content based on the requirements of lawful authorization

PacketCable 1.x System Overview: System Diagram (MGC, MG, SG)



Media Gateway Controller (MGC)

Provides bearer mediation between the PSTN and the PacketCable network

Media Gateway (MG)

Provides media (voice packets) connectivity between the PSTN and the PacketCable network

Signaling Gateway (SG)

Provides signaling mediation between the PSTN and the PacketCable network

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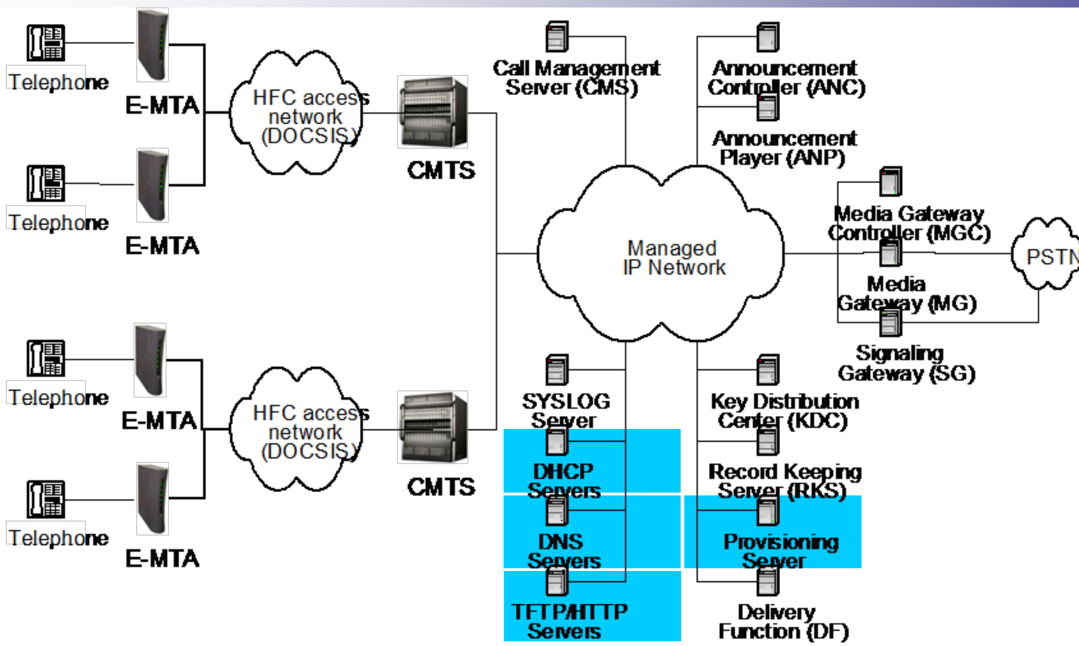
PacketCable 1.x System Overview: System Diagram (PS, DHCP, DNS, TFTP)



Provisioning Server (OSS)

Provides provisioning information for PacketCable devices via SNMPv3

DHCP Servers

Provide initialization information as well as IP address

DNS Servers

Map device Domain Names (foo.bar.com) to IP addresses

TFTP/DHCP Servers

Contain configuration files for device initialization

PacketCable 1.x “Typical” Call Flow

- There is no such thing as a “correct” call flow in PacketCable!
  - PacketCable provides primitives that can be assembled as calls
  - CMS vendors differ from each other (and from reference model)
  - Often, a vendor will change call flow based upon enabled line features
  - All flows valid if they get the job done (some more efficient than others)
- Following is one example of a line on MTA_o calling a line of MTA_T
  - Many other flows exist for this case as well as other cases (MTA-PSTN, PSTN-MTA, etc.)

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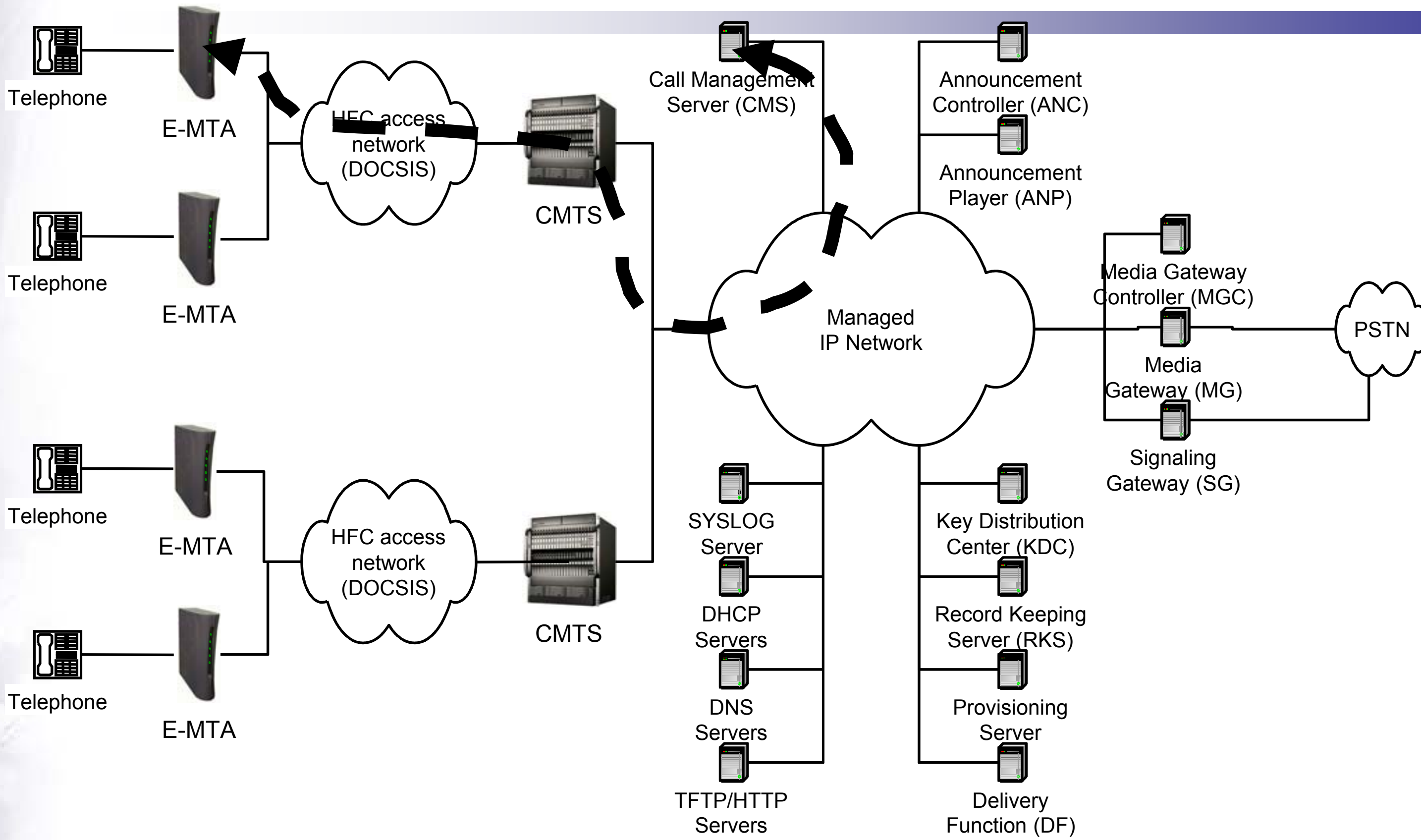
“Typical” PacketCable 1.x Call Flow



When the MTAs first come up, they all register with the CMS and the CMS tells the MTAs to wait for phone to go offhook

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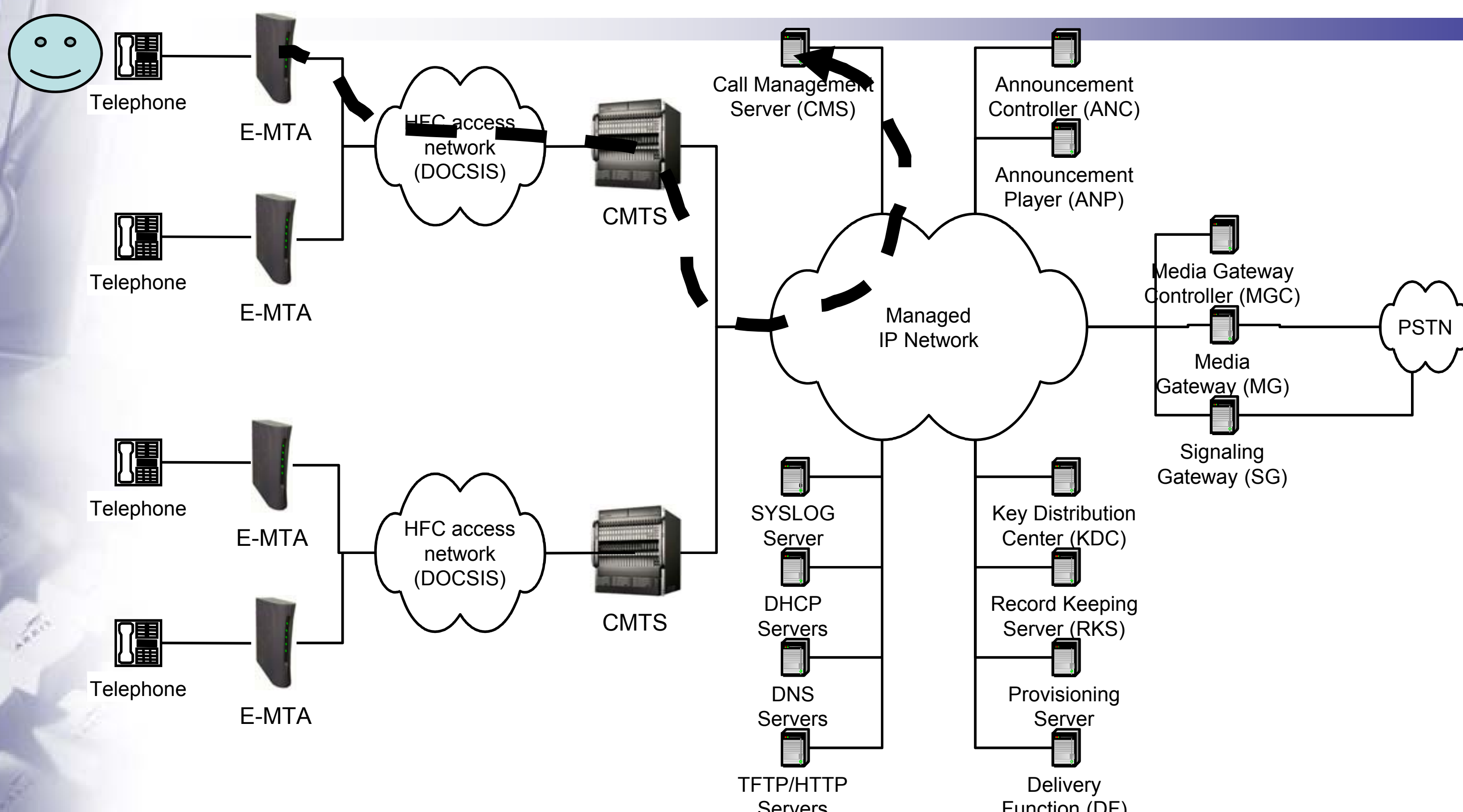
“Typical” PacketCable 1.x Call Flow



User goes offhook on phone to make a call. MTA_O tells CMS

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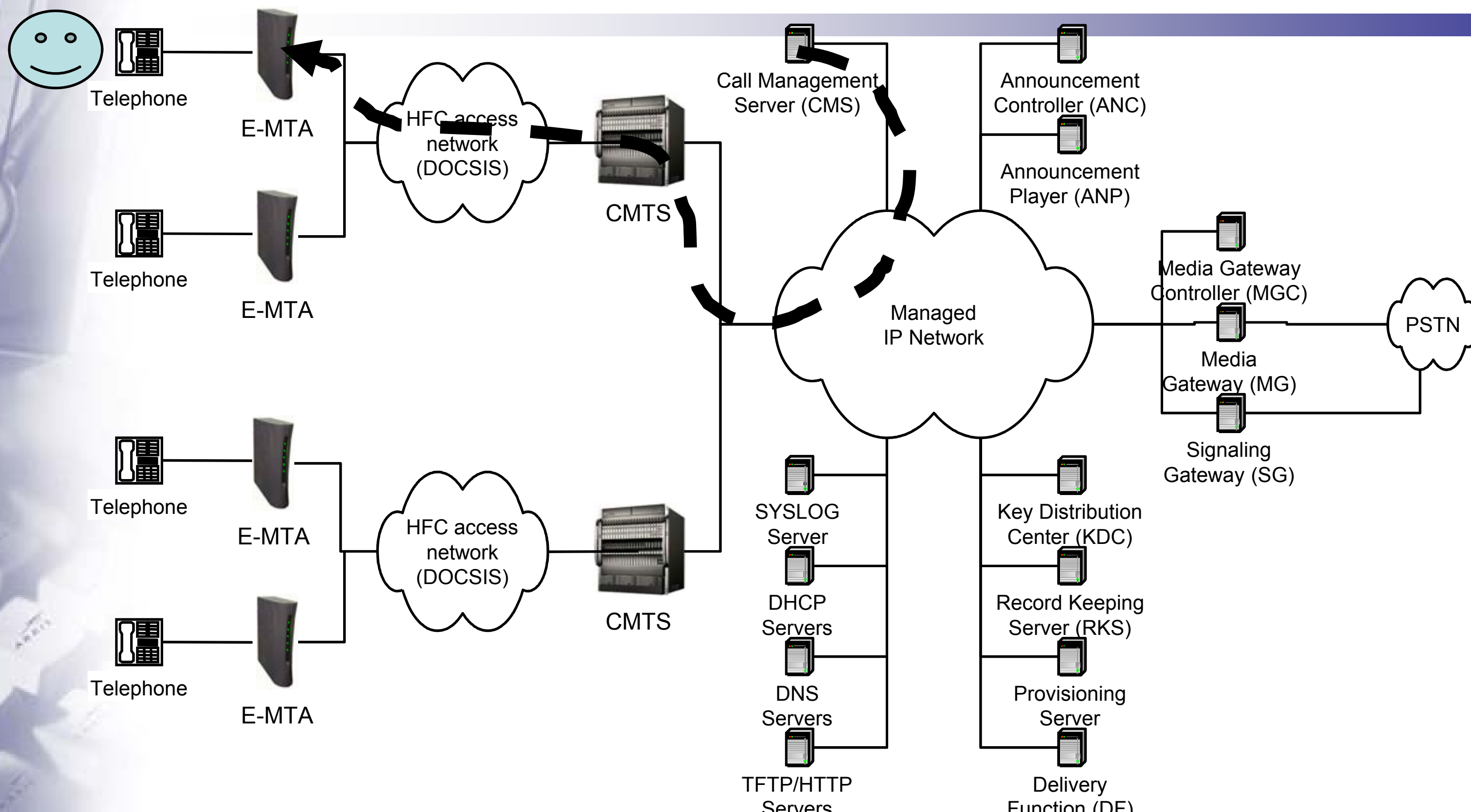
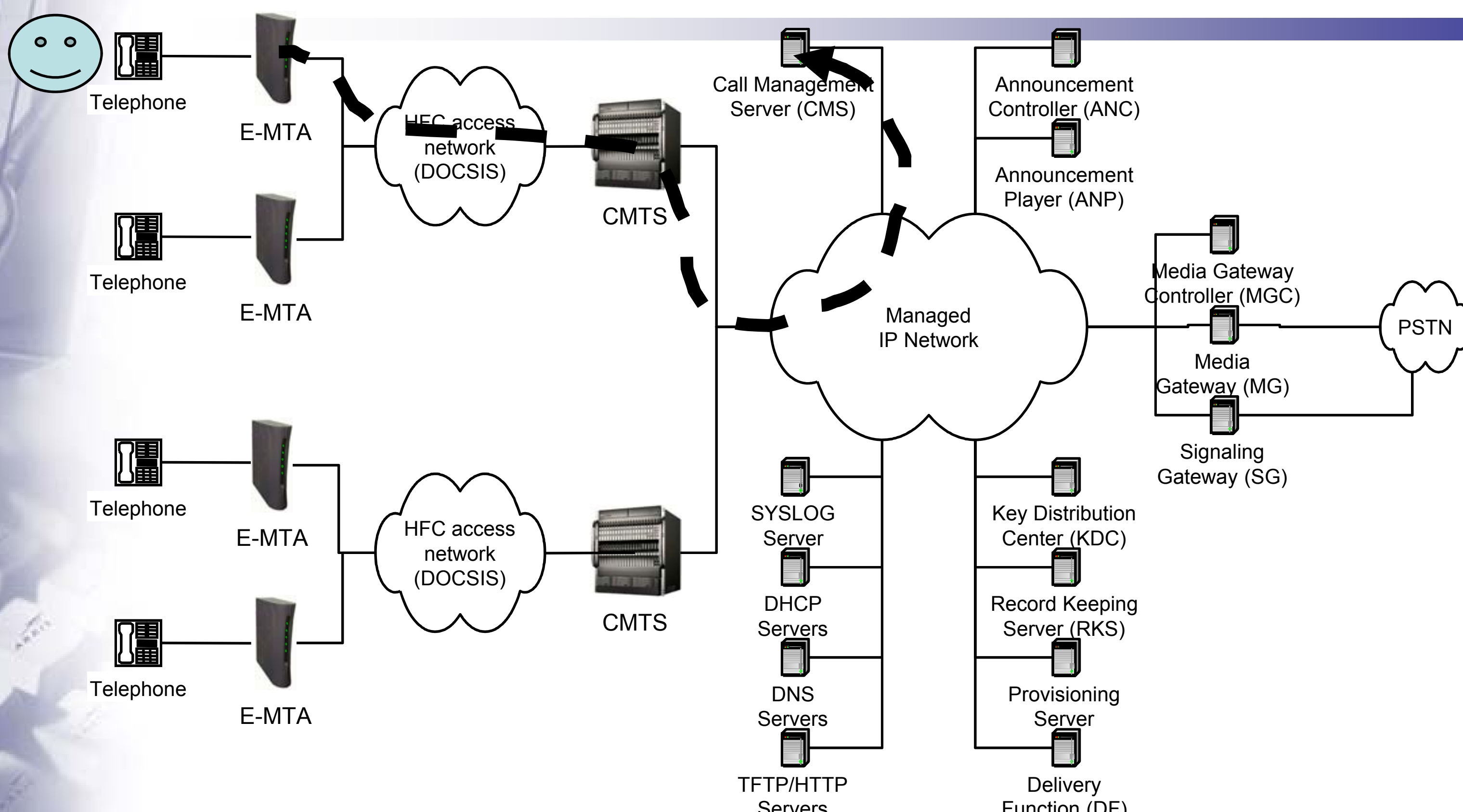
"Typical" PacketCable 1.x Call Flow



CMS creates an inactive connection on MTA_O. CMS tells MTA_O to look for onhook, play dialtone and collect digits according to digit map.

“Typical” PacketCable 1.x Call Flow



MTA_O collects digits per matched a pattern, sends digits to CMS.

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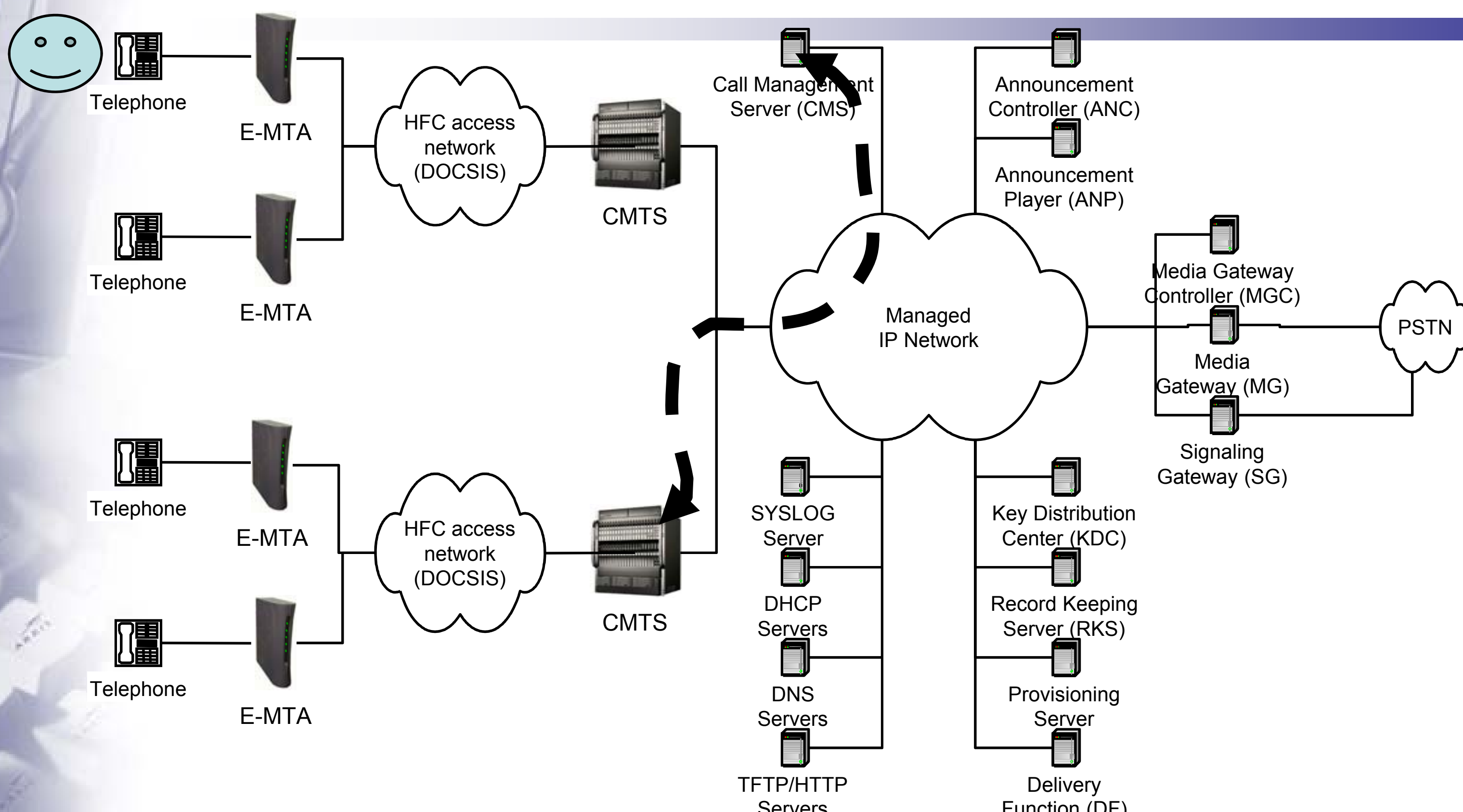
"Typical" PacketCable 1.x Call Flow



CMS retrieves called party information and sends policy object (called a Gate) to $CMTS_T$ and gets Gate-ID_T in return.

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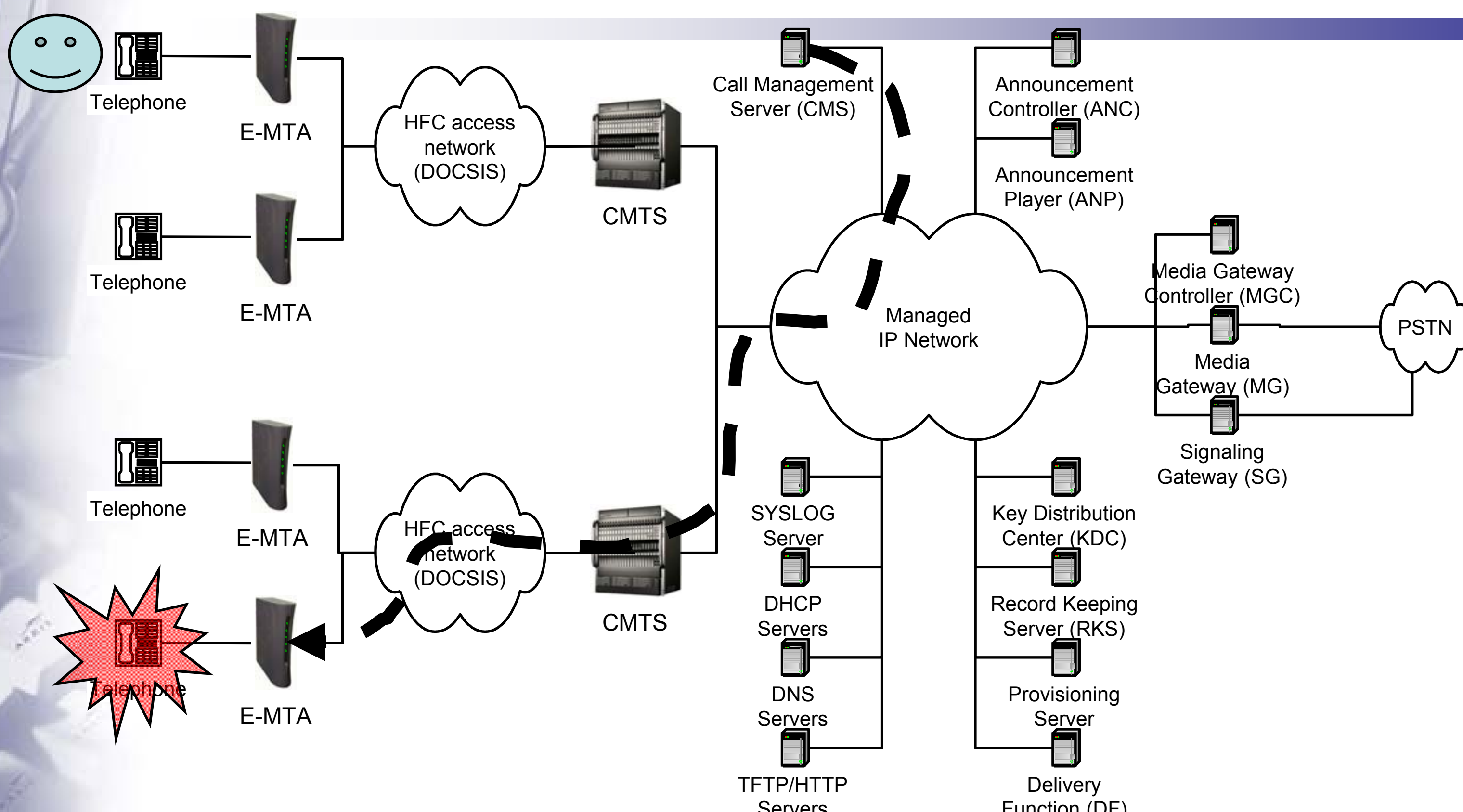
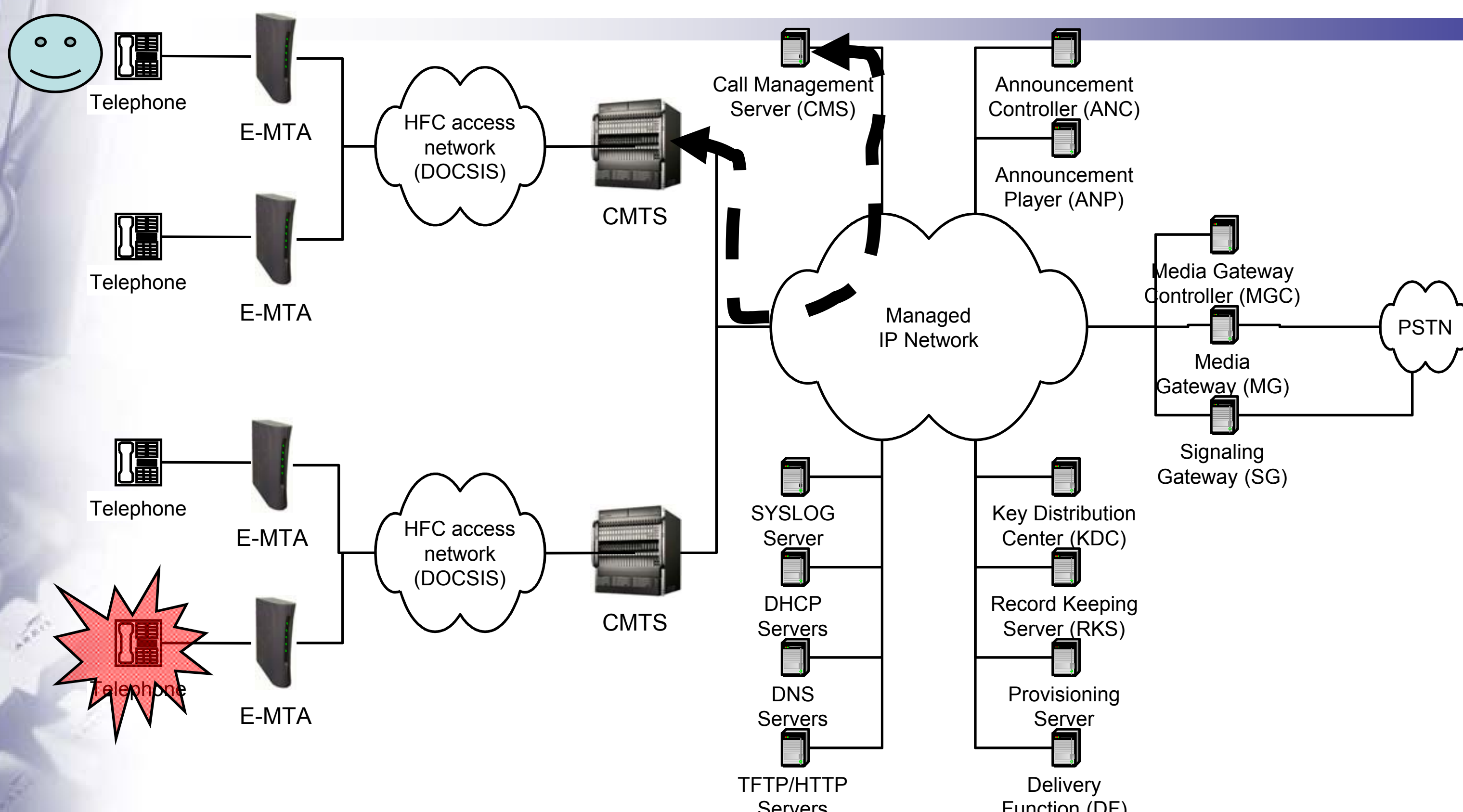
“Typical” PacketCable 1.x Call Flow



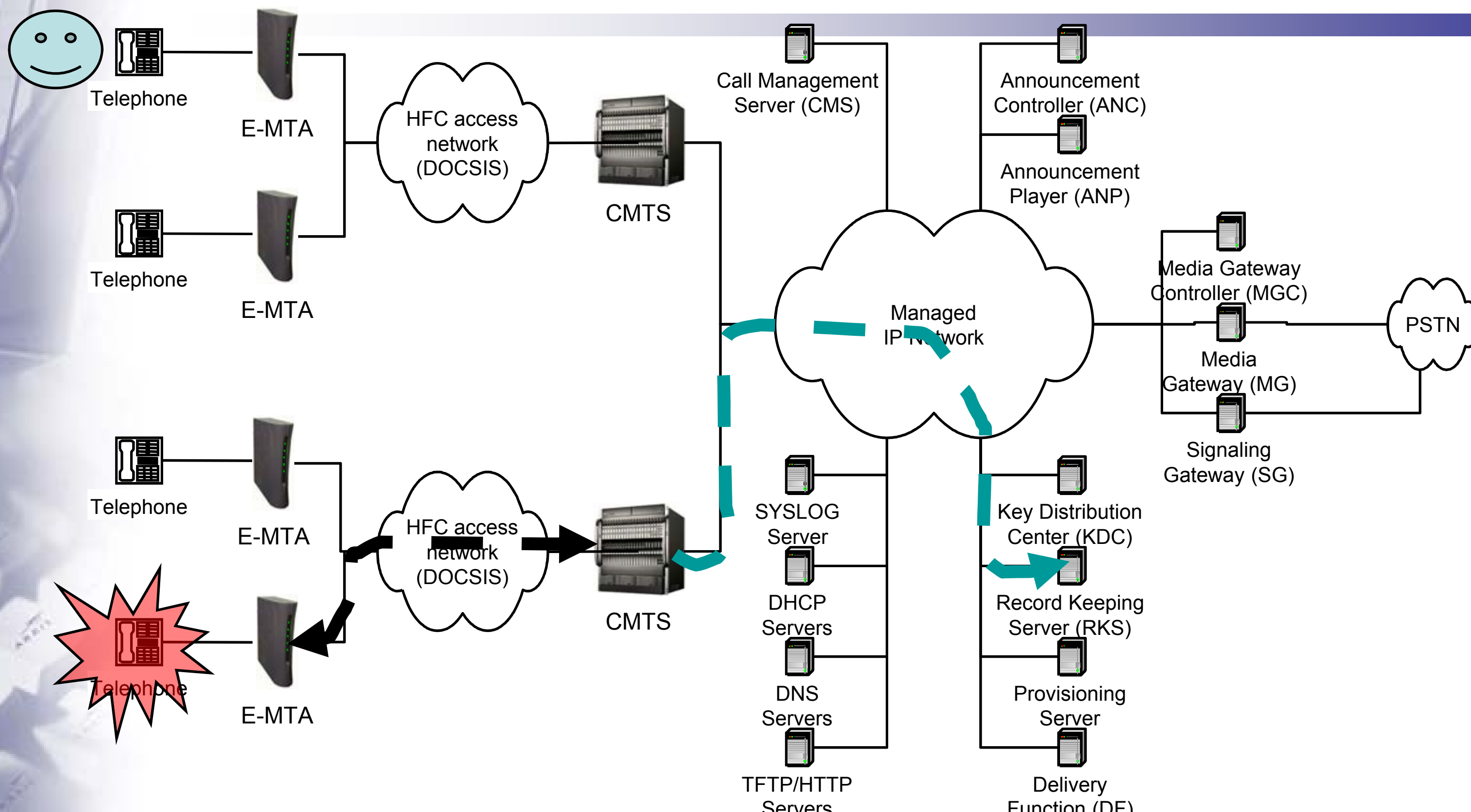
CMS creates an inactive connection on MTA_T and passes the Gate-ID_T that it received in the previous step. CMS instructs MTA_T to ring the phone and asks for notification when the call is answered.

"Typical" PacketCable 1.x Call Flow



CMS sends Gate object to CMTS$_O$ and gets Gate-ID$_O$ in return.

"Typical" PacketCable 1.x Call Flow



Meanwhile, MTA_T uses a DOCSIS Dynamic Services transaction to create admitted upstream and downstream service flows using Gate-ID_T. $CMTS_T$ sends upstream and downstream QoS-Reserve event messages to the RKS.

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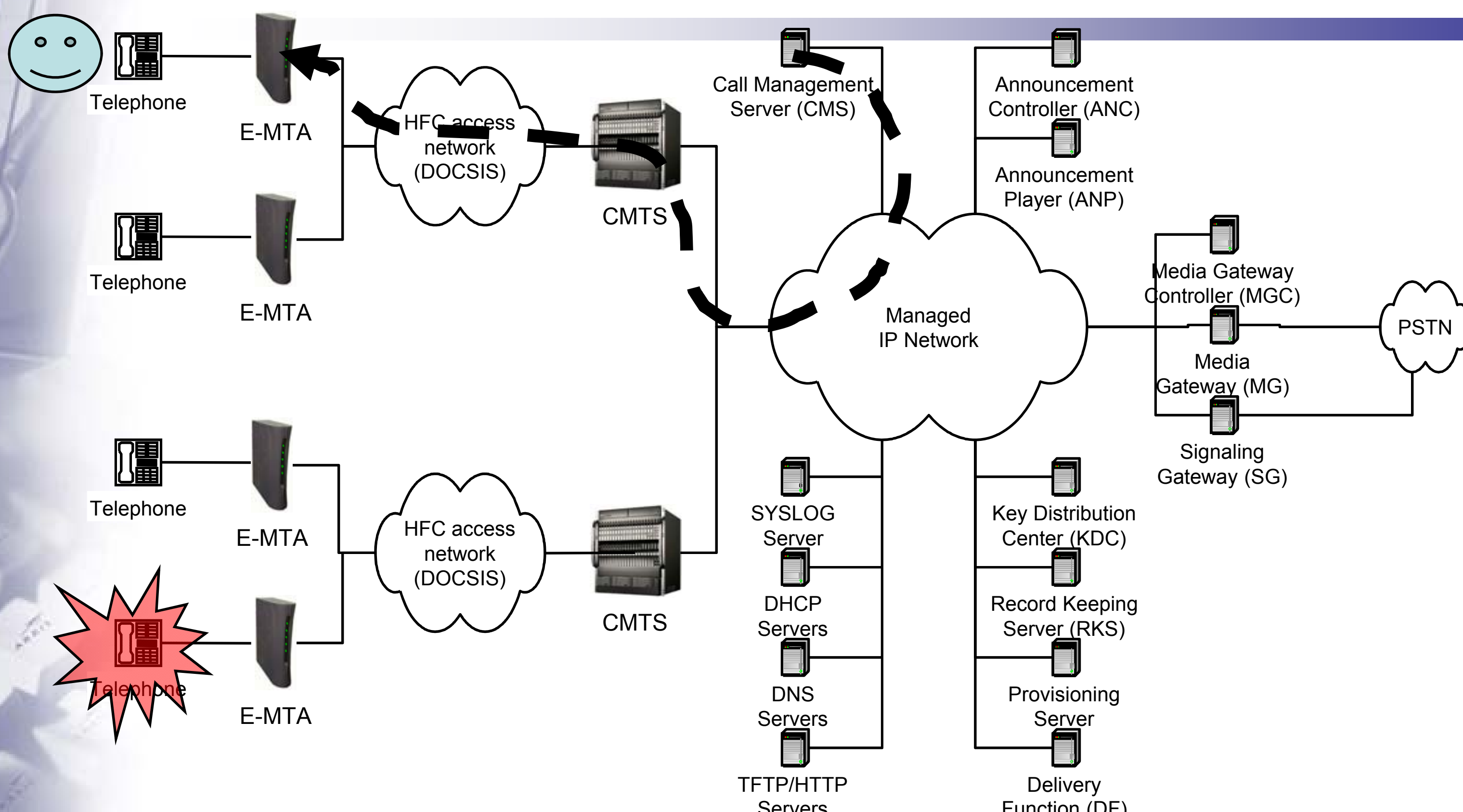
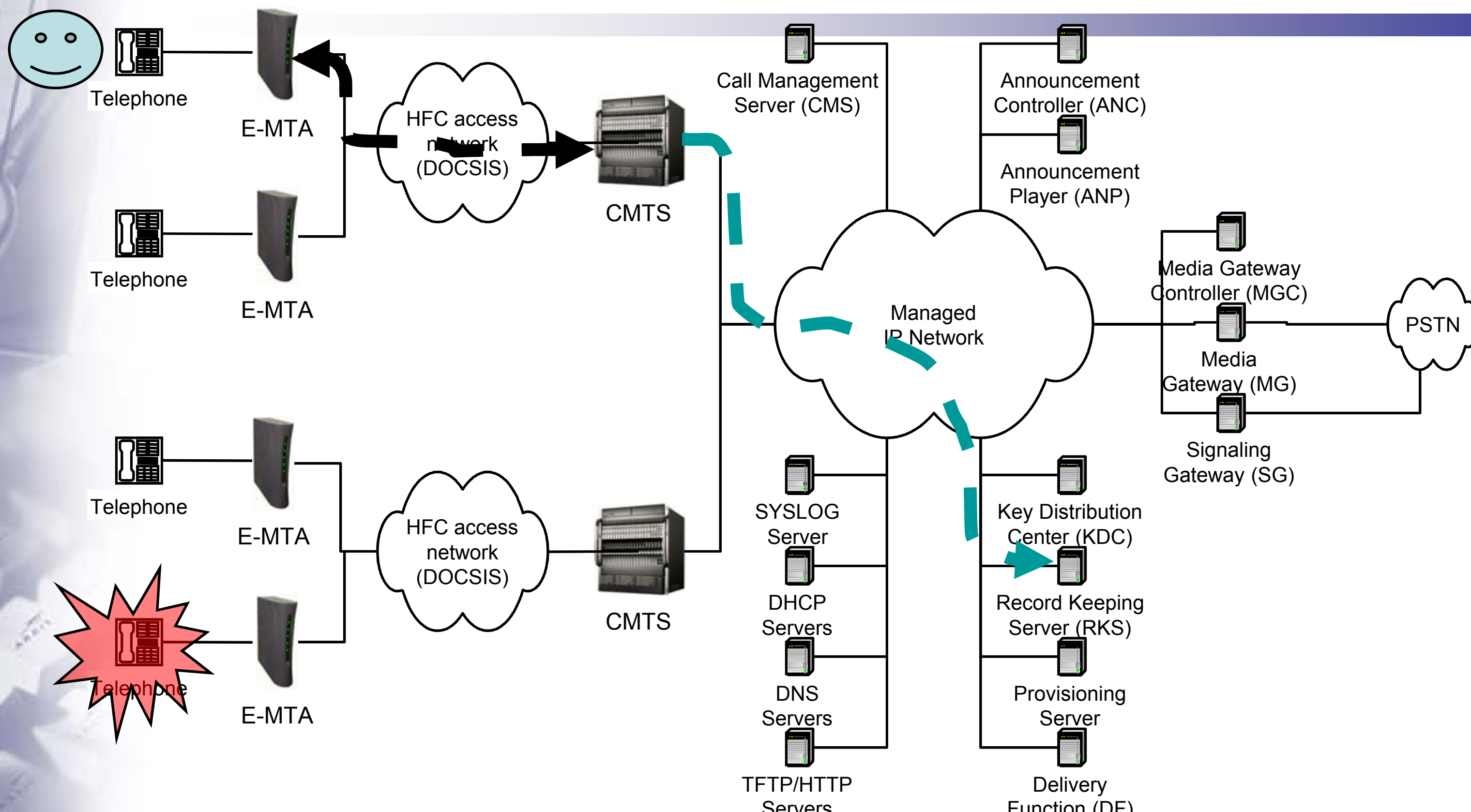
"Typical" PacketCable 1.x Call Flow



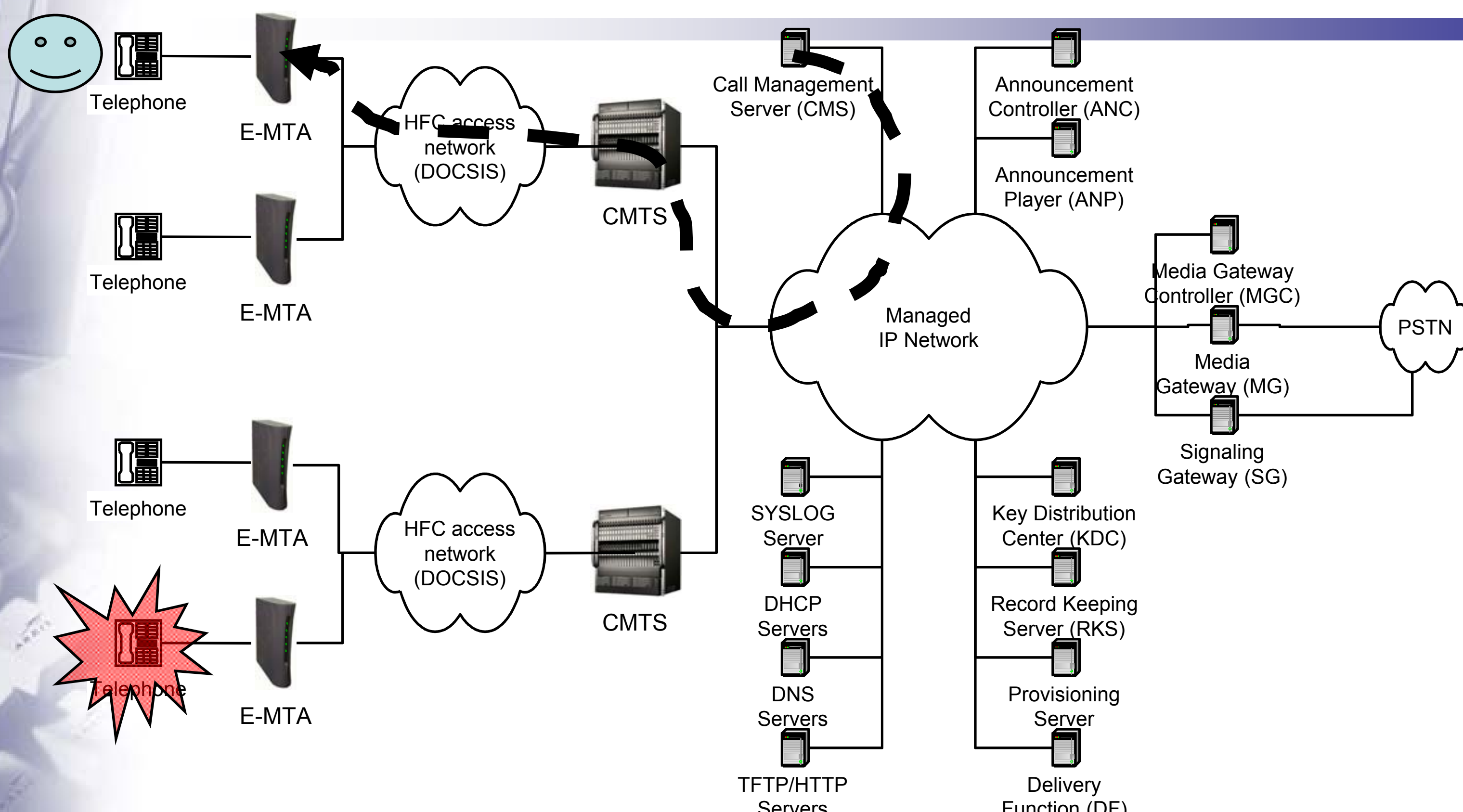
CMS instructs MTA_O to modify its connection to receive-only mode.

“Typical” PacketCable 1.x Call Flow



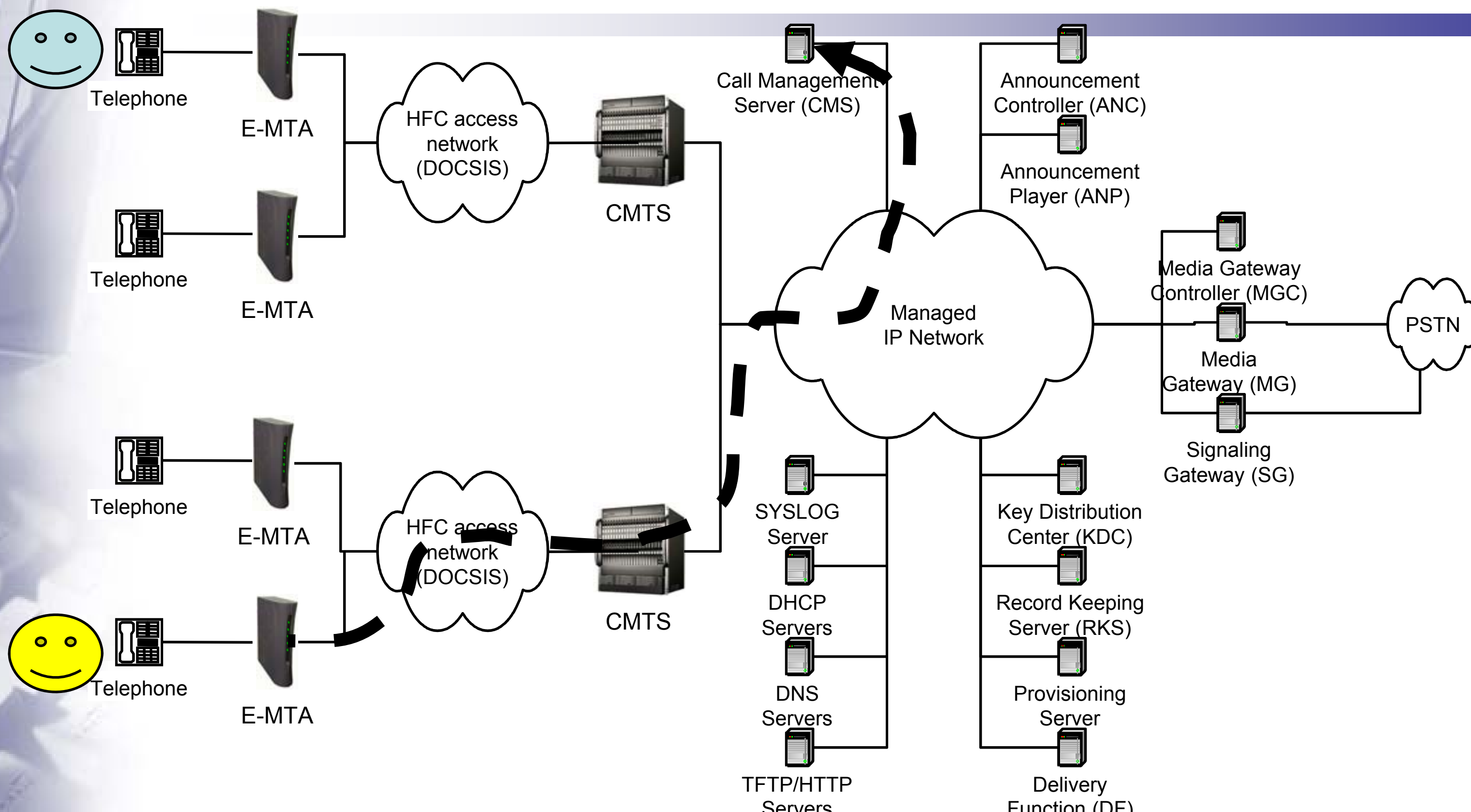
MTA_O uses a DOCSIS Dynamic Services transaction to create an admitted upstream flow and an admitted and activated downstream service flow using $Gate\text{-}ID_O$. $CMTS_O$ sends an upstream QoS-Reserve and downstream QoS-Commit event messages to the RKS.

“Typical” PacketCable 1.x Call Flow



CMS instructs MTA_O to play ringback tones.
At this point, system is stable with terminating line ringing and ringback playing to originator.

“Typical” PacketCable 1.x Call Flow



Terminating user answers phone. MTA_T stops ringing and tells CMS that user went offhook.

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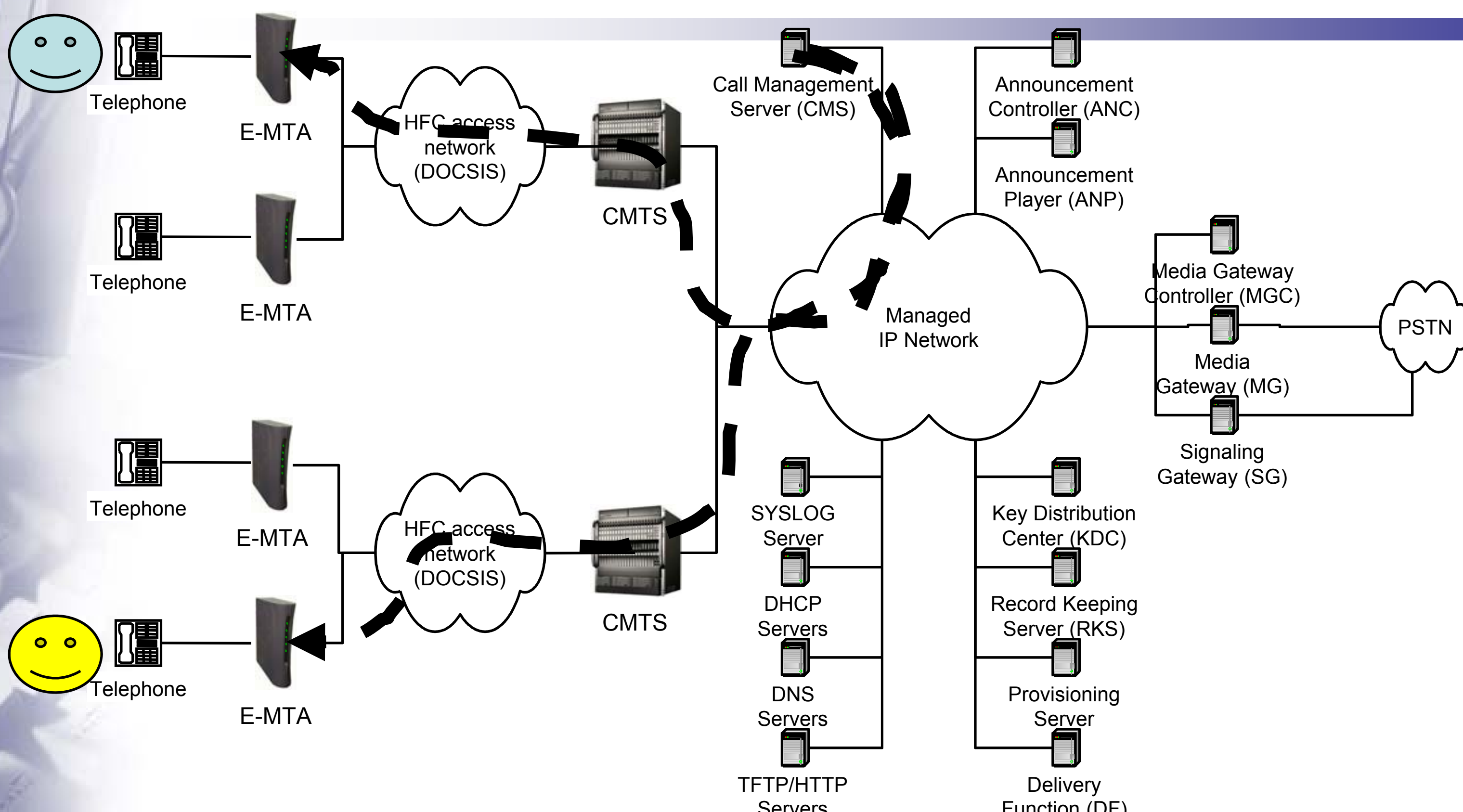
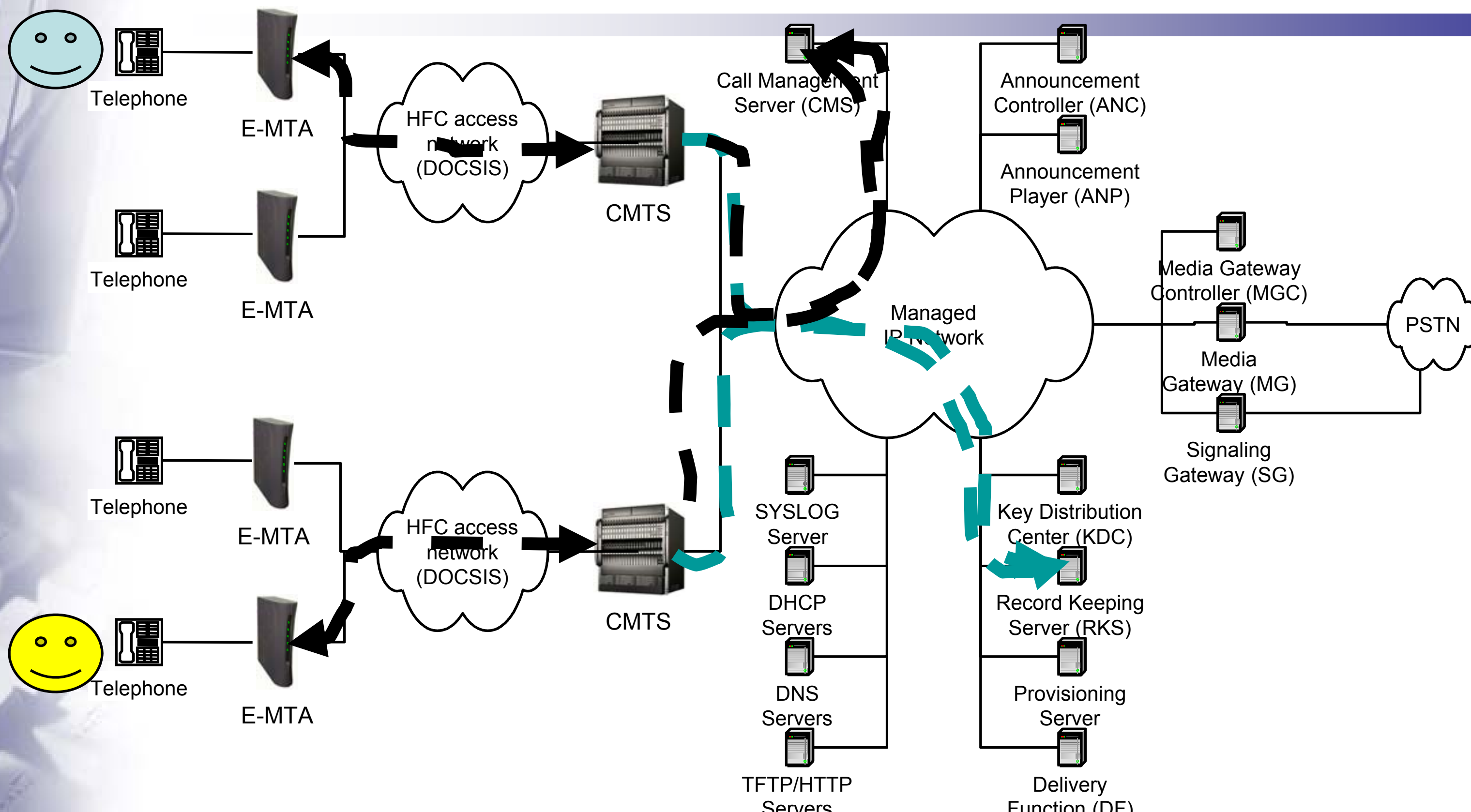
“Typical” PacketCable 1.x Call Flow



Nearly simultaneously, the CMS instructs both MTA_o and MTA_T to change their connections to send-receive mode. CMS tells MTA_T to look for onhook.

“Typical” PacketCable 1.x Call Flow



Each MTA uses a DOCSIS Dynamic Services transaction to change their upstream and downstream service flows to active. The CMTSs send a QoS-Commit event messages to the RKS for the flows that just went active. The CMTSs send Gate-Open notifications to the CMS for the newly committed upstream Gates.

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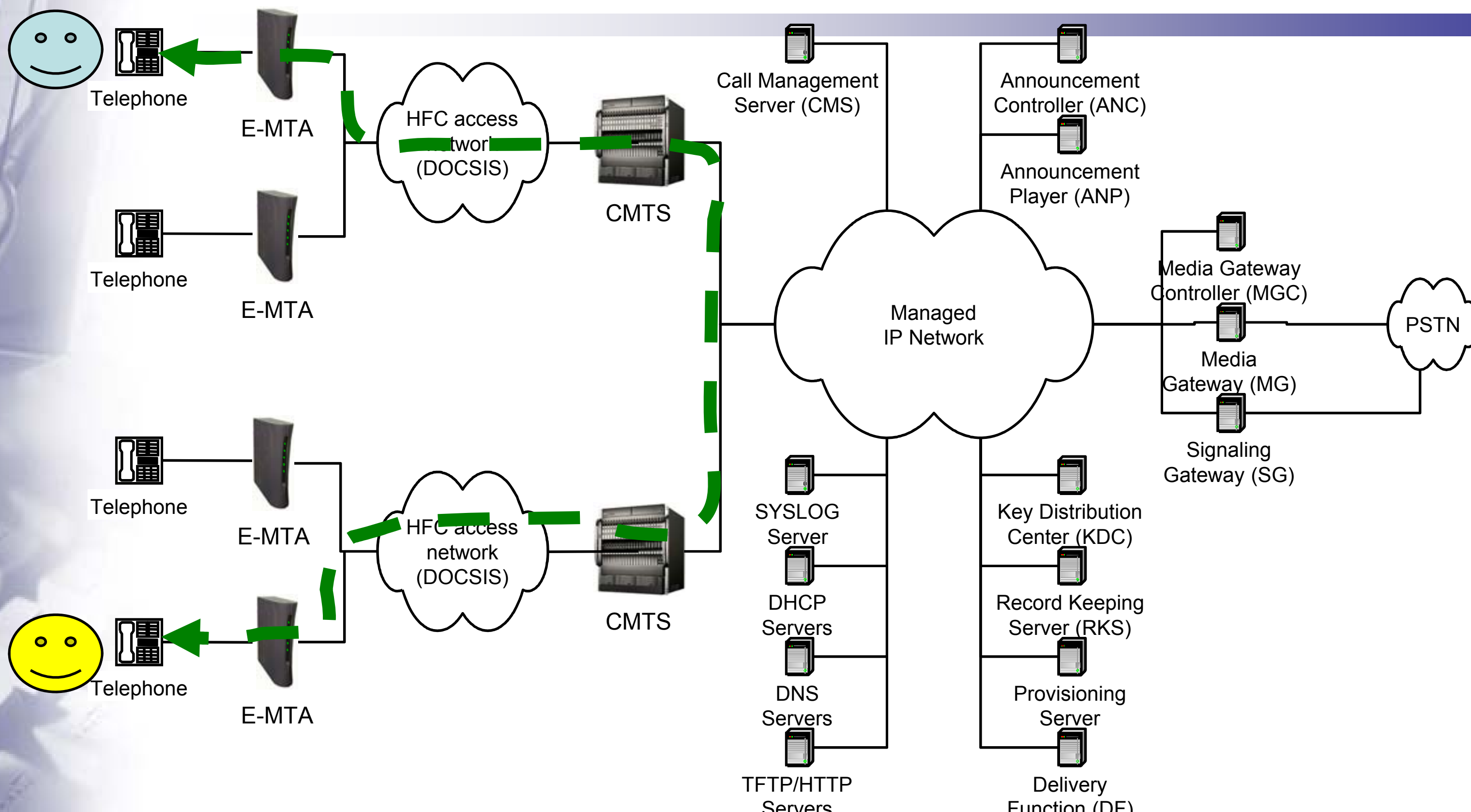
“Typical” PacketCable 1.x Call Flow



Audio passes directly between the devices on the active flows. At this point the call is stable.

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Further Information

- *PacketCable Specifications (http://www.packetcable.com/specifications/)*
  - PacketCable Architecture
  - PacketCable DQoS
  - PacketCable NCS
  - PacketCable Event Messaging
  - PacketCable Security
  - PacketCable Electronic Surveillance
  - PacketCable Provisioning
  - PacketCable CODEC

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Why the Need for Basic Troubleshooting Tools?

- Establishment of a single PacketCable VoIP call requires the coordinated interactions (signaling) between many different network elements
- A media packet from a single PacketCable VoIP call passes through many different network elements
- Problems that occur in a PacketCable VoIP call can originate in many different places
- Isolation and identification of the source (root cause) for a particular problem is challenging

- Troubleshooting tools are essential…

Troubleshooting

Typical Ways that VoIP Problems Are Manifested:

- No dialtone
- Too much voice path delay on calls
- Voice breaking up
- The caller hears voice distortion
- Static on the line
- Calls will not go through
- Voice in only one direction

Troubleshooting

No Dialtone

- Verify that the E-MTA is registered with the CMTS
- Verify that the CMTS has connectivity to the CMS
- Verify the CMS has connectivity to the E-MTA via a ping
- Verify that the E-MTA has the correct IP address (for its CMS)
- Verify that the subscriber is entered in the CMS database and is active

Troubleshooting

Too Much Delay on Calls

- Users notice delay when the end-to-end delay exceeds 150 ms. At this point, voice quality may be acceptable, but the delay is not. When this situation occurs, the processing capabilities of the E-MTA and/or Media Gateway are overwhelmed and not processing voice traffic efficiently
- Buffering reduces jitter in the network; however, it also adds delay as the traffic is adjusted to a fixed rate

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Troubleshooting

Voice Breaking Up

- Congestion in the IP network causes packet loss
- As multiple packets are lost, the listener will hear jumps in the speaker’s voice
- As more packets are lost the distortion becomes worse and conversation is difficult

Troubleshooting

Voice Distortion

- Often caused by tandem encoding
- With this tandem encoding (also called dual encoding or dual compression), VoIP calls routed to a tandem (toll) office are converted to analog form for processing and then are reconverted to digital form for further transmission
- Converting and reconverting more than twice can damage the signal integrity and cause voice distortion

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Troubleshooting

Static

- May be caused by interference with a cordless telephone (operating at a frequency of 2.4 GigaHertz)
- Latency
- Jitter caused by congestion in the network

Troubleshooting

Calls Will Not Go Through

- Problems with the CMS - during busy times it cannot handle another call setup.
- PSTN-bound calls may bombard the MG and/or MGC to the point that they cannot handle any more calls.
- The SG may not be receiving proper messages from the PSTN.
- The call may not be set up properly through the PSTN.
- Congestion in the network
- The signaling information may be dropped by the network.

Troubleshooting

Voice Only in One Direction

- The cause for one-way audio is usually due to an IP routing issue.
- Routing may not be enabled on the routers or there may be a problem with default gateways configured at end stations.

Basic Tools for Troubleshooting (at the Head-end)

Most of the aforementioned problems can be diagnosed using standard sleuthing techniques and a few basic troubleshooting tools:

- Wireshark (Ethereal)
  - Wireshark is an IP network packet sniffer. See http://www.ethereal.com/docs/user-guide/ for users guide.
  - CableLabs has created very useful PacketCable-aware extensions!!!
- Ping and Traceroute
  - Integrity of paths can be checked using these PC-based or CMTS-based applications
- MIB Browser
  - Many PacketCable problems can be diagnosed via MIBs in the CMTS and MIBs in the CM
- CM/MTA Configuration File Generator
  - Every CM or MTA vendor should provide a method to generate the binary configuration files that are downloaded from a TFTP server.
  - Modifications of these binary configuration files are often necessary during troubleshooting efforts

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Advanced VoIP Troubleshooting Tools

- Some advanced EMTAs are capable of performing remote tests on the in-home wiring and providing troubleshooting information

- Two types of remote tests are currently offered:
  - Local Loop Diagnostic tests
  - Transient Voice Quality tests

Loop Diagnostic Testing

Telcordia GR-909

Test	Explanation	Failure Condition
Hazardous Potential Test	Tests for the presence of a foreign AC or DC voltage from tip-to-ground or ring-to-ground	▪ Tip-Ground or Ring-Ground AC voltage is greater than 50 volts RMS ▪ Tip-Ground or Ring-Ground DC voltage is greater than 135 volts
Foreign Electromotive Force (FEMF) Test	Tests for the presence of an AC or DC voltage from tip-to-ground or ring-to-ground	▪ Tip-Ground or Ring-Ground AC voltage is greater than 10 volts RMS * ▪ Tip-Ground or Ring-Ground DC voltage is greater than 10 volts *
Resistive Faults Test	Tests the tip-to-ring, tip-to-ground, and ring-to-ground DC resistance	▪ Tip-Ring, Tip-Ground, or Ring-Ground DC resistances is less than 150 KΩ.
Receiver Off-Hook (ROH) Test	Discriminates between a resistive fault in the loop and a receiver off-hook condition caused by functional terminal equipment.	▪ Detection of a non-linear Tip-Ring DC resistance.
Ringers Test	Tests the AC ringer impedance (or ringer equivalence number, REN) present on the loop due to functional terminal equipment	▪ Measured REN count across Tip-Ring is less than 0.175 REN (no CPE connected condition). ▪ Measured REN count across Tip-Ring is greater than 5 REN (too many CPE connected condition).

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Executing Loop Diagnostics

Loop Diagnostics may be executed during installs or during remote troubleshooting sessions.

There are often three options to executing Loop Diagnostics

- Using SNMP to the CM
- Using a CLI (telnet to the CM)
- Using a Web GUI on the CM

The System Operator can remotely perform these tests, resulting in a simple Pass/Fail result, or can obtain a more substantive test report

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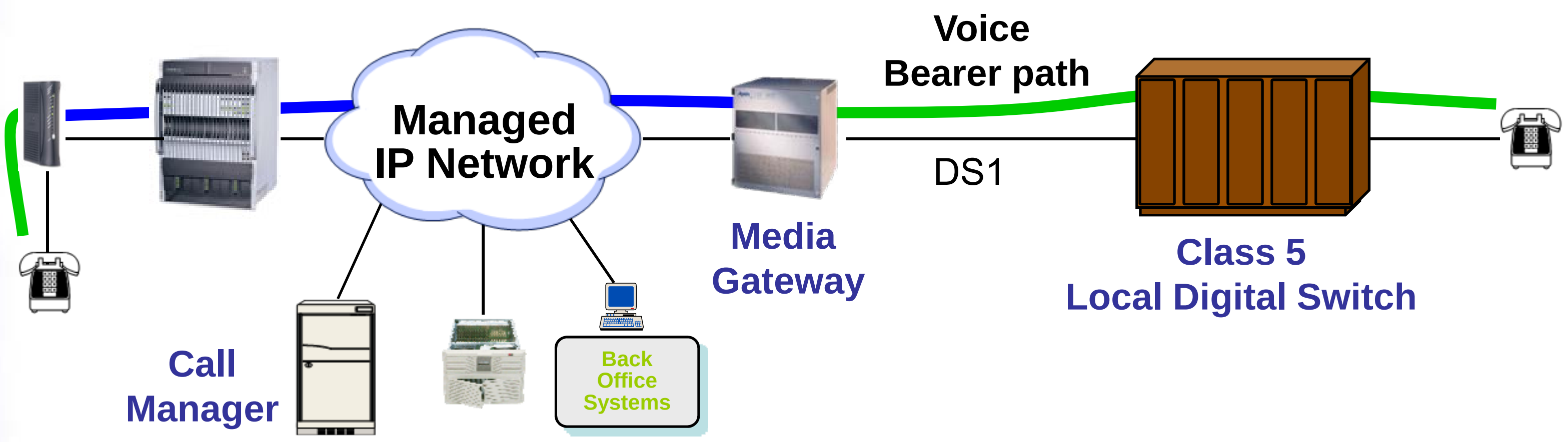
Transient Voice Quality Tests

- Multiple Points of failure in a typical voice call
  - Even a basic VoIP to PSTN call passes through multiple network devices
- Identifying the root cause of basic issues such as noise, 1-way speech path, and echo can require a team of engineers + expensive equipment
- The EMTA is capable of performing advanced speech quality analysis to assist with remote fault segmentation and isolation



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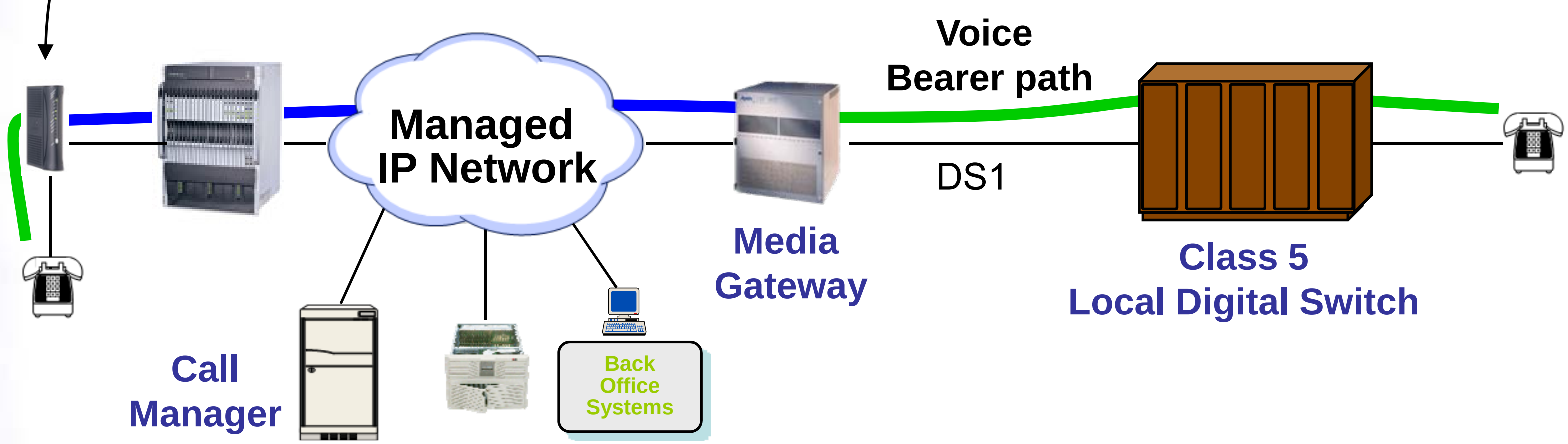
Transient Voice Quality Tests

- **Voice Quality Metrics for last 10 phone calls**
  - Listening Quality and Conversational Quality (MOS) Scores
  - RF Signal and Noise Levels
  - Residual Echo Return Loss scores
  - Packet Loss Analysis
- **IP Address (DHCP) Message Logging**
- **Call Processing Message Logging**



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Conclusions

- Diagnosing & troubleshooting problems in a PacketCable VoIP environment can be challenging

- The advent of new diagnostic tools will help ensure that MSOs can identify problems more rapidly

- This will help ensure higher levels of customer satisfaction for future PacketCable VoIP deployments

Thank you!