Understanding Voice Signaling Protocols

Session 2005
Understanding Telephone Signaling Concepts

Voice Signaling Is the Key to Enhanced Services

Agenda

- Analog/Digital Signaling
- ISDN/SS7 Signaling
- QSIG
- H.323
- SIP
- Media Gateway Controller
In the Beginning—Well Almost!

Switch

Called Station

R—Ring

T—Tip

Telephone Network

Calling Station

Switch

R—Ring

T—Tip

Line Signaling: On-Hook
Line Signaling: Off-Hook

1. Signaling Supervision

Called Station

Telephone Network

Calling Station

R—Ring
T—Tip

Line Signaling: Dial-Tone

2. Call Information (Network to User)

Called Station

Telephone Network

Calling Station

R—Ring
T—Tip
Line Signaling: Addressing

3. Dialing the Destination

 Called Station

 Tones or Pulses

 Called Station

 R—Ring

 T—Tip

Line Signaling: Ringing

4. Call Information (Network to User)

 Ringing

 Ring Back

 R—Ring

 T—Tip
5. Call Completion

300–3400 Hz ≅ 4kHz

R—Ring
T—Tip

Applications of-Line Signaling

Loop Start

FXS

Loop Start

FXS

Loop Start

FXS

Loop Start

PBX

FXO
Trunk Signaling

- Loop start
- Ground start
- E and M

Application—Foreign Exchange (Loop Start)

- Limitations:
  - Simultaneous trunk seizure (glare)
  - Lack of far-end disconnect notification
Ground Start Supervisory Signaling—from PBX

1. PBX Grounds the Ring Lead
   - 48 VDC

2. CO Switch Senses the Ground
   Telephone Switch

3. CO Switch Grounds the Tip Lead

4. PBX Sensing the Ground, Closes the Loop, Removes Ring Ground
   - 48 VDC

Application—Foreign Exchange (Ground Start)

Caller Calls 591-4242 and the Call Appears as a Local Call

LATA 1
591-XXXX

Switch

Ground Start

FXO

LATA 2
555-XXXX

PBX

Ground Start

FXS
Ground Start Benefits and Limitations

- **Benefits**
  - Reduces glare by quickly recognizing incoming seizure (tip ground)
  - Provides remote disconnect supervision

- **Limitations**
  - Mixed voice and signaling
  - Tip and ring cannot be reversed
  - CO Switch and PBX must have same potential ground

Ear and Mouth (E and M) Separate Signaling Path

- Two wire and four wire refer to the voice wires
- The switch listens on the ear (e-lead)
- The switch signals on the mouth (m-lead)
Primary Trunk Signaling Method

- PBX to local exchange office (telephone company)
  - CO trunks
  - Direct Inward Dial (DID) trunks
  - Direct Outward Dial (DOD) trunks
- PBX to PBX connections
  - Tie trunks

E and M Type 1

- 48 VDC
  - Off-Hook
  - On-Hook

PBX

- E-Lead Detector
  - Open
  - Ground
  - Off-Hook
  - 2-Wire, Nonsymmetrical

Line Equipment

Common Ground Must Exist Between PBX and Line Equipment
### E and M Signaling States

<table>
<thead>
<tr>
<th>E and M Type</th>
<th>Condition</th>
<th>M-Lead/SB</th>
<th>E-Lead/SG</th>
</tr>
</thead>
<tbody>
<tr>
<td>Type I</td>
<td>On-Hook</td>
<td>Ground</td>
<td>Open</td>
</tr>
<tr>
<td></td>
<td>Off-Hook</td>
<td>Battery</td>
<td>Ground</td>
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<tr>
<td>Type II</td>
<td>On-Hook</td>
<td>Open</td>
<td>Open</td>
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<td></td>
<td>Off-Hook</td>
<td>Battery</td>
<td>Ground</td>
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<tr>
<td>Type III</td>
<td>On-Hook</td>
<td>Ground</td>
<td>Open</td>
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<td></td>
<td>Off-Hook</td>
<td>Loop Current</td>
<td>Ground</td>
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<tr>
<td>Type IV</td>
<td>On-Hook</td>
<td>Open</td>
<td>Open</td>
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<tr>
<td></td>
<td>Off-Hook</td>
<td>Ground</td>
<td>Ground</td>
</tr>
<tr>
<td>Type V</td>
<td>On-Hook</td>
<td>Open</td>
<td>Open</td>
</tr>
<tr>
<td></td>
<td>Off-Hook</td>
<td>Ground</td>
<td>Ground</td>
</tr>
</tbody>
</table>

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### I’m Ready to Receive the Number Now—Wink Start

- **Sending Switch**
  - On-Hook

- **Receiving Switch**
  - Off-Hook
  - Sending Switch Goes Off-Hook
  - Wink
  - Receiving Switch Goes Momentarily Off-Hook for 140 to 200 msec
  - DTMF Digits
  - Sending Switch Waits a Minimum of 210 msec Before Sending Addressing
  - Receiving Switch Goes Off-Hook after Connection Is Established
I’m Always Ready to Receive—Immediate Start

Sending Switch

On-Hook

Sending Switch Goes Off-Hook

Receiving Switch

Off-Hook

Sending Switch Waits a Minimum of 150 msec Before Sending Addressing

Off-Hook

Receiving Switch Goes Off-Hook after Connection Is Established

On-Hook

E and M Signaling and Wink Start or Immediate Start

Switch

E and M

PBX

E and M

The Only Caveat—Make Sure Your Signaling Type Matches up with Your PBX

2005
1156_05_2000_c1 © 2000, Cisco Systems, Inc.
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Voice Evolution: The Formative Years

Distorted Analog Communications

Clear Digital Communications

300–3400 Hz ≈ 4kHz

2 x 4 kHz = 8 KHz

8 KHz x 8 bits = 64 kbps

Digital Signaling—Better and Cheaper
T1 Signaling Format

Time Slot #X

Channel Associated Signaling (CAS) or Robbed Bit Signaling

Time Slot #24

Extended Super Frame

Usable for ISDN or PBX Signaling

Common Channel Signaling

E1 CAS Signal Format

Time Slot #16

Semi-Multiframe (SMF) 1

Semi-Multiframe (SMF) 2

Channel Associated Signaling

Common Channel Signaling

Usable for ISDN or PBX Signaling
Sample Call

CO Switch

Telephone Network

CO Switch

2 Wire

Loop Start Signaling

T1 Leased Line

T1 Robbed Bit Signaling
Using E and M, A and B Bits = 0, off-Hook

PBX

Voice Evolution: Dawn of the Digital Age

Touch Tone Phones
Faxes
Wow! This Is Better than Fire
The Advanced Intelligent Network

- More efficient
- Support real-time operations of telephone network’s capabilities
- Supports transport transparency
- Customer can create new applications and is provided greater control

Public N-ISDN Intelligent Access to the Network

- BRI—Basic Rate Interface (2B+D)
- PRI—Primary Rate Interface (23B + D or 30 B + D)
Functional Devices and Reference Points

Network Boundary per ITU-T

ISDN Terminal Equipment (TE1)
S
Customer Premises Switching Equipment (NT2)
T
Local Loop Terminator (NT1)
U
ISDN Local Exchange (LE)

Network Boundary in U.S. (No ITU-T Standard)

Non-ISDN Terminal Equipment (TE2)
R
Terminal Adapter (TA)
S/T

Standard Per TA Manufacturer

ISDN Protocol Stack

Application
Presentation
Session
Transport
Network
Data Link
Physical

End-to-End User Signaling
I.451 = Q.931 Basic Call Control
I.452 = Q.932 Supplementary Services

I.45X Call Control
X.25
Further Study

I.465/V.120
LAPD

I.430 Basic Interface + I.431 Primary Interface

Control Signaling
Packet Telemetry
D Channel

Circuit Switch
Semi-Permanent Switched B Channel

Packet Switched
**Q.931/932 Message Format**

- **Call reference**—establishes a unique value between user and the network
- **Message type**—can be grouped into call establishment, call info phase, call clearing and misc.
- **Information elements**—are self contained entities that further define the message

![Message Format Diagram]

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**Supplementary Services**

- **Types:**
  
  Number identification, call offering, call completion, multiparty, community of interest, charging, additional information transfer

- **Control and invocation, 3 generic protocols:**
  
  Keypad—Uses the keypad and display information elements

  Feature key—Uses the feature activation and feature indication Information elements

  Functional protocol—Facility message and facility Information element or specific messages like HOLD
Q.931/Q.932 Call Completion

Calling Party
- Setup
- Setup Acknowledge
- Information
- Call Proceeding
- Alerting
- Connect
- Connect Acknowledge

Network
- Setup
- Call Proceeding
- Alerting
- Connect
- Connect Acknowledge

Called Party

The Intelligent Network

PBX1
Switch
Signaling Network
PBX2
Switch
Telephone Network
SS7 Local Architecture

- Service control point—database information
  - A or Access links—access to STPs
- Signal transfer point—routing of SS7 packets
  - B or Bridge links—connect STPs at the same level
  - C or Cross links—connect mated pairs of STPs, B and C links together make up a quad
  - A or Access links—access to STPs
- Service switching point—access switches or their proxy

SS7 Connection to Remote Network

- D or Diagonal Links for Connecting a Local STP Pair to a Regional STP Pair
- E or Extended Links Used to Connect SSPs to a Secondary Pair of Mated STPs
- F or Fully Extended Links Directly Interconnect Two SS7 Switches
**SS7 Protocol Stack**

- **OMAP**—Oper., Maint. and Admin. Part
- **ASE**—Application Service Element
- **ISUP**—ISDN User Part
- **DUP**—Data User Part
- **MTP**—Message Transfer Part

**Network**

- **Msg Handling, Network Management**
- **Bit Oriented Protocol**
- **Speeds up to 64kbps**

** OSI Reference Model**

- **Presentation, Session, Transport**

**Protocol Usage—Network Elements**

- **TCAP**
- **ISUP**
- **SCCP**
- **MTP**
- **STP**
- **SSP**
- **SCP**
MTP Messages (Level 2 and 3)

BSN/BIB—Backward Sequence Number/Backward Indicator Bit
FSN/FIB—Forward Sequence Number/Forward Indicator Bit
Length—0 = FISU, 1 or 2 = LSSU, 3 or greater = MSU
SIO—Service Information Octet, Type of Protocol and Standard
SIF—Service Information Field, Transfer Control Information
FCS—Frame Check Sequence

Fill-in Signal Unit (FISU)

1-2 Bytes (Always 1)

Link Status Signal Unit (LSSU)

Variable 1 Byte

Message Signal Unit (MSU) Up to 272 Bytes in Length

Message Signal Units (MSUs)

Routing Label

Varies by Protocol/Function

ITU

SLS = Signaling Link Selection Code

• Message discrimination—looks at the routing label, point code and subsystem number to determine if the msg. is local
• Message distribution—uses the SIO to determine the user (application) part
• Message routing—attaches a new routing label and determines the proper link for next signal point
ISDN User Part (ISUP)

- ISUP is used to set up and tear down all circuits used in the PSTN. Telephone User Part (TUP) is used internationally as well.
- Support for non-voice calls and supplementary services with end to end significance.

**Sample Message Types**
- Initial Address Message (IAM)
- Address Complete (ACM)
- Answer (ANM)
- Continuity (COT)

52 message types (ANSI, Bellcore and ITU)
ISUP Call Flow

- TE Setup
- SSP IAM
- STP ACM
- SSP ANM
- SS7 Trunk
- TE Connect
- B Channel
- Disconnect
- TE Release
- SSP Connect
- SS7 Trunk
- SSP Alerting
- TE Call Proceeding
- Q.931

Signaling Connection Control Part (SCCP)

- Larger more complete address space supplements MTP addressing by adding called party and calling party numbers (Subsystem Numbers—SSNs)
- Protocol used for accessing databases and other network entities, i.e. TCAP
- Global title translation
Signaling Connection Control Part (SCCP) Message Format

Routing Label

Message Type
Mandatory Fixed Parts
Mandatory Variable Parts
Optional Parts

Transaction Capabilities Application Part (TCAP)

- General purpose remote operation function for SS7
- Originally designed to support database queries, such as calling cards and 800 numbers
### TCAP Message Format

<table>
<thead>
<tr>
<th>FCS</th>
<th>SIF</th>
<th>SIO</th>
<th>0</th>
<th>0</th>
<th>Length</th>
<th>FSN/FIB</th>
<th>BSN/BIB</th>
<th>FLAG</th>
</tr>
</thead>
<tbody>
<tr>
<td>TCAP</td>
<td>SCCP</td>
<td>xxxx</td>
<td>0011</td>
<td>Routing Label</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

**Transaction Portion (Nature of the Message)**
- Invoke Component
- Parameter Set Sequence

### TCAP Call Flow

- **SS7**
  - Setup
  - Begin
  - Continue
  - Play Announcement and Collect Digits
  - Continue
  - Pass Collected Digits
  - Continue
  - End
  - IAM
  - Actual Network Address

- **Q.931**
  - Call Proceeding
  - Setup
  - Setup Acknowledge
  - Information

- **TP**
  - Begin
  - Continue

- **SCP**
  - Begin
  - Continue
AIN Limitations

- Sporadic deployment
- Poor consistency
  - Lack of interoperability between vendors and between service providers
- Master slave relationship
- Intelligence resides in the network

Voice Evolution:
AIN Services

These 900 Numbers, What Will They Think of Next
Private N-ISDN (QSIG)

- **Purpose:**
  To extend facilities normally only available between extensions on a single PBX to all extensions on PBX’s that are connected together in a private Network

QSIG Benefits

- Multivendor ISDN PBX-based network
- Networking of remote ISDN PBX’s
- Interconnecting voice/fax and DP servers
- Network wide applications
- Support mobility
QSIG Reference Points

PINX—Private Integrated Services Network Exchange

QSIG Protocol

<table>
<thead>
<tr>
<th>Layer 4–7</th>
<th>ROSE: Remote Operation Service Elements</th>
<th>ACSE: Association Control Service Elements</th>
<th>Network Specific</th>
</tr>
</thead>
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<tr>
<td></td>
<td>ISO 11582, ETS300 239 ECMA156, 161, 165</td>
<td>QSIG Basic Call</td>
<td>QSIG Basic Call</td>
</tr>
<tr>
<td>ISO 11574, ETS300 171/172/173, ECMA 106, 142, 143</td>
<td>LAPD, ETS300 402</td>
<td>Interface Dependent Protocols</td>
<td></td>
</tr>
</tbody>
</table>

QSIG Basic Call

Physical

- Basic Rate I.430
- Primary Rate I.431
- Copper
- Copper
- Optical

Link Layer

Physical

Media

Network

End-to-End Protocol

Network Transparent

Supplementary Svcs and ANFs
Basic Call Completion

PINX X
- Setup
- Call Proceeding
- Alerting
- Connect
- Connect Acknowledge

Terminal Node

PINX Y
- Setup
- Call Proceeding
- Alerting
- Connect
- Connect Acknowledge

Transit Node

PINX Z

Terminal Node

QSIG Generic Functional Procedures

End PBX

Application
- SS Control n
  - ...
- SS Control 1
- Serv. Elem
- Coordination func

ROSE—Remote Operation Service Element
ACSE—Association Control Service Element
DSE—Dialog Service Element

Transit PBX

Generic Functional Transport
- Protocol Control
  - DLC
  - PHY

Generic Functional Transport
- Protocol Control
  - DLC
  - PHY

Generic Functional Transport
- Protocol Control
  - DLC
  - PHY
Why Packet-Based Telephony?

- Data networks are growing at a faster rate than voice networks
- One network is cheaper and easier to manage than two
- Leverage the flexibility inherent in data networks for voice
**H.323**

- International (ITU) standard for: *Packet-based multimedia communications systems*
- Version 1 established in 1996, Version 2 in 1998 and Version 3 in the works
- Refer to various annexes for more details
- Leverages previous developments within ITU, i.e. Q.931

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**H.323 Terminal**

- Multimedia communications services over packet-based networks
- Real-time audio, video and/or data communication
- Point to point, multipoint, or broadcast
H.323 Gateways

- Appropriate translation between transmission formats
- Translation between communication procedures
- Call setup and clearing on both sides

H.323 Gatekeepers

Gatekeepers

- H.323 (Over ISDN)
- H.324 (Over POTS)
- Speech Only (Telephone)
H.323 Gatekeeper

- Optional
- Required features
  - Address translation (alias to transport within zone)
  - Admissions control (maybe null)
  - Bandwidth control during the call (maybe null)
- Optional features
  - Call control signaling/routing (under GK control)
  - Call authorization
  - Call management (call status, tracking, PBX-like services etc.)

H.323 Multipoint Control Unit (MCU)
H.323 Terminal—Multipoint Control Unit (MCU)

- An endpoint which provides support for multipoint conferences
- A MCU consists of a multipoint controller (MC) and one or more multipoint processors (MP)
- Endpoints establish a point to point connection with the MC
- Actual video or audio distribution maybe centralized or distributed

Scope of H.323 Recommendation

- Video I/O Equipment
- Audio I/O Equipment
- User Data Applications T.120, etc.
- System Control User Interface
- Video Codec
  - H.261, H.263
- Audio Codec
  - G.711, G.722, G.723, G.728, G.729
- Receive Path Delay (Sync)
- System Control
  - H.245 Control
  - Call Control H.225.0
  - RAS Control H.225.0
- RTP
- RTCP
- UDP
- TCP
- IP
### Call Signaling Procedures—No Gatekeeper

- **H.323 Calling Party**
  - Setup - H.225 Call Signaling Channel
  - Call Proceeding
  - Alerting
  - Connect

- **H.225**
  - Setup message to Called party’s well known call signaling channel identifier

- **H.245 Control Channel**
  - Capabilities exchange and master/slave determination over reliable channel

- **H.245**
  - H.245 Logical Channel setup (RTP)
  - H.245 Logical Channel setup (RTCP)

- **H.245 Logical Channel setup (RTP)**
  - Logical channels setup over unreliable channels for multimedia streams

### H.450 Supplementary Services

- **H.450-1** Generic Functional Protocol
- **H.450-2** Call Transfer Suppl. Serv.
- **H.450-3** Call Diversion Suppl. Serv.
H.323 Limitations

- Relatively slow—due to the extensive amount of message exchange
- Designed with peer to peer multimedia communications in mind
- Protocol format has some limitations

Voice Evolution: End of the Intelligent Network?

I Guess I Better Reboot my Phone
Session Initiation Protocol (SIP)

- Internet telephony not telephony over Internet
- Currently underdevelopment within the IETF (Multiparty Multimedia Session Control Working Group)

SIP Architecture: Contacting the Proxy
SIP Architecture: Proxy Responding to Client

- User Agents
- Client
- Server
- Proxy
- IP-Based Network
- Redirect
- Response OK

SIP Architecture: Client Acknowledgement

- User Agents
- Client
- Server
- Proxy
- IP-Based Network
- Redirect
- Ack

SIP Architecture: Contacting the Redirect Server

SIP Architecture: Contacting the User
SIP Protocol

- SIP addressing takes the form of a mail to URL, i.e. user@host examples sip: squan@cisco.com
- Session Description Protocol (SDP) is used to form the message, analogous to Q.931 messages and info. elements
- Modeled around HTTP, but with UDP

Internet Telephony Protocols

SDP

Media

H.323
RTSP
SIP
RTCP
RTP
RSVP

TCP

UDP

IP
SIP vs. H.323

- SIP uses text for encoding of messages; H.323 uses ASN.1
- SIP uses a single request to send all necessary information
- UDP-based; recent changes by H.323 will allow utilization of UDP as well
- H.323 has widespread usage

Internet Telephony Evolution: Alien Life Form?
Gateway Control Protocols

- Allows remote control of various devices
- Create, modify and delete connections; generates and detect events (tones); tracks resource states
- Fits in well with multimedia call signaling, i.e. H.323 and SIP
- Strong support for existing telephone networks

Gateway Control Migration

- SGCP—Simple Gateway Control Protocol
- IPDC—IP Device Control
- MGCP—Media Gateway Control Protocol
- MDCP—Media Device Control Protocol
- MEGACO—Media Gateway Controller

- SGCP (IETF) July '98 Bellcore Cisco
- IPDC (IETF) Aug '98 Level 3
- MGCP 1.0 (IETF) Oct '98
- MDCP (IETF) Dec '98 Lucent
- MEGACO (MGCP+) (IETF) April '99

Cisco Systems, Inc.
GCP Protocol Relationship

SS7 Network → Signaling Gateway → Media Gateway Controller → Media Gateway → PSTN Network

SS7 Network → Signaling Gateway → Media Gateway Controller → Media Gateway → PSTN Network

- Sigtran
- H.323+, SIP+, ISUP+
- MGCP, H.GCP, Megaco
- Sigtran
- H.323 or SIP

MGCP

- Retains SGCP simplicity
- Uses established standards (SDP)
- Additional SDP functions for other network types
  - IP, ATM
- Uses IPDC features
  - Wildcards
  - Event grouping
  - Control extensions
  - Endpoint audit
  - Connection audit
  - Restart
MGCP Call Agent

Signaling between Gateway and Call Agent

GW1

Call Agent

Notification

IP Cloud

Signaling between Call Agent and Gateway

GW2

MGCP Call Agent

Signaling between Gateway and Call Agent

GW1

Analog Call Using MGCP

Monitor

Calling Session

Dialed Digits

Called Session

RTP Connection

GW1

GW2

Calling Session

Dialed Digits

Called Session
Summary

- Call Signaling has taken an evolutionary path. The underlying core is to provide basic call control. The next step is understanding how new services are added (supplementary services)
Understanding Voice Signaling Protocols

Session 2005

Please Complete Your Evaluation Form

Session 2005