BDNOG 3, Dhaka

Introduction to IP Telephony

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Session Goal

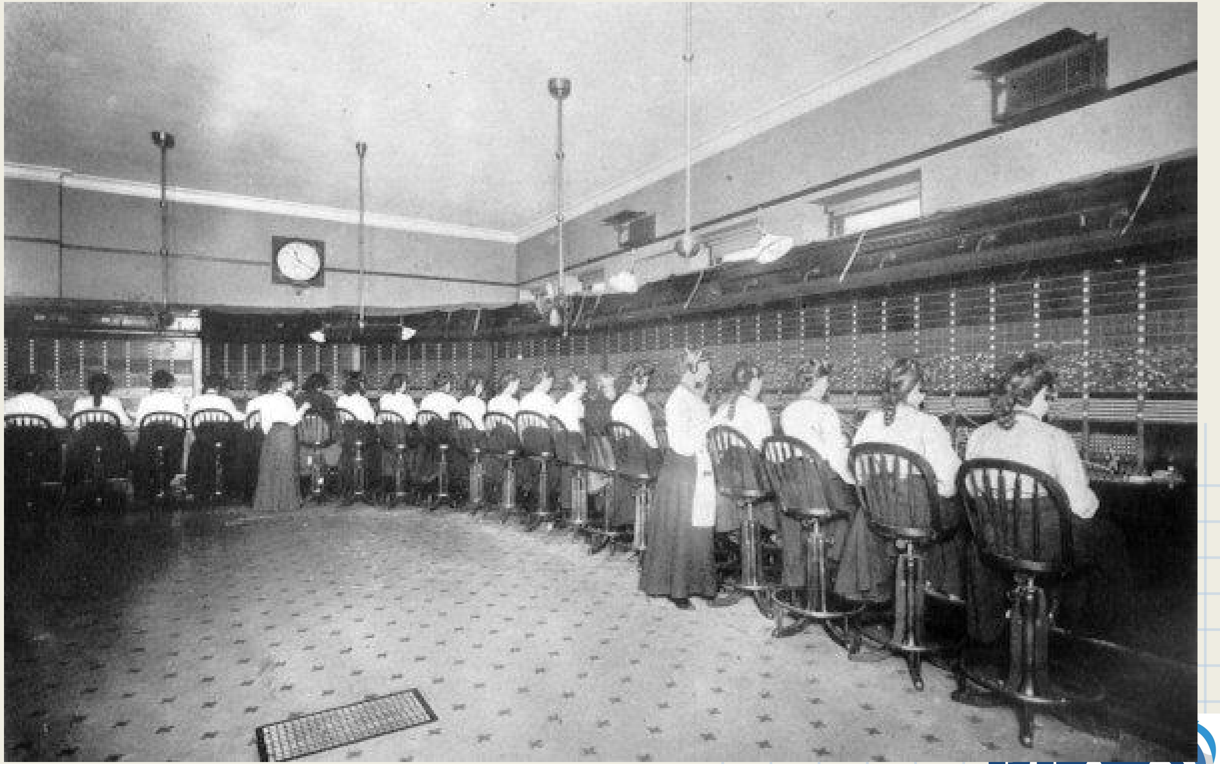
To provide you an basic understanding about IP Telephony, Old days to the future….. ..

Agenda

- Telephony Old Days
- Basic of Telephony
- Introduction to IP Telephony

Telephony The Old Days

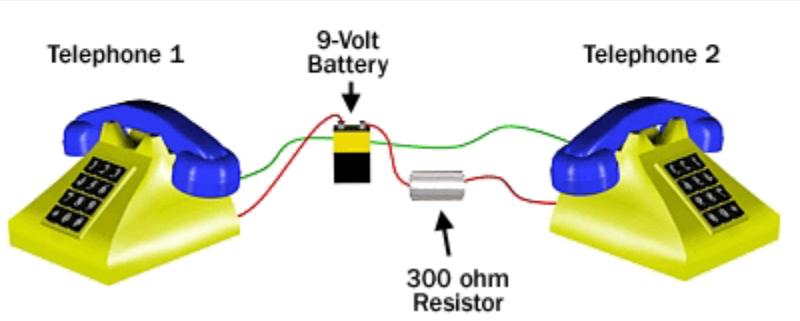
Telephone Exchange in 1910, more than 105 years back.



Basic of Telephony

Existing phone systems are driven by a very reliable but somewhat inefficient method for connecting calls called **circuit switching**.

Circuit switching is a very basic concept that has been used by telephone networks for more than 100 years. When a call is made between two parties, the connection is maintained for the duration of the call. Because you're connecting two points in both directions, the connection is called a circuit. This is the foundation of the Public Switched Telephone Network (PSTN).



Basic of Telephony

Here's how a typical telephone call works:

- You pick up the receiver and listen for a dial tone. This lets you know that you have a connection to the local office of your telephone carrier.
- You dial the number of the party you wish to talk to.
- The call is routed through the switch at your local carrier to the party you are calling.
- A connection is made between your telephone and the other party's line using several interconnected switches along the way.
- The phone at the other end rings, and someone answers the call.
- The connection opens the circuit.
- You talk for a period of time and then hang up the receiver.
- When you hang up, the circuit is closed, freeing your line and all the lines in between.

Basic of Telephony

Let's say you talk for 10 minutes. During this time, the circuit is continuously open between the two phones. In the early phone system, up until 1960 or so, every call had to have a dedicated wire stretching from one end of the call to the other for the duration of the call. So if you were in Kandy and you wanted to call a number in Dhaka, the switches between Dhaka and Kandy would connect pieces of copper wire all the way across the path. You would use all those pieces of wire just for your call for the full 10 minutes. You paid a lot for the call, because you actually owned more than 3000 km long copper wire for 10 minutes.

its a lot of price to be paid…..

Analogue Telephony

- ❑ PSTN allows connection between any two endpoints
- ❑ Human speech typically in the range 250 - 3,000Hz
- ❑ Humans can hear in the region f 20 - 20,000Hz
- ❑ PSTN analogue channel regionally designed t carry 300 - 3,500Hz
- ❑ Most analogue lines delivered via copper from the local exchange (or CO, Central office)
- ❑ Average line ~3Km.

Analogue Telephony

- Even in the day and age of VoIP, this is still important!
  - Analogue telephone adapters (ATAs)
  - Fax - it just won't go away :)
  - Echo
  - Voice and sound is most definitely analogue
    - First and last conversions in a VoIP call

Analogue Telephony

- Analogue telephones connect to a copper pair
  - A two wire circuit
- Analogue telephones are comprised of five major parts:
  - Ringer
  - Dial Pad
  - Hybrid
  - Hook switch
  - Handset

Ringer

- The exchange provides DC (~48vDC) to power the phone
  - Exchange = big centralised UPS
- Exchange provides a burst of AC (~80vAC) to ring the phone's bell
  - Originally a mechanical bell, these days an electronic buzzer
- These days phone have a Ringer Equivalence Number (REN)
  - Exchange can power up to a sum of 5 RENs
  - Phones these days typically < 0.5 REN
  - ATAs have same limitation

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Dial Pad

- ❑ Telephones need to signal back to the exchange
- ❑ Originally done with a rotary dialler making and breaking the copper loop
  - ❑ Pulse Dial, still typically supported by exchanges and some VoIP kit
- ❑ All done with audio tones now
  - ❑ Dual Tone Multi Frequency (DTMF)
  - ❑ Telephone handsets a matrix of switches
  - ❑ One tone per column, one per row
  - ❑ Each switch generates two tones, hence Dual Tone

DTMF Tones

	1209 Hz	1336 Hz	1477 Hz	1633 Hz
697 Hz	1	2	3	A
770 Hz	4	5	6	B
852 Hz	7	8	9	C
941 Hz	*	0	#	D

Hybrid

- The heart of an analogue telephone
- The transformer that couples two signals onto one line
  - Send (Tx) and receive (Rx)
- Creates sidetone ('good echo')
  - Allow speaker to hear himself
- Creates echo unless perfectly balanced

Hook Switch

- Telephone uses it to signal state to the exchange
  - On Hook, closes the copper loop
    - Phone idles, waiting for incoming ring
- Off Hook, breaks the copper loop
  - Requests dial tone from the exchange, and then allows audio to pass
- Also used to signal 'advanced' features, e.g. call waiting
  - Hook Flash - a timed closure of the hook switch, typically ~300ms

Echo

- VoIP does not cause Echo!
  - Hybrids cause echo
  - Echo becomes apparent as latency increases
  - VoIP creates higher latency than circuit switched circuits
- Hybrids must be balanced to the line to effect maximum power transfer and minimal signal reflection
  - Reflection back down the line = echo
  - Reflection back towards the handset = sidetone

Echo

- Sidetone is used to let the user know that the phone 'is working'
  - It's somewhat unnatural to not hear oneself
  - Too much sidetone and you can only hear yourself
  - Too little and it appears the line is dead
- Echo is present on most lines
  - When latency is low (< 20ms or so) the far end perceives it as sidetone

Acoustic Echo

- Caused by the output of the handset's speaker entering the microphone
  - Due to the speaker volume being too loud or microphone sensitivity too loud
    - Very bad with softphones when not using a headset
  - Or flimsy handset construction (acoustic coupling through the handset itself)
  - The telephone handset design hasn't changed much over the years as it is a very good one!
- Indistinguishable to the far end from echo caused by the local hybrid

Reducing Echo

- There are only four ways to reduce echo
  - Remove the two wire (analogue) portion of the call
  - Balance the analogue portion of the call better
    - Hard to do even if you do have access to the endpoint(s)
  - Reduce the latency
    - Often impossible, e.g. long distance calls
  - Cancel the echo

Echo Cancelers

- Measure signal on the line, predict the echo, and create a signal to cancel it
- Echo cancellers are configured for a 'tail' length - the maximum latency of an echo which it can possibly cancel
- Takes time to converge to an echo cancelled state, dependant on the tail length of the canceller
- Echo cancellers aren't perfect, so best to prevent echo in the first place
- Popular misconception that software based echo cancellation is bad.
  - Hardware echo cancellers have very good, often patented algorithms
  - No really good open source software implementations (yet...)
  - Software echo cancellation is not bad - if you have a good algorithm!

Digital Telephony

- Telephony moved digital for the same reason everything else did
- Voice turned to a digital signal using Pulse Code Modulation (PCM)
  - Sample signal in time
- Two important factors:
  - Number of samples per second (highest frequency is half of the sample rate - Nyquist's Theorem)
  - Number of bits used to encode signal
- Tradeoff between quality and bandwidth - standard is 8bits at 8kHz sampling

Digital Telephony

- Standard voice channel (timeslot, or DS0) is 64kbit/s
- Most common codec is G711, a companding codec
  - Two types, ulaw (US) and alaw (Europe)
- Majority of telephone conversation is 'quite' Accurate
- More bits are allocated to quiet signals to improve overall quality

Introduction to IP Telephone

A recent application of Internet technology – Voice over IP (VoIP):
Transmission of voice over Internet

How VoIP works

Continuously sample audio
Convert each sample to digital form
Send digitized stream across Internet in packets
Convert the stream back to analog for playback

Why VoIP
IP telephony is economic; High costs for traditional telephone switching equipments.

Introduction to IP Telephone

Challenge

- Voice transmission delay
- Call setup: call establishment, call termination, etc.
- Backward compatibility with existing PSTN (Public Switched Telephone Network)

IP Telephony Standards:

- ITU (International Telecommunication Union) controls telephony standards.
- IETF (Internet Engineering Task Force) controls TCP/IP standards.

Introduction to IP Telephone

Encoding, Transmission, & Playback

Both groups agree on the basics for encoding and transmission of audio:

Audio is encoded using a well-known standard such as *Pulse Code Modulation* (PCM).

Audio is transferred using the *Real-time Transport Protocol* (RTP).

RTP message is encapsulated in a UDP datagram that is further encapsulated in an IP datagram for transmission.

Introduction to IP Telephone

Encoding, Transmission, & Playback

UDP is used for transport because

- lower overhead: audio must be played as it arrives.
- Playback cannot be stopped to wait for a retransmitted packet.

Two independent RTP sessions exist, because an IP phone call involves transfer in two directions

IP phone acts as sender for outgoing data, and
IP phone acts as receiver for incoming data.

Introduction to IP Telephone

Encoding, Transmission, & Playback

Main complexity of VoIP: Call setup and call management.

The process of establishing and terminating a call is called *Signaling*.

In traditional telephone system, signaling protocol is *SS7 (signaling System 7).*

In VoIP, signaling protocols are:

SIP (Session Initiation Protocol), by IETF
H.323, by ITU
Megaco & MGCP, jointly by IETF and IUT.

VoIP signaling protocols should be able to interact with SS7.

Introduction to IP Telephone

Encoding, Transmission, & Playback

IP telephone system needs to interoperate with PSTN or another IP telephone system.

Two additional components needed for such interconnection:

- Media Gateway
- Signaling Gateway

Introduction to IP Telephone

Journey towards Packet Switching

While circuit switching keeps the connection open and constant, packet switching opens a brief connection -- just long enough to send a small chunk of data, called a packet, from one system to another. It works like this:

- The sending computer chops data into small packets, with an address on each one telling the network devices where to send them.
- Inside of each packet is a payload. The payload is a piece of the e-mail, a music file or whatever type of file is being transmitted inside the packet.
- The sending computer sends the packet to a nearby router and forgets about it. The nearby router send the packet to another router that is closer to the recipient computer. That router sends the packet along to another, even closer router, and so on.
- When the receiving computer finally gets the packets (which may have all taken completely different paths to get there), it uses instructions contained within the packets to reassemble the data into its original state.

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Introduction to IP Telephone

In IP Telephony

- You pick up the receiver, which sends a signal to the ATA.
- The ATA receives the signal and sends a dial tone. This lets you know that you have a connection to the Internet.
- You dial the phone number of the party you wish to talk to. The tones are converted by the ATA into digital data and temporarily stored.
- The phone number data is sent in the form of a request to your VoIP company's call processor. The call processor checks it to ensure that it's in a valid format.
- The call processor determines to whom to map the phone number. In mapping, the phone number is translated to an IP address (more on this later). The soft switch connects the two devices on either end of the call. On the other end, a signal is sent to your friend's ATA, telling it to ask the connected phone to ring.

TBC ….

Introduction to IP Telephone

- Once your friend picks up the phone, a session is established between your computer and your friend's computer. This means that each system knows to expect packets of data from the other system. In the middle, the normal Internet infrastructure handles the call as if it were e-mail or a Web page.
- You talk for a period of time. During the conversation, your system and your friend's system transmit packets back and forth when there is data to be sent. The ATAs at each end translate these packets as they are received and convert them to the analog audio signal that you hear.
- You finish talking and hang up the receiver.
- When you hang up, the circuit is closed between your phone and the ATA.
- The ATA sends a signal to the soft switch connecting the call, terminating the session.

Codecs

Codec	Payload Bitrate
G.711	64 Kbit/s
G.726	16,24 or 32 Kbit/s
G.723.1	5.3 or 6.3 Kbit/s
G.729	8 Kbit/s
GSM	13 Kbit/s
iLBC	13.3 or 15.2 Kbit/s
Speex	2.15 to 22. 4 Kbit/s
Silk	6 to 40 Kbit/s

- G711 gives highest quality
- Some wide bandwidth codecs supported now
- GSM very popular - good CPU time vs. bandwidth tradeoff
- Speex well suited to changing network conditions
- Silk is now supported by asterisk

H.323

- Very old ITU protocol developed to carry multimedia traffic across an IP network
  - Actually a suite of protocols, the signalling component being H.245
  - Originally designed for video conferencing
- Quickly became de-facto standard for VoIP - and is still used today in many large carrier environments
- Relatively secure and bug free due to its maturity
- Does not work well with NAT at all
- Has all but disappeared in end stations over the past few years

MGCP

- IETF standard, RFC 3345 (obsoletes RFC2705
- Still widely deployed
  - Slowly being displaced by SIP
- Being a gateway protocol, has very good gateway features useful for a carrier environment
- Some end phone support for MGCP but never big

Skinny / SCCP

- Cisco Proprietary protocol
  - Originally developed by Selius Systems in the mid 1990's
  - Cisco bought them and entered the telephony market :)
- Cisco CallManager based on Skinny, though finally moving to the more standard SIP
- Called Skinny as phones are 'dumb'.
  - SCCP phone events: button X pressed, turn on lamp X, turn off lamp X

IAX

- Developed by Digium, creators of Asterisk
- Apparently it's pronounced "eeks". I still say "eye-aye-ex"
- Primarily designed to connect Asterisk servers together
- Has unique ability to trunk multiple calls down one dataflow
- Includes some extra signalling
- Uses a single UDP port, so NAT friendly
- Can use plaintext, MD5, or RSA key exchange for authentication
- IAX, although open source, is not a widely adopted standard

SIP

- SIP is the VoIP protocol these days - RFC 3261 (obsoletes RFC 2543)
- Original (simple!) draft created in 1996
- We'll be concentrating on SIP and largely ignoring the rest
- It is worth playing around with IAX if you are going to be using Asterisk
- Largely ignored early on it's life (H.323 was used)
- Largely standard implementations of SIP now
- Not overly NAT friendly, although workarounds exist
- Worthy of a more in-depth look!

SIP Characteristic

- Operates at the application layer.
- Encompasses all aspects of signaling, e.g. location of called party, ringing a phone, accepting a call, and terminating a call.
- Provides services such as call forwarding.
- Relies on multicast for conference calls.
- Allows two sides to negotiate capabilities and choose the media and parameters to be used.
- SIP URI is similar to email address. (with prefix “sip:”) E.g. sip: sujon@somewhere.com

SIP Characteristic

SIP Methods

- **INVITE** = Establishes a session.
- **ACK** = Confirms an INVITE request.
- **BYE** = Ends a session.
- **CANCEL** = Cancels establishing of a session.
- **REGISTER** = Communicates user location (host name, IP).
- **OPTIONS** = Communicates information about the capabilities of the calling and receiving SIP phones.
- **PRACK =** Provisional Acknowledgement.
- **SUBSCRIBE =** Subscribes for Notification from the notifier.
- **NOTIFY =** Notifies the subscriber of a new event.
- **PUBLISH =** Publishes an event to the Server.
- **INFO =** Sends mid session information.
- **REFER =** Asks the recipient to issue call transfer.
- **MESSAGE =** Transports Instant Messages.
- **UPDATE =** Modifies the state of a session.

SIP Characteristic

SIP Response

1xx = Informational responses, such as 180 (ringing).
2xx = Success responses.
3xx = Redirection responses.
4XX = Request failures.
5xx = Server errors.
6xx = Global failures.

SIP Characteristic

Informational
- 100 Trying
- 180 Ringing
- 181 Call forwarded
- 182 Queued
- 183 Session Progress

Success
- 200 OK

Redirection
- 300 Multiple Choices
- 301 Moved Perm.
- 302 Moved Temp.
- 380 Alternative Serv.

Request Failure
- 400 Bad Request
- 401 Unauthorised
- 403 Forbidden
- 404 Not Found
- 405 Bad Method
- 415 Unsupported Content
- 420 Bad Extensions
- 486 Busy Here

SIP Characteristic

Server Failure

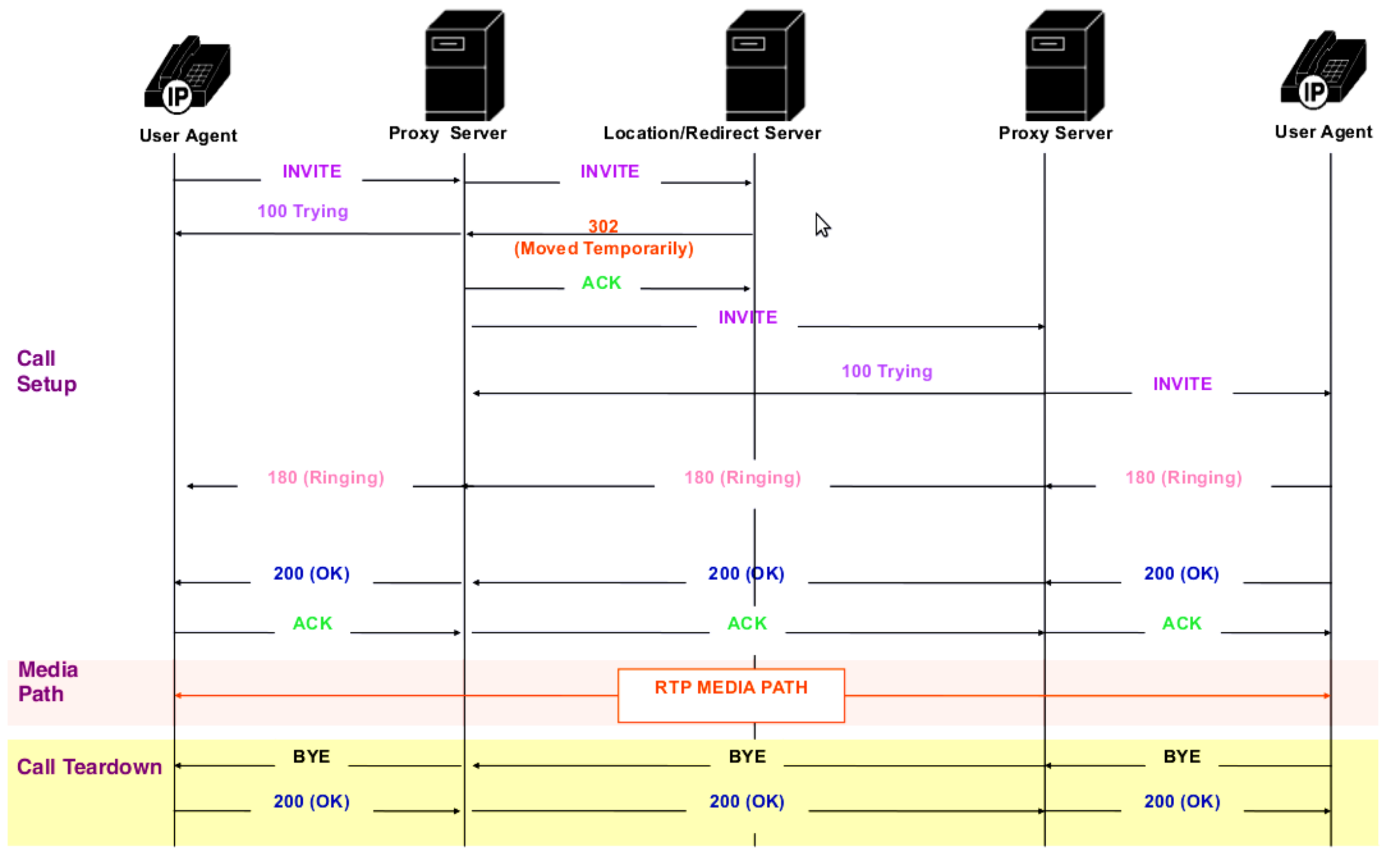
- ❑ 504 Timeout
- ❑ 503 Unavailable
- ❑ 501 Not Implemented
- ❑ 500 Server Error

Global Failure

- ❑ 600 Busy Everywhere
- ❑ 603 Decline
- ❑ 604 Doesn’t Exist
- ❑ 606 Not Acceptable

SIP Characteristic

SIP INVITE



SIP Registration

```
Sip read:
REGISTER sip:203.114.148.130 SIP/2.0
Via: SIP/2.0/UDP 10.71.0.222:5060;rport;branch=z9hG4bK75D24E71C03111DB8A1300112476567E
From: sujon <sip:4989560@203.114.148.130>;tag=1675365723
To: sujon <sip:4989560@203.114.148.130>
Contact: "sujon" <sip:4989560@10.71.0.222:5060>
Call-ID: 7574F569C03111DB8A1300112476567E@203.114.148.130
CSeq: 19613 REGISTER
Expires: 1800
Authorization: Digest
username="4989560",realm="asterisk",nonce="25a752f4",response="
ea87d99f48b43a97b39819e3fedbf8b8",uri="sip:
203.114.148.130"
Max-Forwards: 70
User-Agent: X-Lite release 1105x
Content-Length: 0
```

SIP Registration

```
Transmitting (NAT) to 202.146.237.70:5060:
SIP/2.0 200 OK
Via: SIP/2.0/UDP
10.71.0.222:5060;branch=z9hG4bK75D24E71C03111DB8A1300112476567E;received=202.146.237.70;
rport=5060
From: sujon <sip:4989560@203.114.148.130>;tag=1675365723
To: sujon <sip:4989560@203.114.148.130>;tag=as52d7bb4c
Call-ID: 7574F569C03111DB8A1300112476567E@203.114.148.130
CSeq: 19613 REGISTER
User-Agent: Asterisk PBX
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER
Expires: 1800
Contact: <sip:4989560@10.71.0.222:5060>;expires=1800
Date: Mon, 19 Feb 2007 14:54:56 GMT
Content-Length: 0
```

SIP INVITE

```
Sip read:
INVITE sip:0212304323@203.114.148.130 SIP/2.0
Via: SIP/2.0/UDP 10.71.0.222:5060;rport;branch=z9hG4bKC19D7202C03111DB8A1300112476567E
From: sujon <sip:4989560@203.114.148.130>;tag=1386353914
To: <sip:0212304323@203.114.148.130>
Contact: <sip:4989560@10.71.0.222:5060>
Call-ID: C06D0E06-C031-11DB-8A13-00112476567E@10.71.0.222
CSeq: 32821 INVITE
Max-Forwards: 70
Content-Type: application/sdp
User-Agent: X-Lite release 1105x
Content-Length: 205
v=0
o=4989560 81389423 81389572 IN IP4 10.71.0.222
s=X-Lite
c=IN IP4 10.71.0.222
t=0 0
m=audio 8000 RTP/AVP 3 101
a=rtpmap:3 gsm/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
```

SIP INVITE

```
Reliably Transmitting (NAT):
SIP/2.0 200 OK
Via: SIP/2.0/UDP
10.71.0.222:5060;branch=z9hG4bKC242AB12C03111DB8A1300112476567E;received=202.146.237.70;rport=5060
From: sujon <sip:4989560@203.114.148.130>;tag=1386353914
To: <sip:0212304323@203.114.148.130>;tag=as77d3c840
Call-ID: C06D0E06-C031-11DB-8A13-00112476567E@10.71.0.222
CSeq: 32822 INVITE
User-Agent: Asterisk PBX
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER
Contact: <sip:0212304323@203.114.148.130>
Content-Type: application/sdp
Content-Length: 269
v=0
o=root 26612 26612 IN IP4 203.114.148.130
s=session
c=IN IP4 203.114.148.130
t=0 0
m=audio 19918 RTP/AVP 8 0 3 101
a=rtpmap:8 PCMA/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:3 GSM/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16
a=silenceSupp:off - - - -
```

SIP In Detail

There's much more to SIP than we can possibly hope to cover here