Plaintext transmission over Session Initiation Protocol

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# Table of Contents

- Session Initiation Protocol (SIP)
- Asterisk Telephony Server
- Plaintext Transmission Over SIP
- Experiments
- Conclusion
Introducing SIP Protocol (SIP)
RFC 3261
What Are SIP Roles?

- **SIP**: Generate request
- **UAC**: Generate request
- **UAS**: Generate response
- **SRV**: Proxy server
- **SRV**: Redirect server
- **SRV**: Register server
- **SRV**: Location Server

Request – Response
What Does It Do?

- VoIP
- ToIP
- Call
- OSI
- Media
- SEC
- Traffic

Session Signaling
HTTP like / UTF-8
Transfer /Conference / Hold
Application Layer
TCP / UDP /STCP
Firewall
NAT
What Are SIP Messages?

SIP Requests
- REGISTER
- INVITE
- ACK
- CANCEL
- BYE
- OPTIONS

SIP Responses
- Provisional
- Success
- Redirection
- Client Error
- Server Error
- Global Failure
What Is A SIP PBX?

You   SIP   PBX   SIP   Server   Me
Introducing Asterisk
Asterisk

- Open Source
- SIP Server
- Presence Server
- Application Server
- SIP Proxy
- SIP Gateway
- Voice Messaging
- Instant Messaging
Plaintext Transmission over SIP
How Can SIP Facilitates Telecom Industry?
Leveraging SIP To Drive Telecom Industry?

SIP Extension for ToIP

SIP Extension for IM
SIP Extension for ToIP (RFC 5149)

IETF:
RFC 5149
Experiments:
ToIP-SIP/RTP
SIP:
Set up, control, and tear down
RTP:
Established!
Asterisk:
CMD SendText()

INVITE 1
100 TRYING 3
200 OK 5
ACK 6
RTP 7
MESSAGE 8
200 OK 9
INVITE 2
200 OK 4
ACK 5
RTP 6
MESSAGE 7
200 OK 10
MESSAGE 9
RTP 11
200 OK 12
RTP 11
200 OK 13
SIP Extension for IM (RFC 3428)

IETF:
RFC 3428

Experiments:
IM-SIP

SIP:
Set up, control, and
tear down

RTP:
None!

Asterisk:
CMD MessageSend()
Experiments
Configuration

IP-PBX

Asterisk

Tester

SIPp

Codec

G711 A-LAW

CPS

1000 Calls

SendText()
MessageSend()

<scenario>
</scenario>

ix = 0x0055; {}

Relay()
Message Hits

![Message Hits Diagram](image-url)
Average Memory Usage

![Graph showing average memory usage over call rate (cps)]
Conclusion
• SIP protocol offers different technologies such as VoIP, ToIP, FoIP, IM

**ToIP**
- SIP / RTP
- Reliable

**SIP-IM**
- 100%
- Light
- Agile

- Established RTP session
- Transcoding
- Low performance
- Most reliable solution

- SIP MESSAGE method
- Lightweight
- Maximum Performance
- Most agile solution
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No culture can live if it attempts to be exclusive.

Mahatma Gandhi