

Tutto il VoIP in 45 minuti !!



Networking Competence Provider

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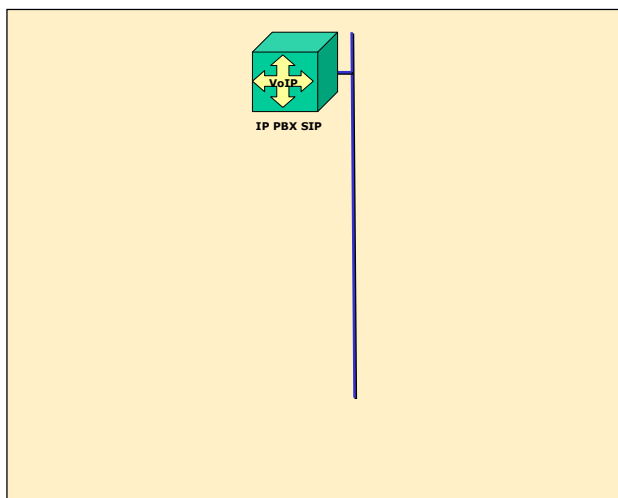


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Ambiente Demo

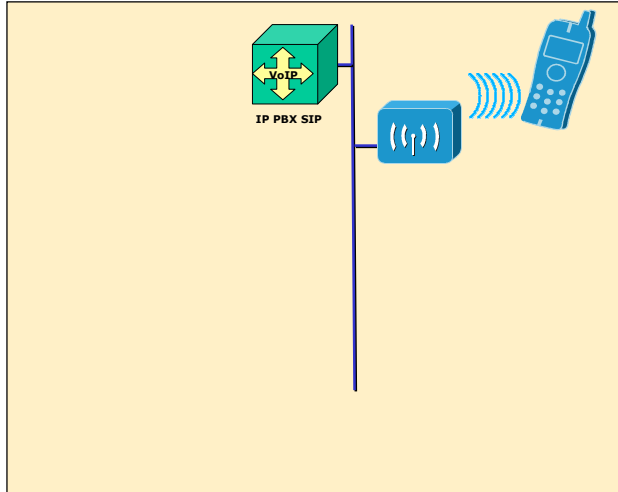


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Ambiente Demo

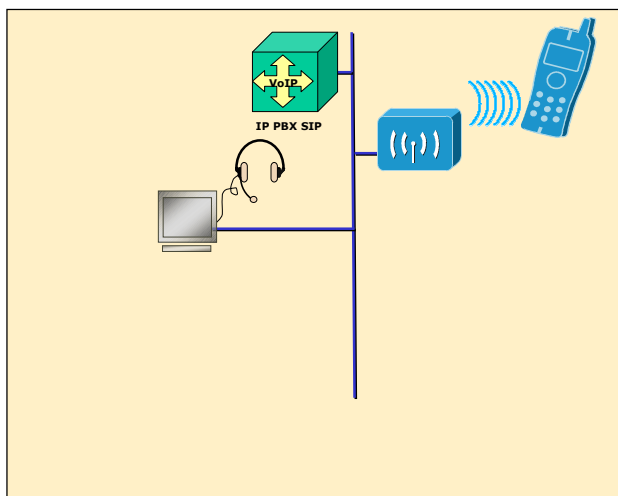


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Ambiente Demo

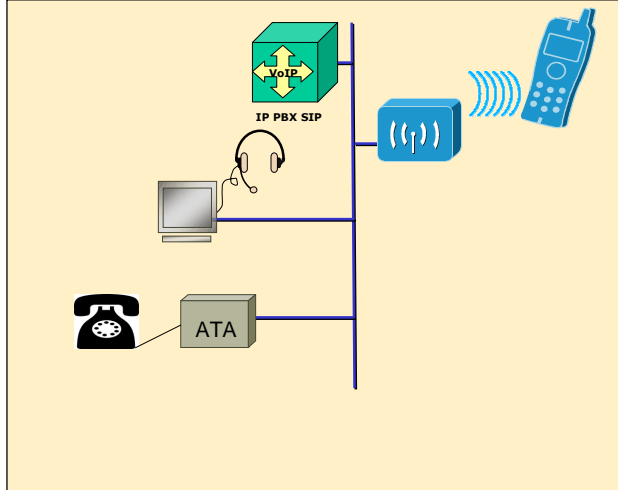


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Ambiente Demo

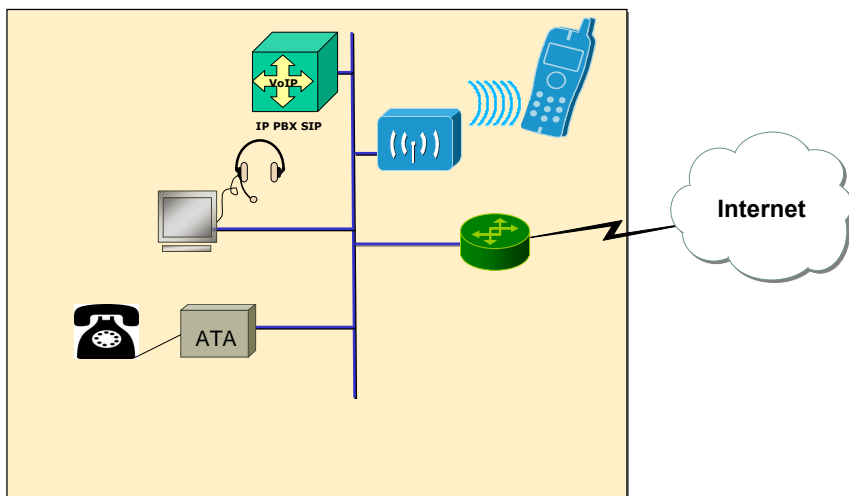


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Ambiente Demo

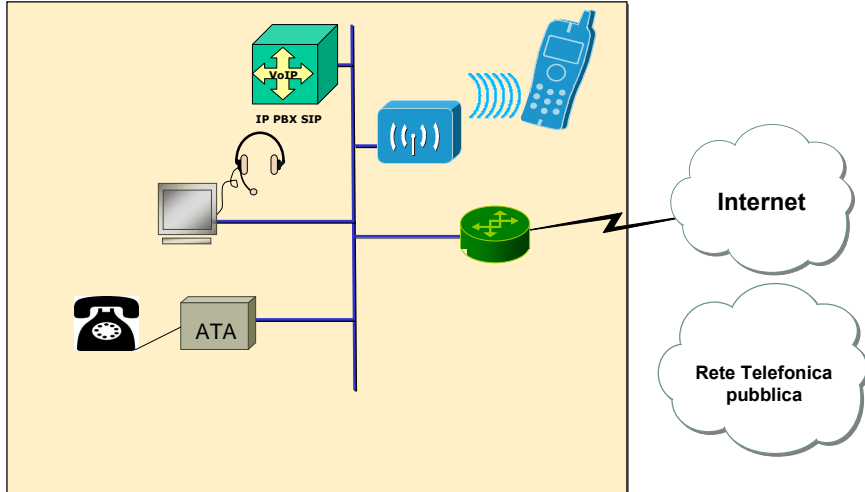


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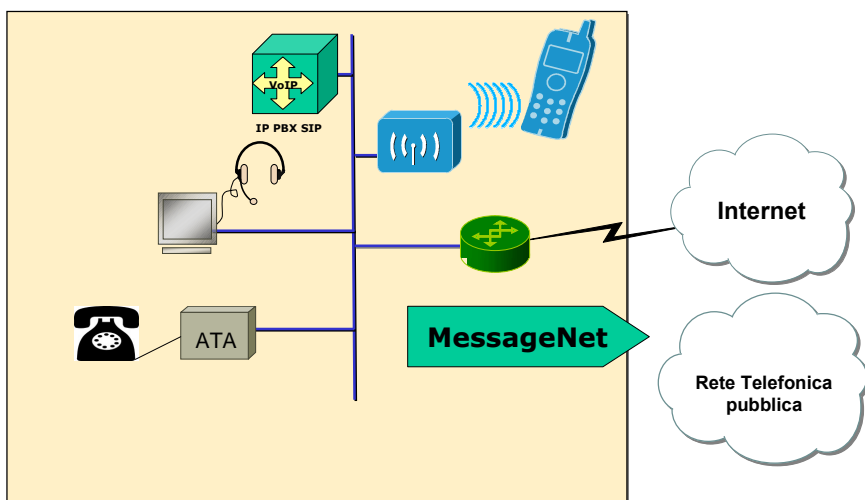
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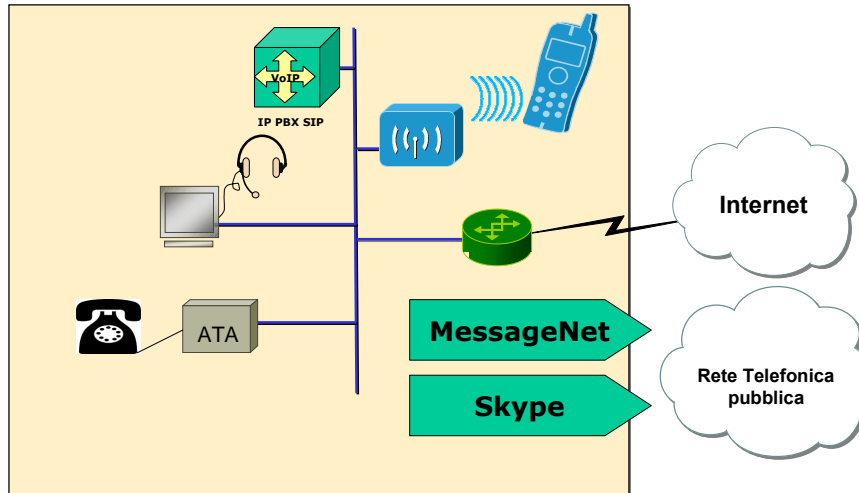
Ambiente Demo



Ambiente Demo



Ambiente Demo



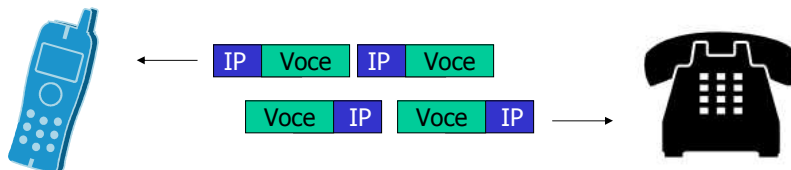
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Obiettivi

- Realizzare un sistema di comunicazione vocale basato interamente su tecnologie VoIP utilizzando:
 - Trasporto Wired e Wireless
 - Telefoni IP
 - Softphone
 - ATA – Analog Telephone Adaptor
 - Protocollo SIP



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Introduzione a SIP

- Session Initiation Protocol (SIP)
 - SIP è un protocollo di livello applicativo che può stabilire, modificare e terminare una sessione multimediale.
 - E' indipendente dal livello di trasporto
 - RFC 3261, 3262, 3903, 3311, 3515, 3428, 3265, 2976 e altri.
 - Basato su un modello transazionale request / response simile al protocollo HTTP

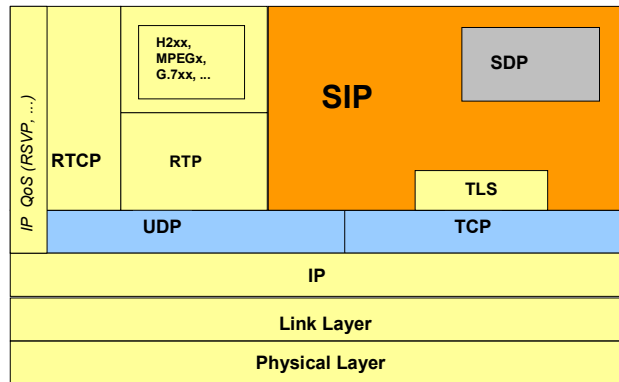


SIP – Componenti e protocolli

- User Agent
 - Risiede in ogni end station SIP: UAC (User Agent Client), UAS (User Agent Server)
- Servers
 - Proxy, Redirect, Registrar, Location.
- URI
 - Formato simile a indirizzi email per identificare utente e dispositivo
 - sip:Alice@atlanta.com
 - sips:proxy-sip.biloxy.com
 - sip:0289678@1.2.3.4
 - sip:1.2.3.4



Architettura protocollare di SIP



Messaggi di Richiesta

- SIP utilizza **sei tipologie** o metodi di Richiesta (*Request*):
 - INVITE—Indicates a user or service is being invited to participate in a call session.
 - ACK—Confirms that the client has received a final response to an INVITE request.
 - BYE—Terminates a call and can be sent by either the caller or the callee.
 - CANCEL—Cancels any pending searches but does not terminate a call that currently in progress.
 - OPTIONS—Queries the capabilities of servers.
 - REGISTER—Registers the address listed in the To header field with a SIP server. Gateways do not support the REGISTER method.



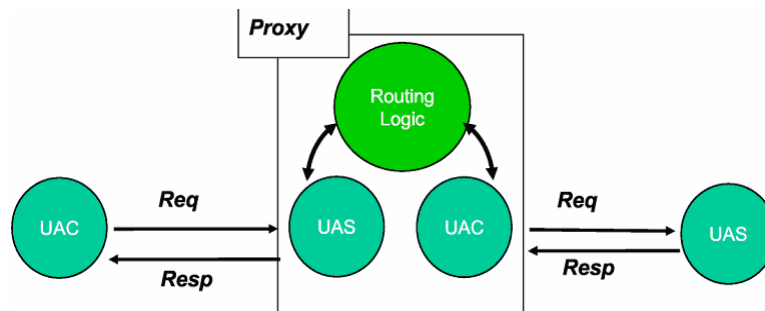
Messaggi di Responso

- Nei messaggi di Responso, SIP utilizza le seguenti categorie:
 - 1xx Informational Messages
 - 2xx Successful Responses
 - 3xx Redirection Responses
 - 4xx Request Failure Responses
 - 5xx Server Failure Responses
 - 6xx General Failure Responses

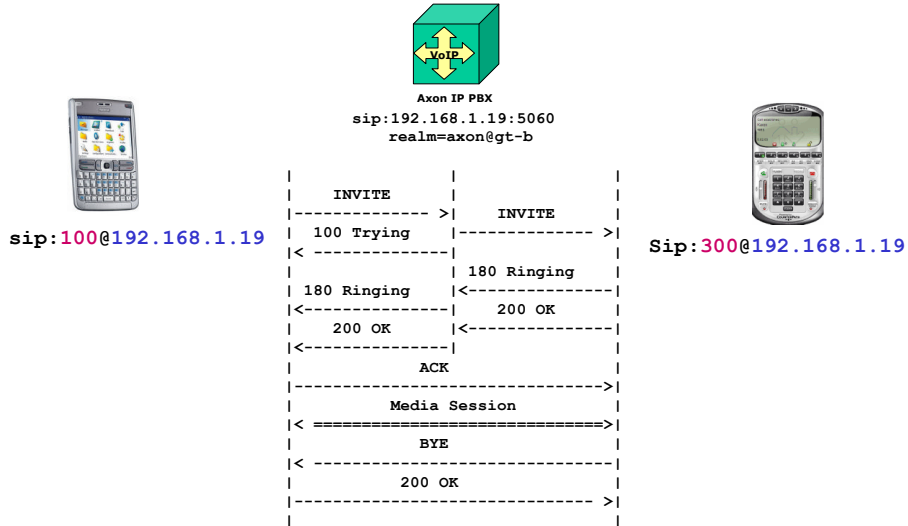


Il proxy Server SIP

- E' un elemento SIP intermedio che agisce sia come UAC sia come UAS, puo' essere: stateless o statefull (Transaction Statefull o Call Statefull)
- La sua principale funzione è di realizzare il *Call Routing*:
 - Instrada le richieste
 - Fa seguire il percorso contrario alle risposte
- Un proxy altera i messaggi SIP in varie parti (nell' URI, nel Via, ecc.)



Nostro obiettivo



Il messaggio INVITE

```
INVITE sip:200@192.168.1.19 SIP/2.0
Via: SIP/2.0/UDP
    192.168.1.100;branch=z9hG4bK776asdhds
Max-Forwards: 70
To: Bob <sip:200@192.168.1.19>
From: Alice <sip:100@192.168.1.19>;tag=1928301774
Call-ID: a84b4c76e66710@192.168.1.100
CSeq: 314159 INVITE
Contact: <sip:100@192.168.1.100>
Content-Type: application/sdp
Content-Lenght: 142
Segue body SDP
```

Dove inviare Response



Dove inviare Request



Il messaggio OK

SIP/2.0 200 OK

Via: SIP/2.0/UDP 192.168.1.19
;branch=z9hG4bKnashds8;received=192.168.1.19

Via: SIP/2.0/UDP 192.168.1.100 ;branch=z9hG4bK776asdhs
;received=192.168.1.100

To: Bob <sip:200@192.168.1.19>;tag=a6c85cf
From: Alice <sip:100@192.168.1.100>;tag=1928301774
Call-ID: a84b4c76e66710@192.168.1.100
CSeq: 314159 INVITE

Contact: <sip:bob@192.168.1.200>
Content-Type: application/sdp
Content-Length: 131

Copriati dal messaggio di INVITE di Alice

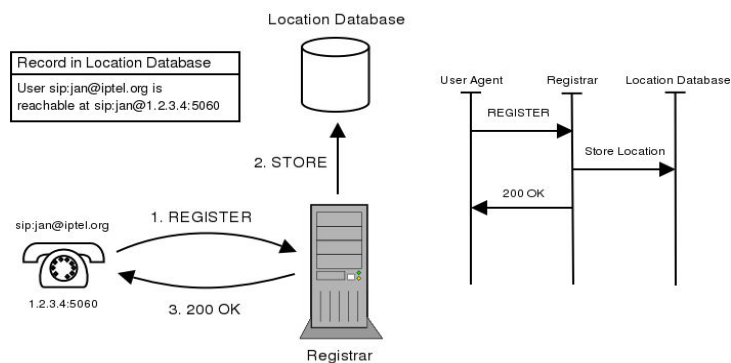


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Registrazione

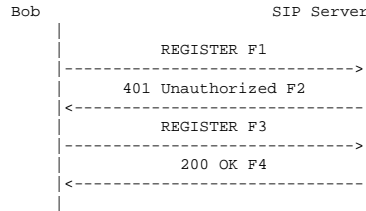


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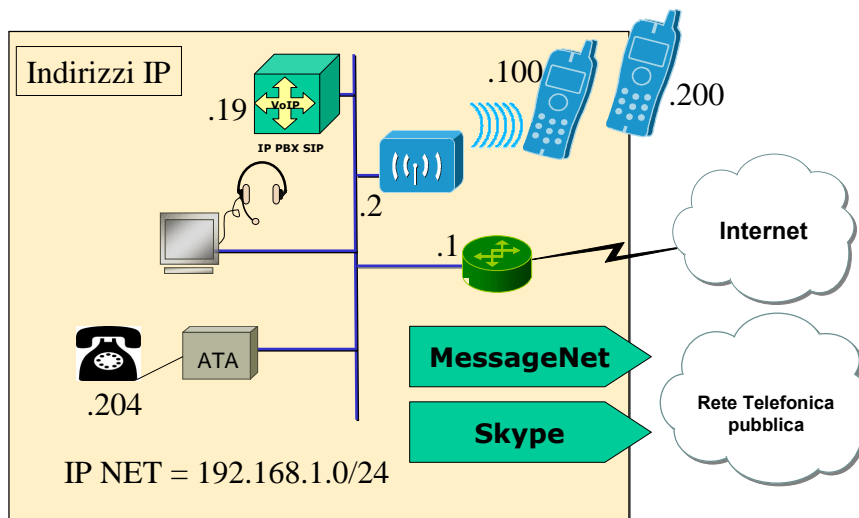
Procedura di Registrazione



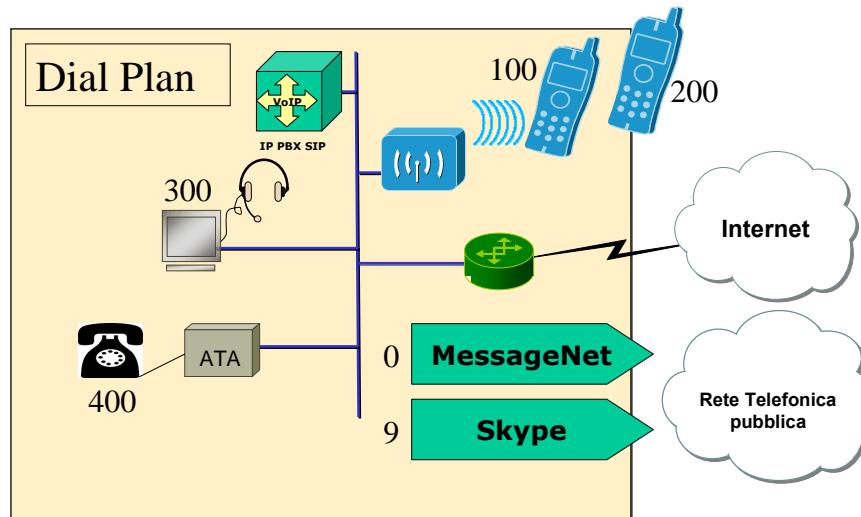
- Bob sends a SIP REGISTER request to the SIP server. The request includes the **user's contact list**. This flow shows the use of HTTP Digest for authentication using TLS transport. TLS transport is used due to the lack of integrity protection in HTTP Digest and the danger of registration hijacking without it, as described in RFC 3261 [1].
- The SIP server provides a challenge to Bob.
- Bob enters his valid user ID and password. Bob's SIP client encrypts the user information according to the challenge issued by the SIP server and sends the response to the SIP server.
- The SIP server validates the user's credentials. It registers the user in its contact database and returns a response (200 OK) to Bob's SIP client. The response includes the user's current contact list in Contact headers. The format of the authentication shown is HTTP digest. It is assumed that Bob has not previously registered with this Server. (estratto RFC 3261)



Ambiente Demo



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Componenti

- Access Point Zyxel
- Softphone X-lite (free)
<http://www.xten.com/index.php?menu=download>
- IP PBX Axon (free)
<http://www.nch.com.au/pbx/index.html>
- Collegamento Internet via UMTS
- UPLink Skype
- Vigor Talk ATA - DrayTek
- VoIP Provider "Messagenet"
<http://www.messagenet.it>
- Cellulari Nokia E61
http://www.nokia.it/link?cid=EDITORIAL_18327



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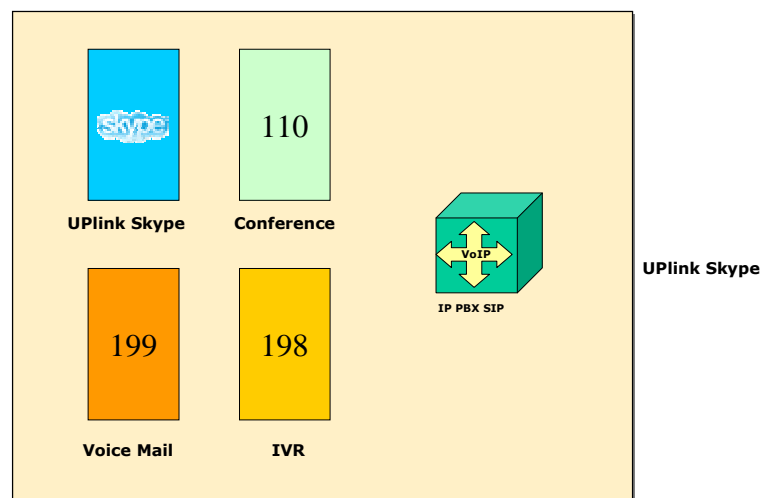


Fasi di configurazione

- Configurazione Access point
- Configurazione IP PBX SIP
- Configurazione Softphone
- Configurazione Nokia E61
- Configurazione ATA
- Configurazione Uplink Skype



Funzionalità



Grazie per la cortese attenzione

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