# MIT's Current SIP Infrastructure

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# Current SIP Implementation

- Utilizes the IETF standards based SIP protocol
- Comprised of several different components
  - SIP Outgoing Proxy
  - SIP Internal Proxy
  - SIP DMZ Proxy
  - Media Proxy
  - Radius server (accounting, authentication and authorization)
  - Asterisk server (voicemail, conferencing, pbx features)
  - Cisco gateway

# SIP Proxies

- Utilizes Sip Express Router (SER) open source software

  http://iptel.org

- Running on a RedHat Linux Enterprise platform

- Our implementation utilizes three functional types of SIP proxy

  - Outgoing proxy - provides authentication and authorization functions, and ensures preferred routing for MIT SIP clients. Redundancy provided through F5 load balancer
  - Internal proxy - provides connectivity to the PSTN gateway and ensures external SIP issues do not impact internal MIT SIP traffic and clients. Redundancy provided through DNS SRV records.
  - DMZ proxy - provides SIP connectivity to external clients outside of MIT, and provides isolation for internal MIT clients to ensure external anomalies do not affect internal MIT clients. Redundancy provided through DNS SRV records.

# Media Proxy

- SIP + NAT = Problems
- NAT, despite the promise of IPv6, continues to be prevalent throughout the Internet and shows little sign of retreat
- Any next generation communications protocol must be able to navigate successfully through NAT devices
- A media proxy enables SIP clients behind a NAT to successfully establish audio calls
- Implemented on a RedHat Linux Enterprise platform using open source software (Its Jeff compliant as it uses Python)
- Scalability and redundancy are provided through the software’s use of SRV records

# Radius Server

- Provides authentication of SIP clients (Usernames/Passwords) to the SIP proxy servers
- Provides authorization for various SIP services (long distance calling, international calling, voicemail settings, call forwarding etc…)
- Maintains accounting records for placed and received SIP calls by MIT clients (call detail records)
- Provides key integration point for adding new features and functionality to the SIP environment
- Implemented on RedHat Linux Enterprise platform using open source software from http://freeradius.org

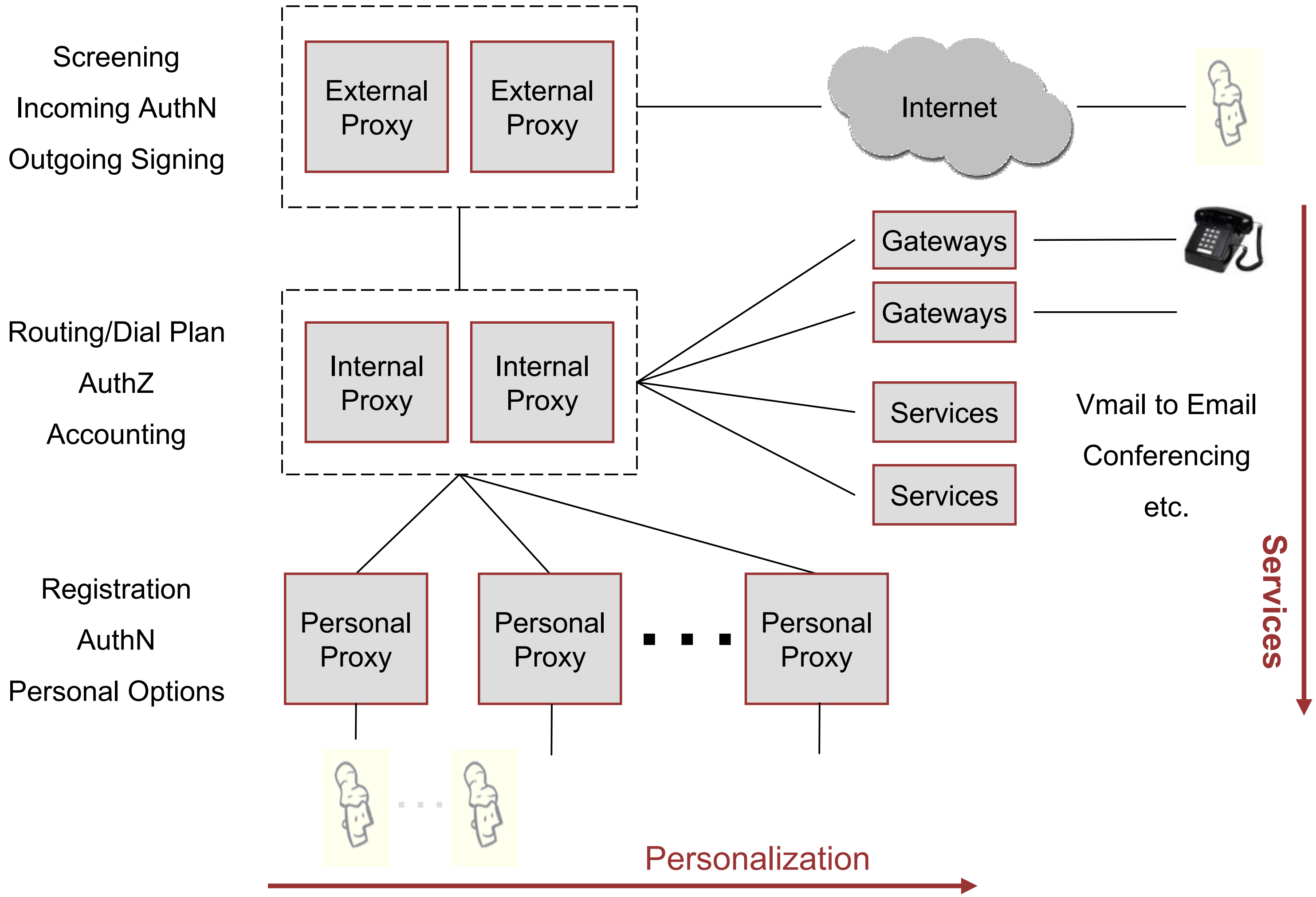
# Asterisk Server

- The “Swiss Army Knife” of the SIP infrastructure, scalable and extensible to implement many existing features and new ones
- Open source IP PBX, supporting a variety of VOIP protocols http://www.asterisk.org
- Provides voicemail service and voicemail to email service to SIP subscribers
- Provides conference bridge services and other PBX type functionality through the VOIP infrastructure
- Implemented on a RedHat Linux Enterprise platform. Redundancy implemented as hot standby server or server clustering

# SIP Gateway

- ISDN PRIs to our Lucent 5ESS PBX
- Upgraded from Cisco 2600 to 3825
- Moving to Cisco AS5800 platform
- Using Cisco Remote Party ID (RPID) to insert PSTN/PBX caller ID
- Redundancy provided through a combination of DNS SRV records and Hunt groups from the 5ESS

# Putting It All Together



# Management Tools

- Developed management tools for the configuration and management of individual MIT SIP accounts
- Built on Oracle, Apache and X509 certificate authentication
- Provides access to a variety of personalization options to allow the user to configure their SIP account to meet their needs