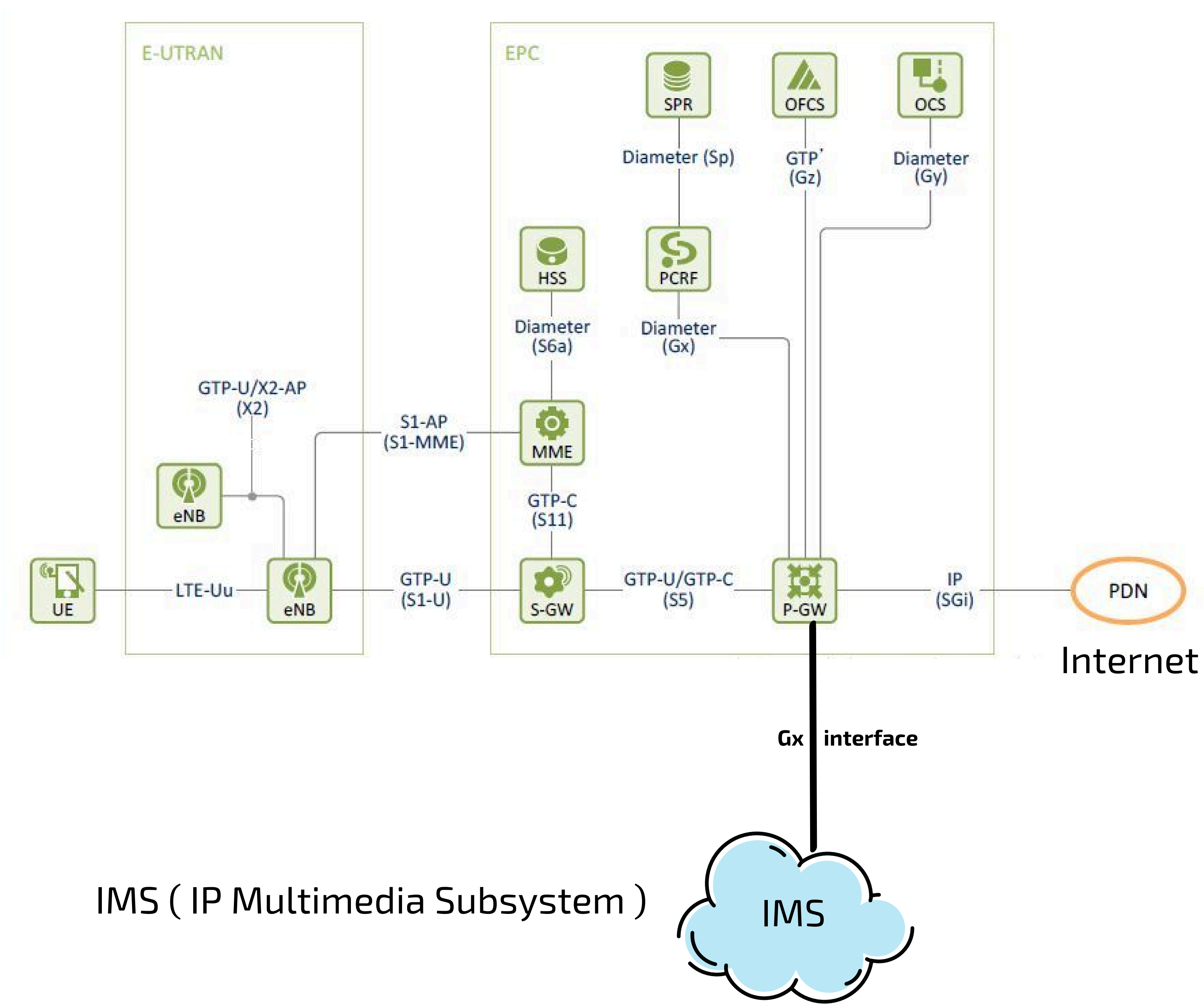
# LTE with IMS


E-UTRAN
EPC
SPR
OFCS
OCS
Diameter (Sp)
GTP'
(Gz)
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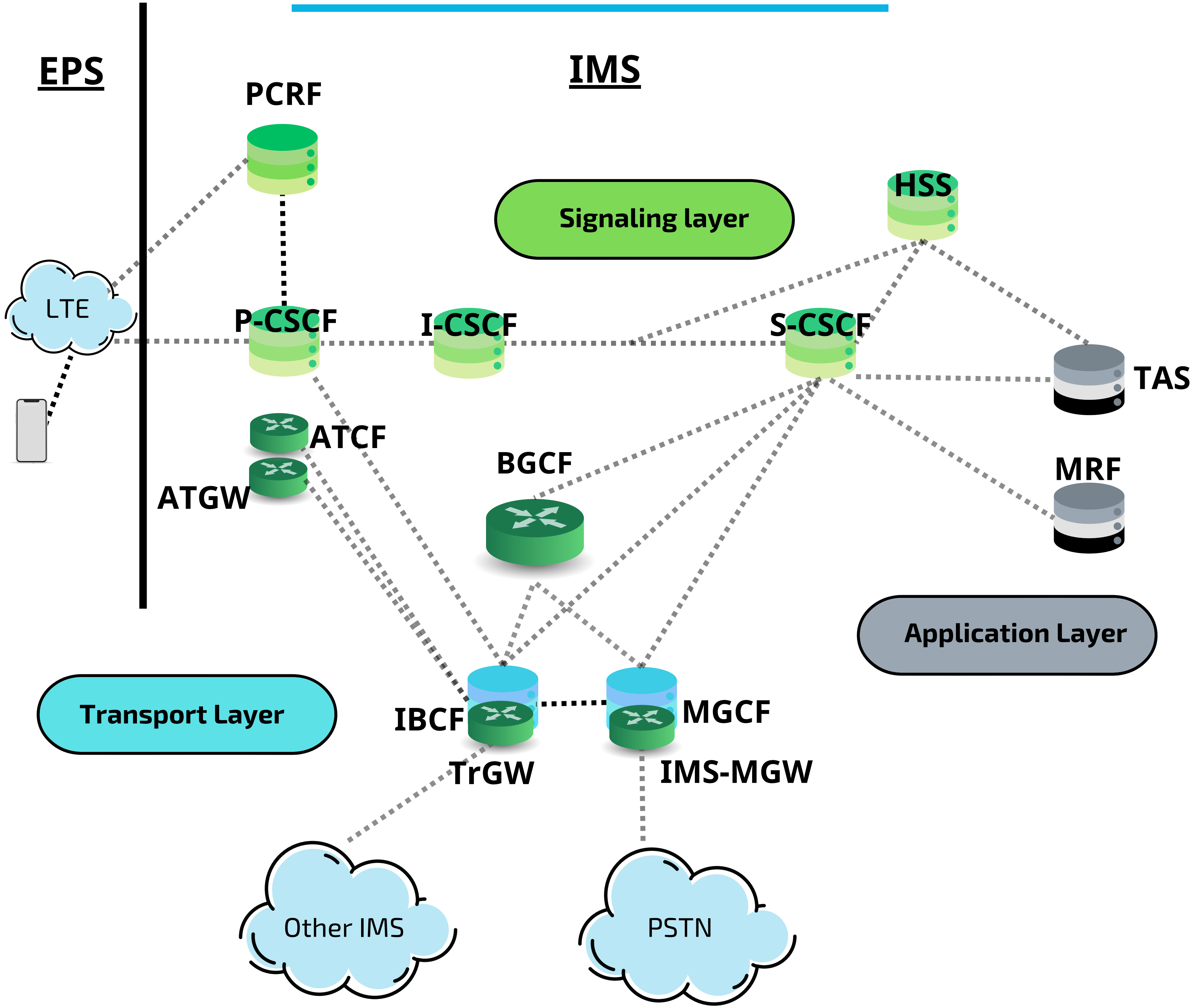
HSS
PCRF
Diameter
(S6a)
Diameter
(Gx)
GTP-U/X2-AP
(X2)
S1-AP
(S1-MME)
MME
eNB
GTP-C
(S11)
UE
LTE-Uu
eNB
GTP-U
(S1-U)
S-GW
GTP-U/GTP-C
(S5)
P-GW
IP
(SGi)
PDN
Internet
Gx interface
IMS ( IP Multimedia Subsystem )
IMS

# IMS Core Architecture

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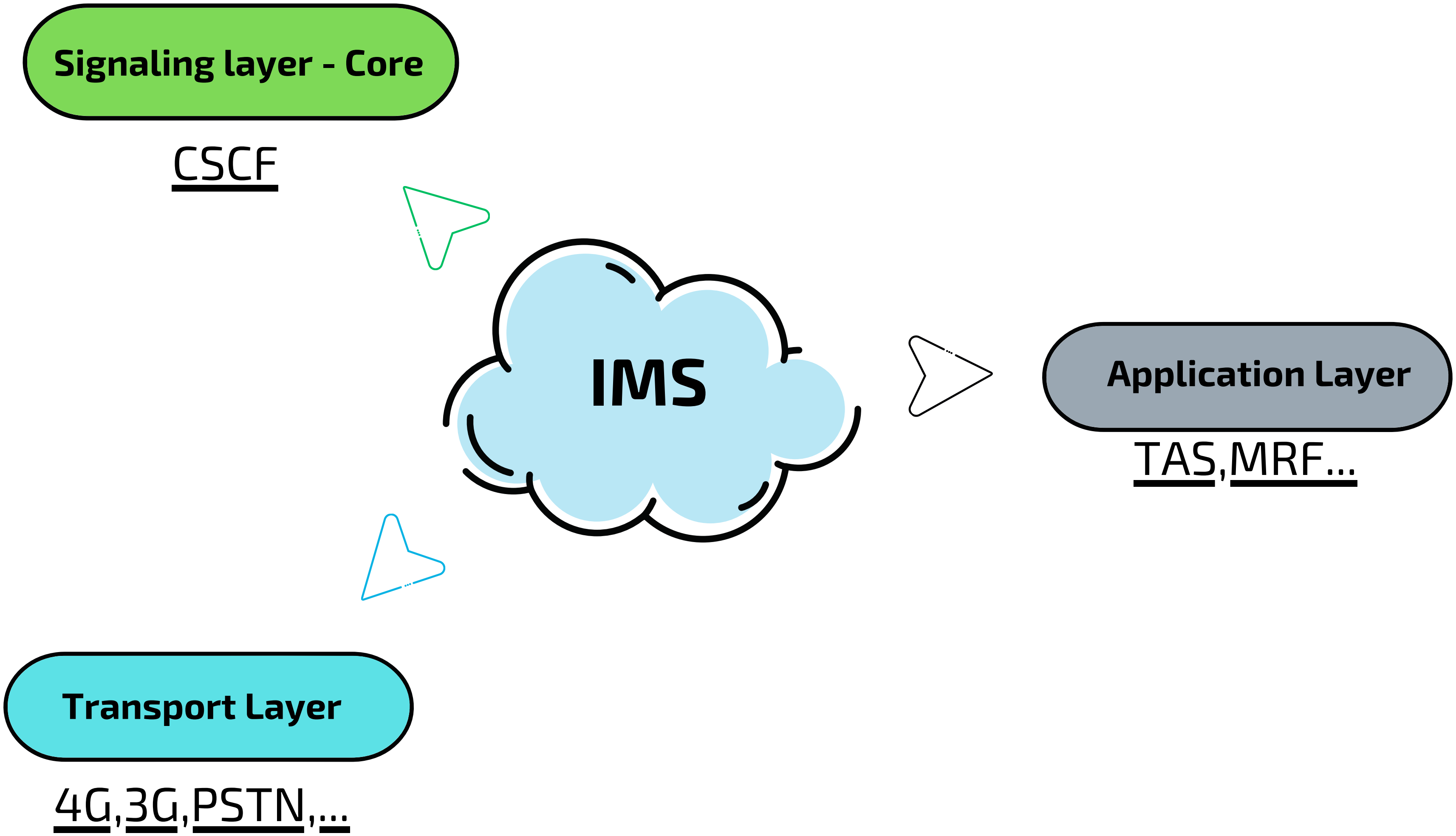
# Simple Core Overview

**IMS architecture involves a clear separation of three layers**



# Signaling layer (CSCF)

The CSCF is at the **core** of IMS, handling **SIP signaling** for all user sessions.

It has three main types:

**Proxy-CSCF (P-CSCF)** The entry point for SIP signaling.

1-Ensures secure communication using IPSec.

2-Routes SIP requests to the Serving-CSCF.

Signaling layer

**Serving-CSCF (S-CSCF)** The brain of IMS.

1-Handles SIP session control and user authentication.

2-Interacts with the HSS to retrieve user profiles and services.

3-Ensures the correct routing of calls and multimedia sessions.

**Interrogating-CSCF (I-CSCF)** The network gateway for inbound SIP messages.

1-Finds the correct S-CSCF for the user.

2-Acts as a firewall for SIP signaling.

# Application Layer

**Telephony App Server (TAS)**

provides essential services like **call waiting, call forwarding, and call hold**. While the IMS Core handles the signaling (setting up and ending calls), the TAS manages the extra services that users need during calls.

**Example:** When you're on a call and another call comes in, the TAS handles the call waiting feature, allowing you to put the current call on hold and answer the new one. In simple terms, the TAS adds value to VoLTE by providing these extra features, while IMS Core takes care of the call setup.

**Application Layer**

**Media Resource Function (MRF)**

handles media tasks like **call conferencing** and **RTP mixing** (mixing audio or video). It also plays network announcements and tones.

**Example**: During a VoLTE conference call, the MRF mixes the audio of all participants so everyone can hear each other clearly.

# Transport Layer

In VoLTE, we need gateways to connect **IMS** networks and **PSTN**. These gateways help with call handovers between different networks.

## IMS Gateway

Transport Layer

This connects **IMS networks**. It has two parts:

- IBCF (Control Plane) manages signaling.
- TrGW (User Plane) handles the actual media (voice or video).
- **Example**: When a VoLTE call moves from one IMS network to another, the **IBCF** controls the call setup, and the TrGW ensures the media is delivered properly.

## PSTN Gateway

This connects VoLTE to traditional **PSTN** networks. It also has two parts:

- MGCF (Control Plane) manages signaling.
- IMS-MGW (User Plane) handles the voice media.
- **Example**: When a VoLTE call is handedover to a traditional phone, the **MGCF** handles the call setup, and the IMS-MGW ensures the voice is transmitted correctly.

# Transport Layer

**ATCF** (Access Transfer Control Function) and **ATGW** (Access Transfer Gateway) are important nodes that help manage media and voice calls when users roam between networks.

**ATCF**

This node controls media handling for voice calls when users are connected to the LTE network. It works with the LTE PGW to manage voice traffic.

- It also plays a key role in **SRVCC** (Single Radio Voice Call Continuity), which helps transfer ongoing voice calls between different networks (like from LTE to 3G).

**Example**: If you're on a VoLTE call and move from LTE coverage to 3G, the **ATCF** ensures the call continues without dropping.

**ATGW**

This node handles the media anchor between LTE and the PGW, ensuring voice media is properly routed.

**Example**: When a VoLTE call is established, the ATGW ensures that the voice data flows smoothly between the LTE network and the PGW.

# BGCF

**Breakout Gateway Control Function**

BGCF helps decide where **to route a call in a VoLTE network**, just like a routing table in other networks. It determines the next step for SIP messages, which are used for setting up calls.

- **What BGCF Does:**

BGCF helps the Serving CSCF (another network function) decide whether a call should go to a PSTN network or another IMS network (like another VoLTE network).

- **How BGCF Routes Calls:**

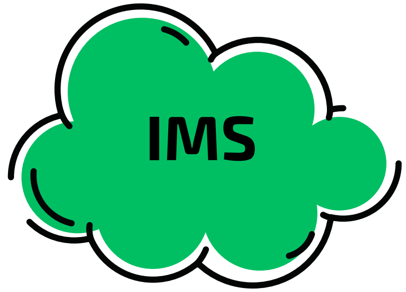
If the call is to a PSTN network, the **BGCF** selects the right **MGCF** to connect to that network.

If the call is going to another **IMS** network, the BGCF chooses the correct **IBCF** to connect the networks

**Example:**

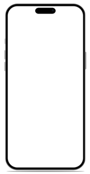
If you're making a **VoLTE** call to a **landline**, the **BGCF** decides which network to use and sends the call through the right gateway (**MGCF**). If the call is going to another VoLTE network, the BGCF ensures it connects to the correct gateway (**IBCF**).

# VoLTE Call flow Simple Overview 1

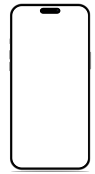

VoLTE User

**User A**



VoLTE User

**User B**



**SIP Invite** →

caller(A) starts the VoLTE call by sending a SIP INVITE message.

This includes an SDP offer with media details like codec and bandwidth for the call.

← **100 Trying**

The receiver (B) acknowledges the SIP INVITE by replying with 100 Trying

← **183 Session in Progress**

The receiver (B) sends a 183 Progress message with an SDP answer showing supported codecs.

It also requests network resources.

**At this step**, dedicated bearers are set up for both A and B via the PCRF

**SIP PRACK** →

The caller (A) sends a provisional acknowledgment for the 183 Progress message

← **SIP 200 Ok**

The receiver (B) responds to the PRACK with 200 OK

# VoLTE Call flow Simple Overview 2


**SIP Update** →

**User A** The caller (A) confirms resources are ready by sending a SIP UPDATE with a new SDP offer, showing preconditions are met and media is activ **User B**

← **SIP 200 Ok**

The receiver (B) replies with 200 OK confirming that resources are ready and the call can proceed

← **180 Ringing**

The receiver (B) starts ringing and sends a 180 Ringing message to the caller (A)

← **SIP 200 Ok** **Answer Calll**

When the receiver (B) answers, it sends 200 OK to the caller (A)

**SIP ACK** →

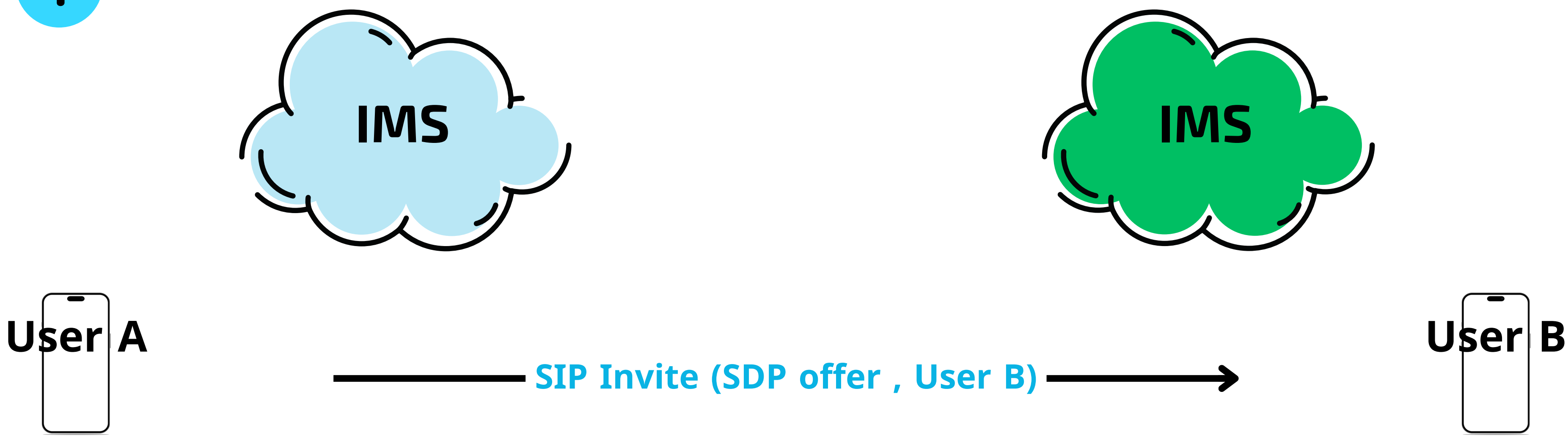
caller (A) sends an ACK, and the call is established. Voice traffic flows over the LTE network using dedicated bearers

# VoLTE Call flow


# VoLTE Call Flow Step by Step in Detail with Example

# SIP Call Flow – SIP Invite

1




**SIP Invite**: UE sends an **INVITE request**, which includes the destination subscriber's details (Request-URI), and uses the Route header to include the P-CSCF and S-CSCF addresses from registration.

The SIP INVITE also carries an **SDP** for media negotiation. Key details in the SIP INVITE:

SDP Offer:

- IMS media capabilities (media type, port, protocol)
- Requested bandwidth (max RTP session bandwidth in kbytes)
- Supported codec info (payload type, encoding name, clock rate)

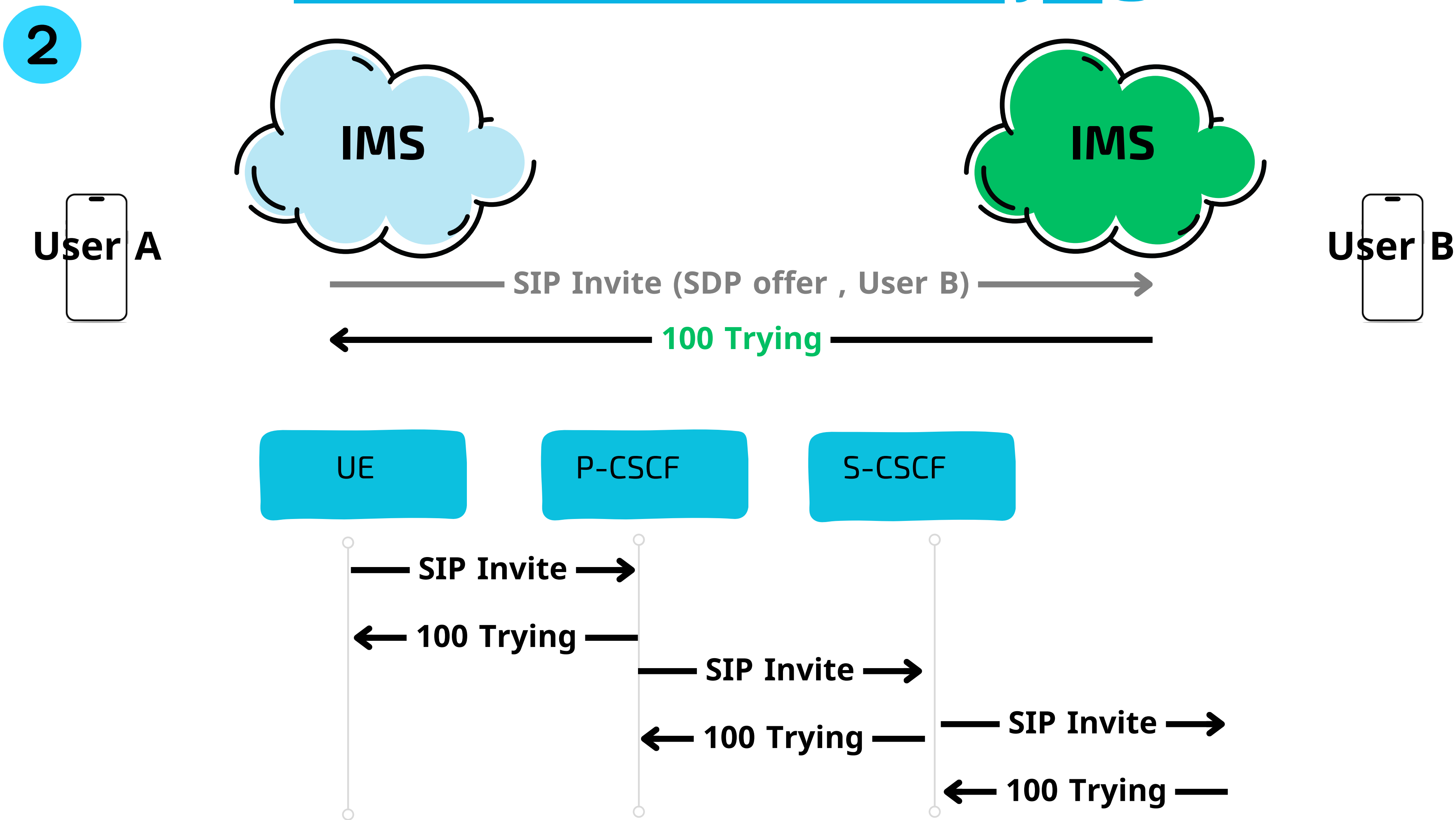
Preconditions showing that resources need to be reserved but aren't yet reserved

Other Information:

- A Party details (IMPU, IMPI)
- B Party details (tel-URI)

This SIP INVITE is used to set up the call with necessary media and network information.

# SIP Call Flow – 100 Trying




After receiving the SIP INVITE, each network hop responds with a 100 Trying message.

This is a provisional one-way response that informs the calling party that the request is being processed.

It's just an informational message and is not guaranteed to be delivered. If it's lost, the call setup can still continue without it.

# SIP Call Flow – 183 Session in progress

3


At this stage, the call preconditions are still not met, so the User B can't alert the user with a 180 Ringing. Instead, it sends a 183 Session Progress message with an SDP answer in response to the original SIP INVITE.

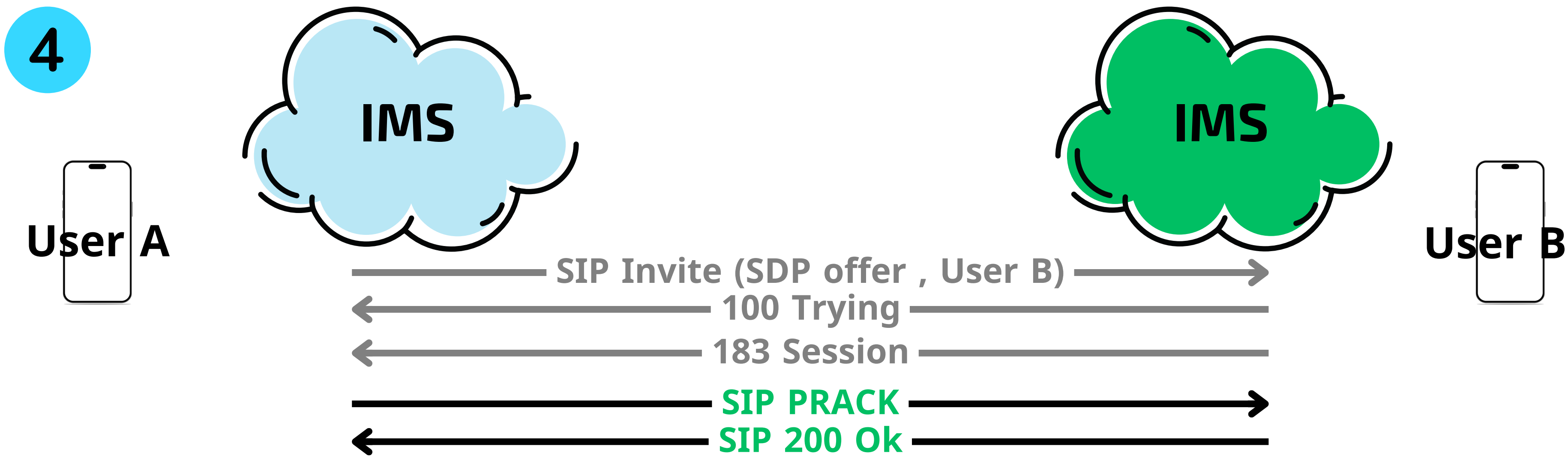
This SDP answer shows which codecs the called party supports, allowing both parties to agree on a common codec.

**Bearer Creation**

Dedicated bearers are created on both the originating and terminating sides for the call. When the 183 Session Progress is received, **the P-CSCF** triggers an Authentication & Authorization Request (AAR) to the **PCRF** to set up a new session.

**PCRF** generates rules for the SGW/PGW to create a bearer with **QCI=1 (for voice calls**) via the Gx interface. The PGW then initiates the EPS bearer creation and sends a response back to the PCRF, which informs the P-CSCF. Finally, the P-CSCF forwards the 183 Session Progress to the originating party.

# SIP Call Flow – SIP PRACK , 200 OK PRACK




**PRACK** (Provisional Response Acknowledgment) is used to ensure that temporary responses (like the "183 Session Progress" message) are reliably received during a call setup.

- When User A (the caller) receives the "183 Session Progress" message from User B (the receiver), User A sends a PRACK message to confirm that it got the 183 message.
- To make sure the 183 message is delivered correctly and isn't lost, the '100rel' extension is used. This tells User B to keep sending the "183 Session Progress" until it gets a PRACK response from User A or the timer runs out.

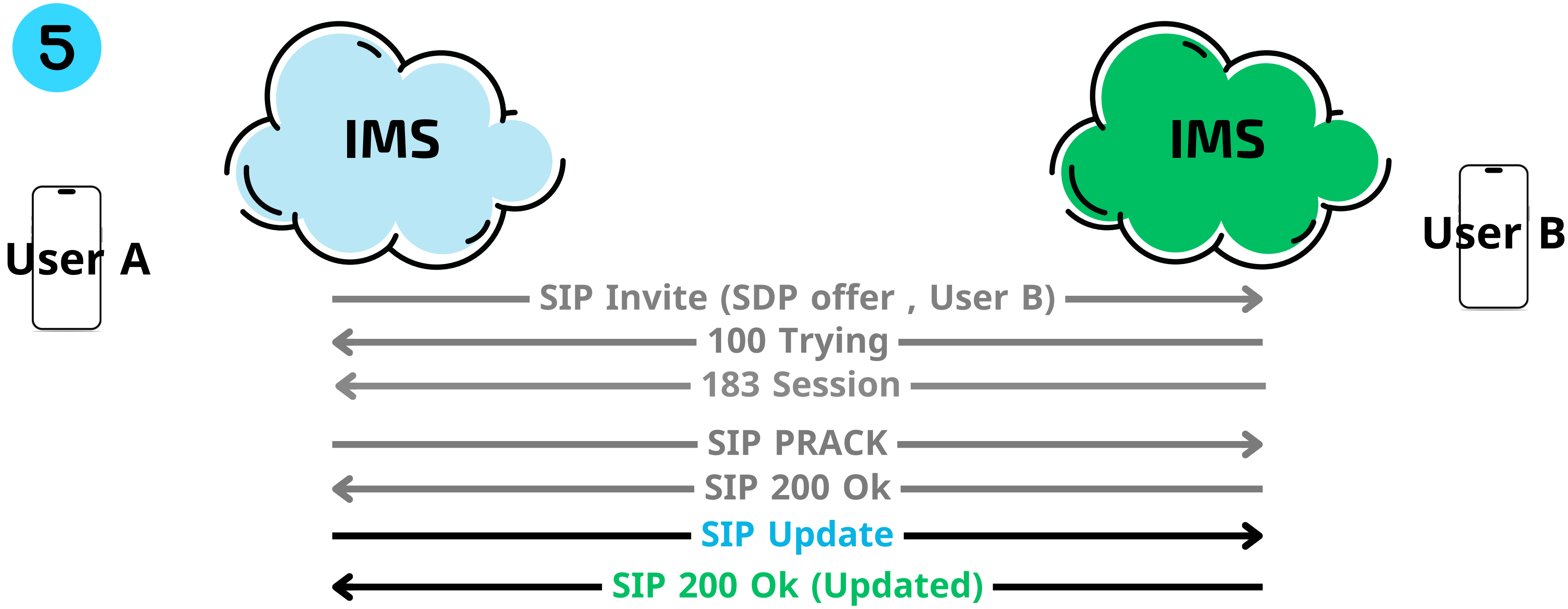
**Final Codec Selection** : When User A sends the PRACK message, it also offers the final codec it wants to use for the call (the audio format, like AMR or G.711). This is done through a second SDP offer.

**200 OK (PRACK)**

- **User B** responds to the PRACK message with a 200 OK This means User B accepts the codec chosen by User A
- At this point, both A and B have agreed on the codec for the call, **but resource reservation (setting up the actual network resources for the call) is still pending.**

# SIP Call Flow – SIP UPDATE , 200 OK UPDATE

5




**SIP UPDATE**

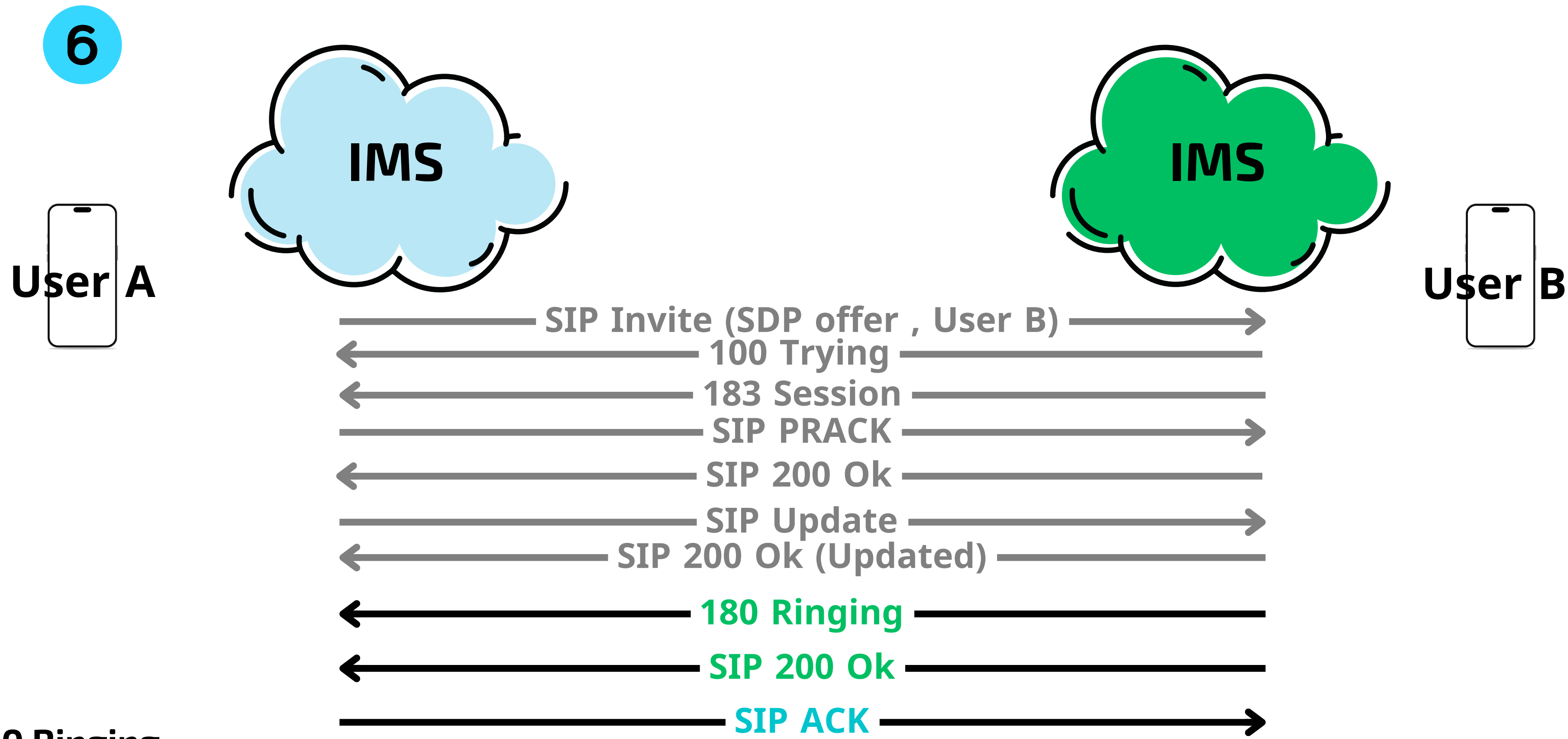
At this stage, User A sends a third offer to User B in the form of an UPDATE request. This request shows the status of resource reservation.

- User A doesn't change anything like the codec (since it's already agreed on through the PRACK message).
- User A's main task here is to reserve the resources for the call and send the UPDATE request.

**200 OK (UPDATE)**

In response, User B sends a 200 OK message. This confirms that User B also reserves resources and agrees with the request from User A.

# SIP Call Flow – 180 Ringing , SIP 200 OK INVITE , SIP ACK




**180 Ringing**

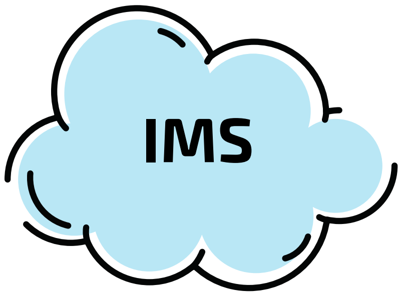
The 180 Ringing message is sent from User B to User A. This indicates that the call is being alerted on the User B side and that the phone is ringing.

**SIP 200 OK (INVITE)**

Once User B answers the call, it sends a 200 OK message back to User A. This confirms that User B has accepted the call.

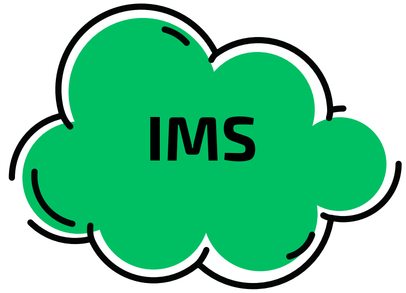
**SIP ACK**

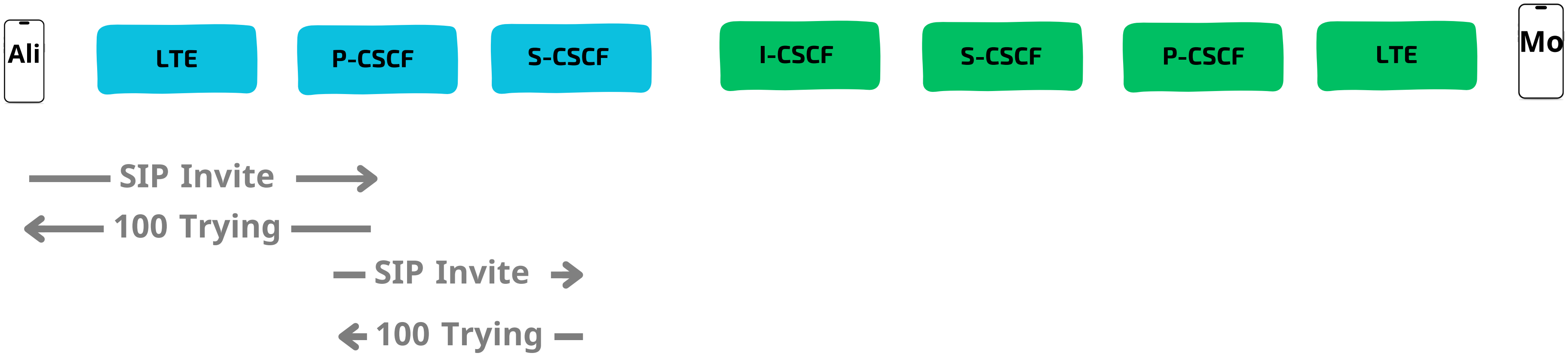
User A sends an ACK message to User B as confirmation. The ACK message indicates that the call is now established. From this point on, the voice traffic is transmitted over the dedicated bearer from User A's IMS to User B's IMS and finally to User B.




# Example

## Step1




**Ali picks up his phone and dials Mo's number**

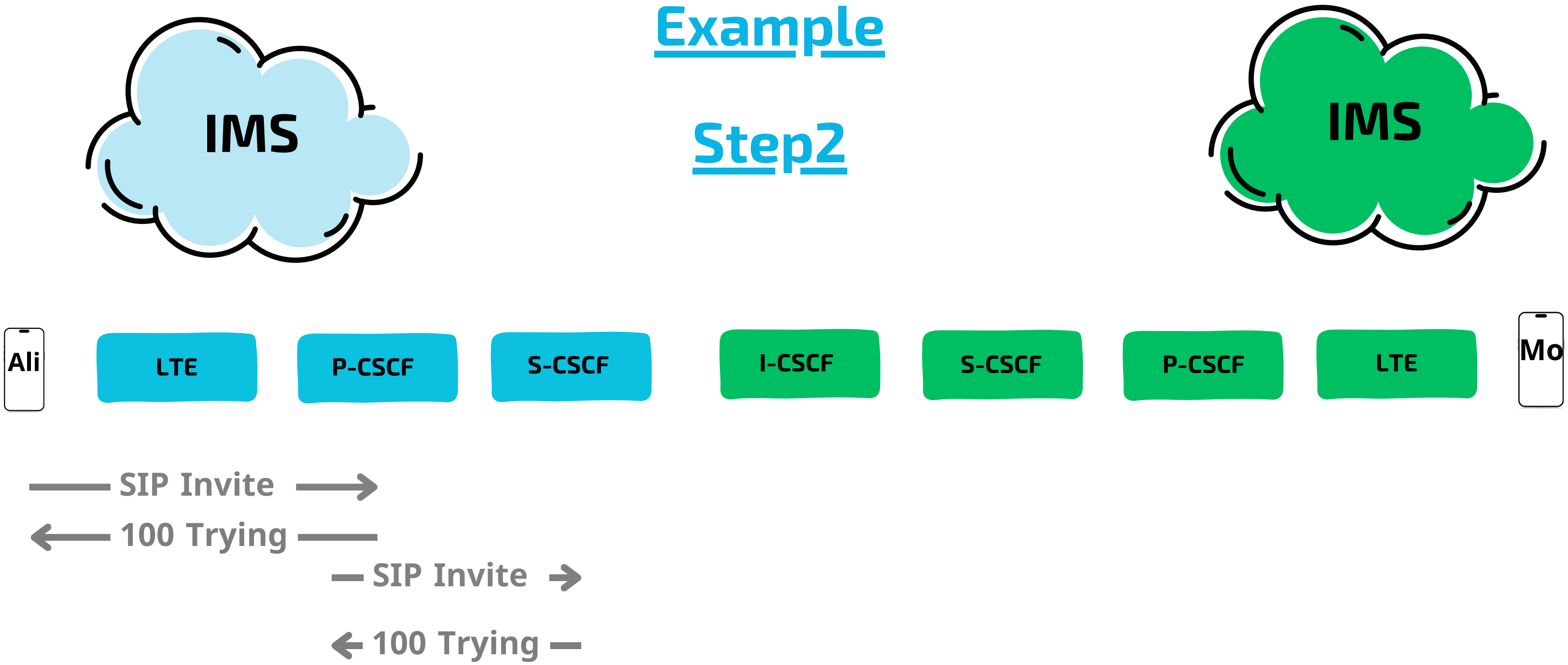
His phone sends a SIP INVITE message to his network's P-CSCF. This INVITE says, "I want to call MO" and includes initial information like the type of call (voice) and Ali's phone details.

The P-CSCF checks the message and forwards it to the S-CSCF in Ali's IMS.

The S-CSCF sends back a 100 Trying message to confirm, "I've received your request and I'm working on it."

# Example

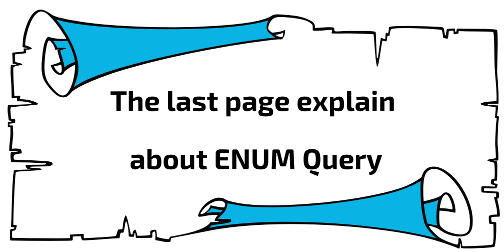
## Step2




### Checking Services for Ali (Outgoing Services)

Before proceeding, the **S-CSCF** asks the **TAS** if there are any special services for Ali, **like call forwarding or restrictions**.

If everything is clear, it moves to the next step.

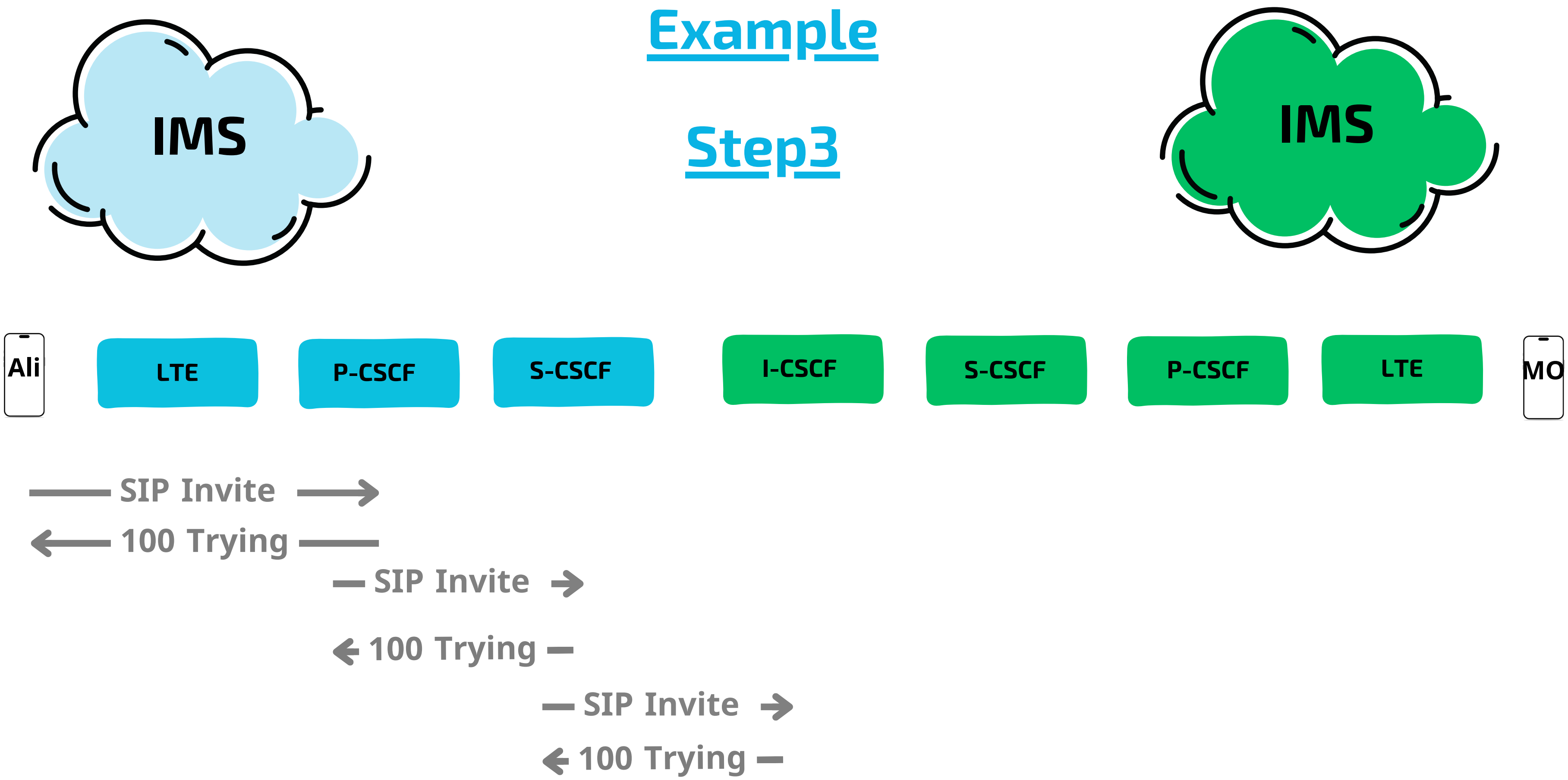



### Finding MO's Network (ENUM Query)

The **S-CSCF** doesn't know which network MO is on, so it performs an **ENUM query** to figure out which IMS handles MO 's phone number.

# Example

## Step3




**Handoff to MO's IMS Network:**

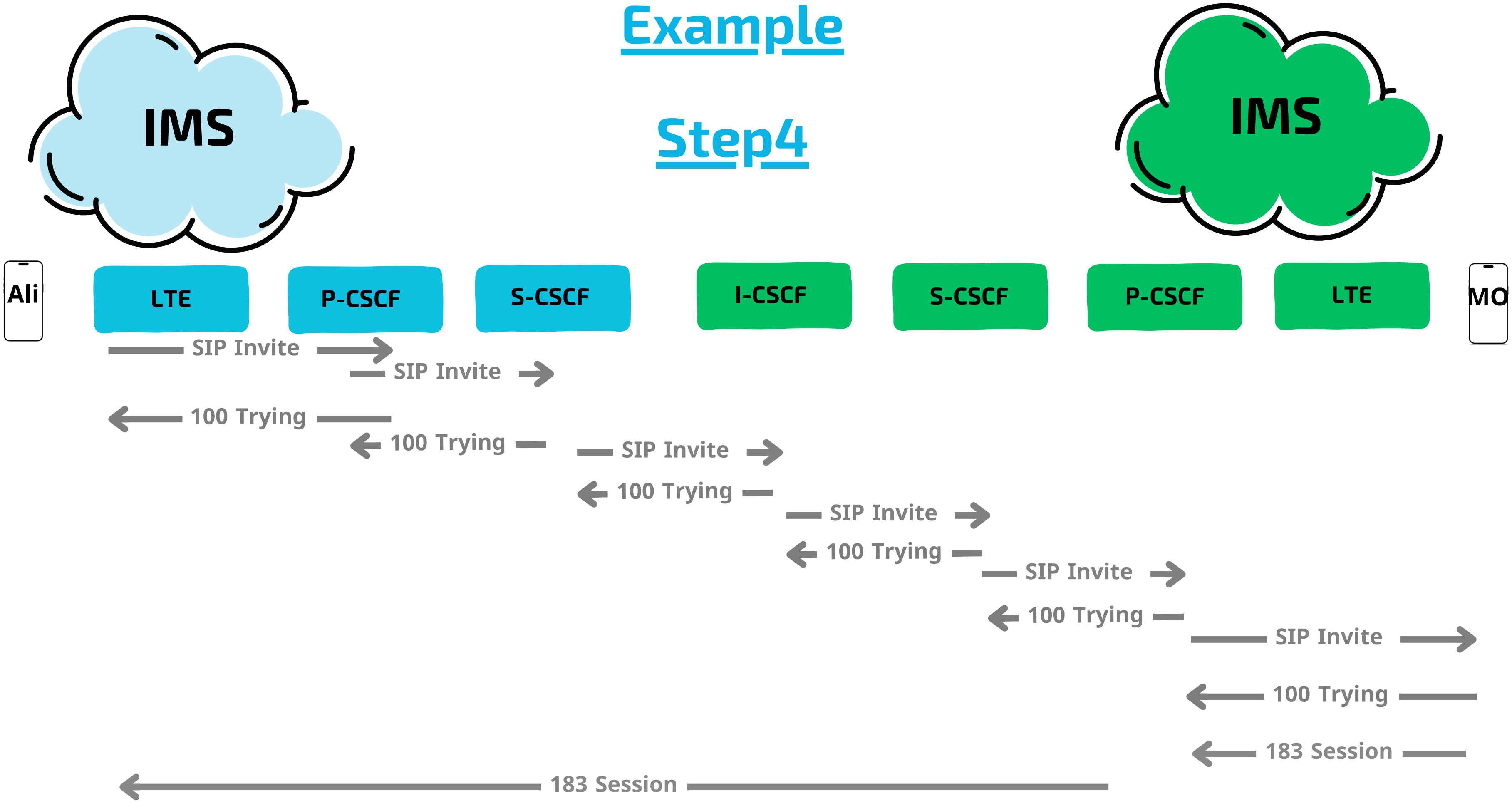
- Once Ali's IMS identifies MO's network, it handsover the call request to MO's IMS
- In MO's IMS, the **I-CSCF** checks with the **HSS** to locate MO's profile and find out which **S-CSCF** is managing his phone.

**Checking Services for MO (Incoming Services)**

- MO's **S-CSCF** checks with its **TAS** for any terminating services, like voicemail or call blocking. Once approved, it sends the <u>SIP INVITE to MO's phone.</u>

# Example

## Step4




## MO's Phone Starts the Setup:

MO's phone receives the SIP INVITE and responds with a 183 Session Progress message. This means, I'm getting ready for the call, and includes information about the SDP to agree on call settings (codec and media streams)

## Dedicated Bearer for MO's Voice Traffic:

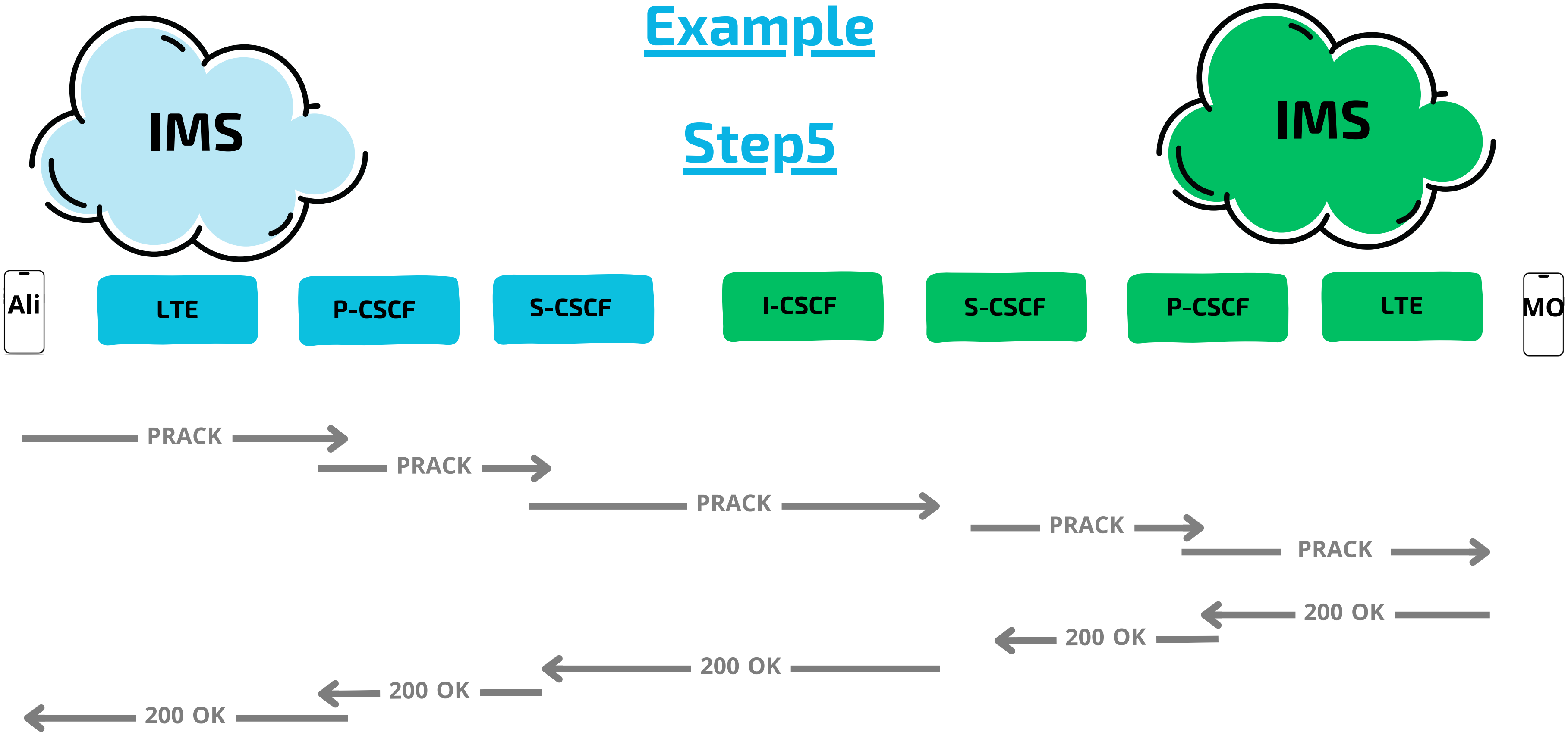
MO's network creates a dedicated bearer for the voice traffic using PCRF with QCI=1, This ensures the voice quality is excellent.

## Ali Gets the Session Progress:

MO's network sends the 183 Session Progress back to Ali's network, passing through all IMS nodes in reverse order. Ali's network also creates a dedicated bearer for his voice traffic using **PCRF**.

# Example

## Step5



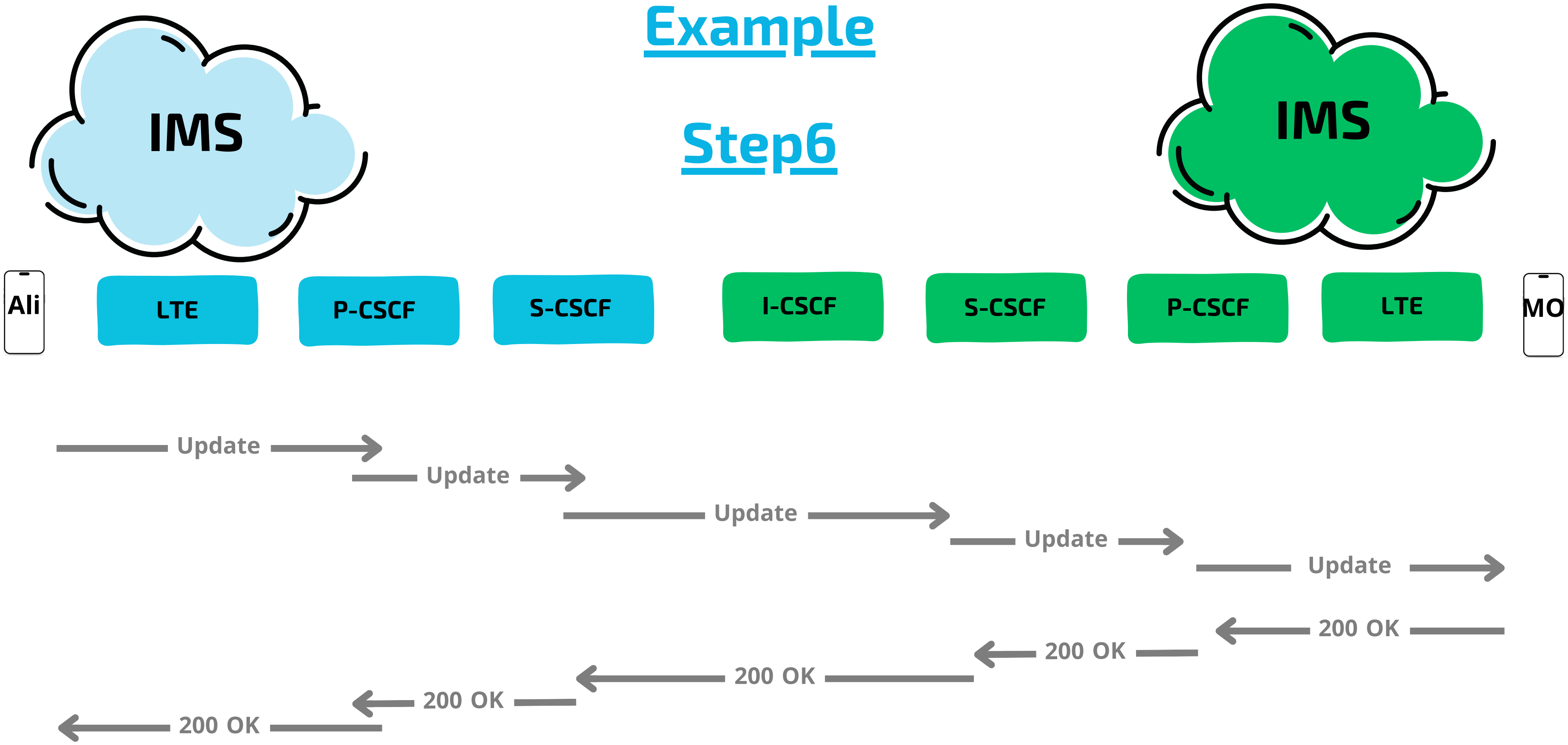

**Acknowledging Progress (PRACK):**

Ali's phone confirms that it received the 183 Session Progress by sending a **PRACK** message to MO's network.

MO's phone replies with a **200 OK**, saying, Got it, let's move forward.

# Example

## Step6




**Resource Reservation (SIP UPDATE)**

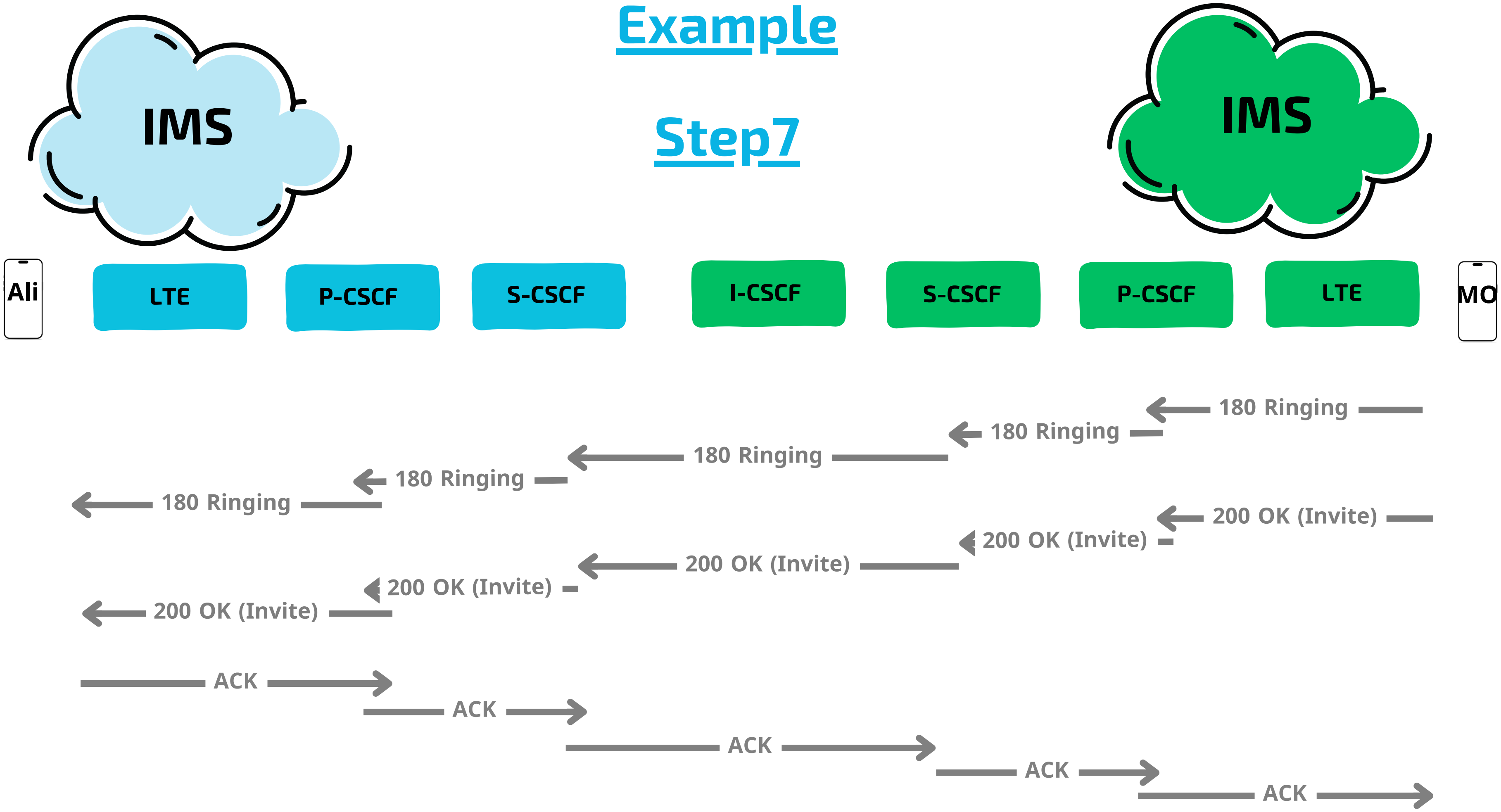
Ali's phone now reserves its internal resources (esetting up codecs) based on the SDP answer.

It sends a SIP UPDATE to MO's phone to finalize the setup.

MO responds with 200 OK, confirming that the setup is complete.

# Example

## Step7




**MO's Phone Rings (SIP 180 Ringing)**

MO's phone starts ringing and sends a SIP 180 Ringing response back to Ali, letting his know that the call is ringing on MO's side.

**Call Answered (SIP 200 OK)**

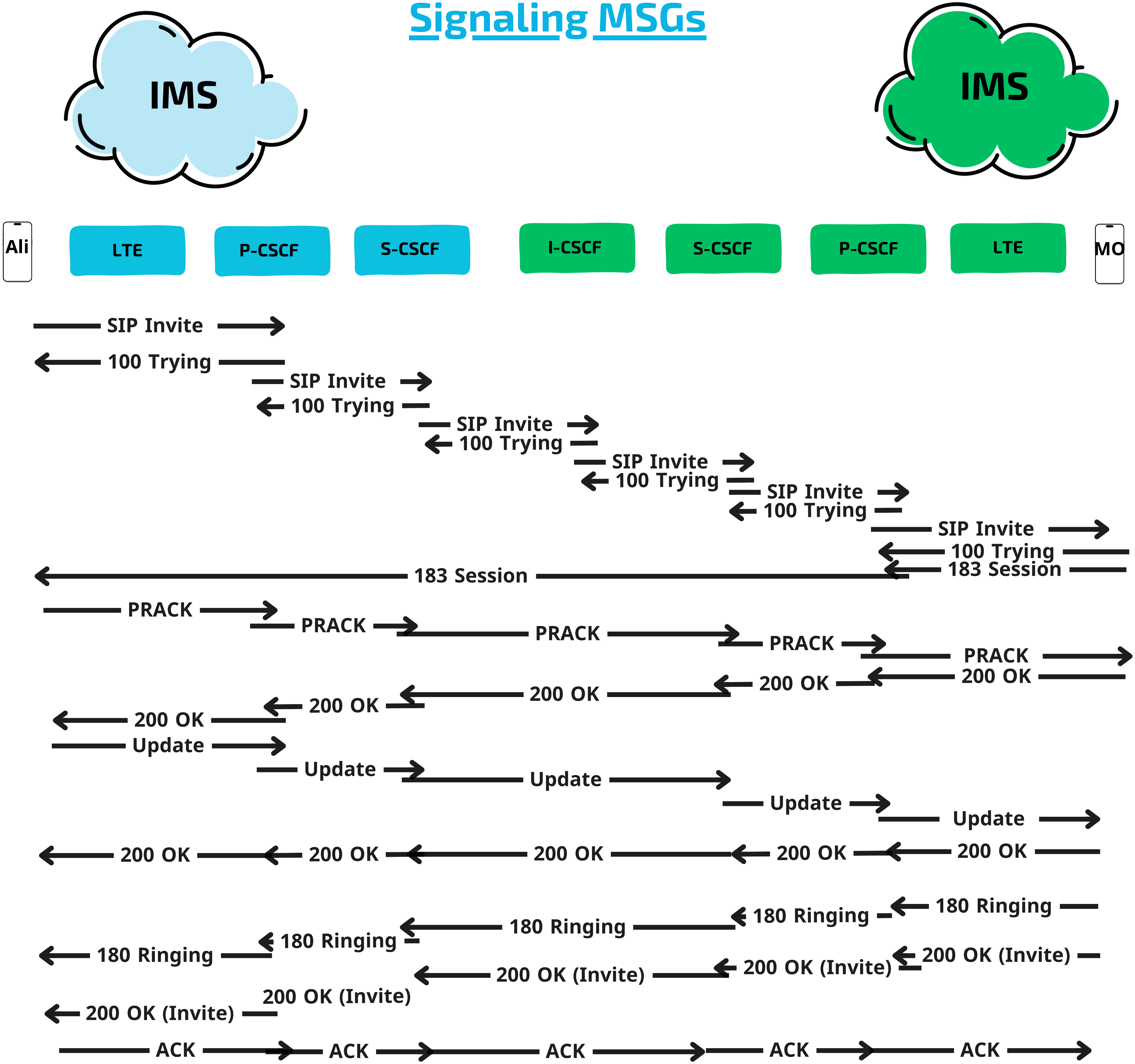
When MO answers the call, his phone sends a SIP 200 OK back to Ali's phone.

**Final ACK (Call Established)**

Ali's phone sends a final ACK message to confirm the call.

At this point, the signaling phase is complete.

# Signaling MSGs

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# ENUM Query

When Ali calls MO, Ali's **S-CSCF** only has **MO's phone number**.

But it doesn't know:

Which IMS network manages MO's number ?

How to route the call to MO ?

To figure this out, the **S-CSCF** sends MO's phone number to an **ENUM system**

**How Does ENUM Work?**

ENUM system takes MO's phone number (+31612345678) and translates it into a domain name using DNS. For example:

1.Remove the + sign.

2.Reverse the digits and add special suffixes.

MO's number **+31612345678** becomes **8.7.6.5.4.3.2.1.6.3.e164.arpa**

3.The ENUM system checks its database to see if this domain exists and returns the associated IMS network or SIP URI (sip:MO@ims.provider.com)

**ENUM is not a node in IMS network. Instead, it is a service or function that is external to the IMS architecture**

**It uses a DNS-based system to perform the translation of phone numbers to SIP addresses or domain names**

**Eng. Al-Ali**