## What is FMC (Fixed Mobile Convergence)?

In the text below, the FMC service refers to the service of connecting a SIM card to a Virtual PBX in such a way that all incoming and outgoing calls go through the PBX. In this case, the end user continues to use their SIM card as usual. However, all calls will be displayed in the History and Statistics, can be recorded in the CRM, and the PBX will also save recordings of these calls.

## Principle of Call Separation

When making calls from a phone with the FMC service to another phone with the FMC service, the call is divided into several stages. Generally, the stages are as follows: a) from the Phone Device (PD) to the PBX (MOC – Mobile Originated Call), b) from the PBX (Virtual PBX) to the gateway, c) from the gateway to the PBX, d) and from the PBX to the PD (MTC – Mobile Terminated Call).

FMC implementation allows for the identification of the call's stage, specifically, separating a call initiated by the user on the PD (a) from an incoming call from the gateway to the PBX (c), as well as a call that should go directly to the PD (d) from an outgoing call from the PBX through the gateway (b).

The methods of call separation can be divided into two categories:

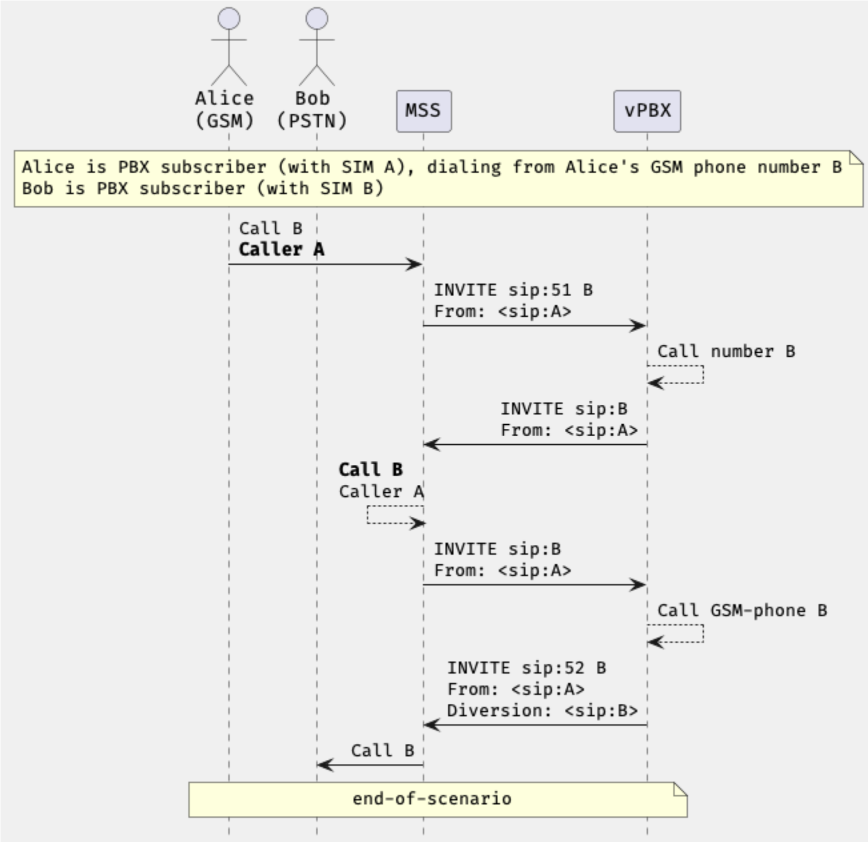
## Connection using IMS (IP Multimedia Subsystem)

This connection method involves using SIP headers to mark the stages of the call. Most often, the operator suggests using prefixes in the dialed number for this purpose.

For example, a call from the PD is sent with prefix 51 (a), an incoming call to the PBX is sent without a prefix (c), an outgoing call from the PBX through the gateway is sent without a prefix (b), and a call that should go directly to the PD is received from the PBX with prefix 52 (d).

Different IPs or ports may also be used (for example, using the standard port 5060 for incoming calls from the gateway, and port 6060 for calls from the phone).

To set up this type of connection, a SIP trunk needs to be organized to the centrex servers.




## Connection using CS (Circuit-Switched)

This method involves connecting the centrex-scp module directly to the STP (or MSS). In the HLR (or HSS), the centrex-scp module must be registered in the DP2 and DP12 triggers and should receive the IDP from calls made from numbers with the FMC service. The details of this connection are more flexible but highly depend on the operator's capabilities and specifics. The most common way to divide call stages involves using one or two technical numbers (calls to which should be routed to the PBX).

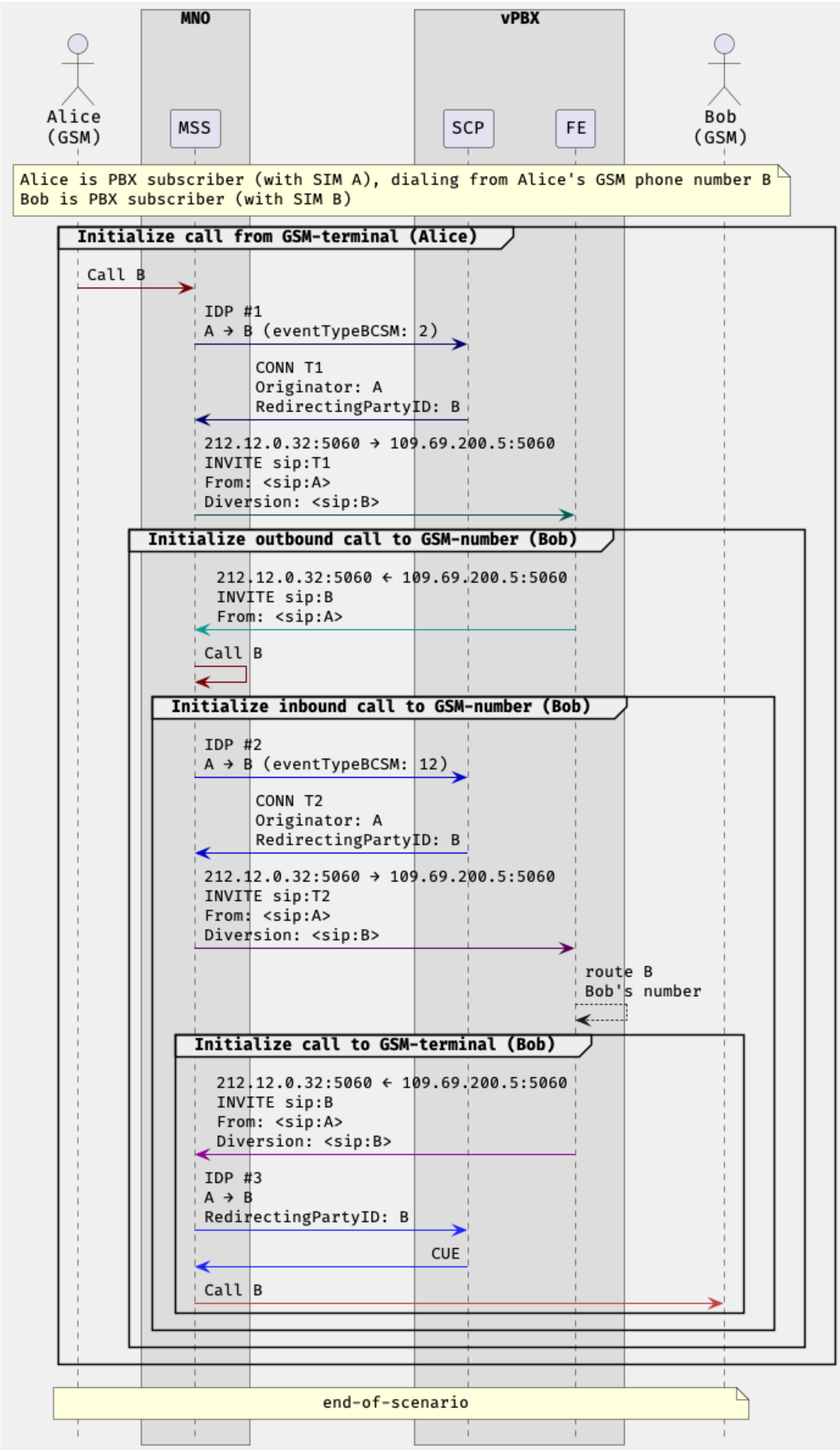
For example, a call from number A to number B, when FMC service is enabled on both numbers, will work as follows using SCP connection with two technical numbers:

a) A call from TA (number A) to any number (number B) triggers the BCSM-2 trigger and generates IDP#1 with the caller's number as A and the called number as B. SCP, upon receiving this IDP#1, sends a CONN response with a technical number marking the call as MOC (number T1) and a RedirectingPartyID field equal to the dialed number (number B). Then, the call is routed to the PBX on number T1 with From: A and Diversion: B.

b) After processing the call, the PBX generates a call to the dialed number (number B), showing the caller ID as A without the Diversion field, marking the call as outgoing.

c) When receiving an incoming call to a number with the FMC service, the BCSM-12 trigger is activated, generating IDP#2 with the caller's number as A and the called number as B with an empty RedirectingPartyID field, marking the call as a regular incoming call. SCP responds with a CONN packet to the technical number T2, filling the RedirectingPartyID field. The PBX receives the call on number T2 with From: A and Diversion: B and processes the call as incoming.

d) After processing the call, the PBX sends the call to number B, showing the caller ID as A in the From field and Diversion as B. SCP receives IDP#3 after the BCSM-12 trigger is activated, but with a filled RedirectingPartyID field from Diversion, marking the call as MTC. SCP responds with a CUE packet, leading to the call to number B through the phone network.



**Summary:**

- The first method (with prefixes) involves a simpler integration, where the platform receives a number with a prefix that helps identify the type and source of the call.
- The second method (with SCP) requires deeper integration, where the platform directly interacts with the operator's core, providing more flexibility in configuration. Specifically, it allows for bypassing the PBX in case of an emergency or if the subscriber hasn't paid for the PBX service (in this case, the outgoing call from TA will proceed as if the PBX was not involved, directly through the operator's phone network).