Advanced Call Control Features

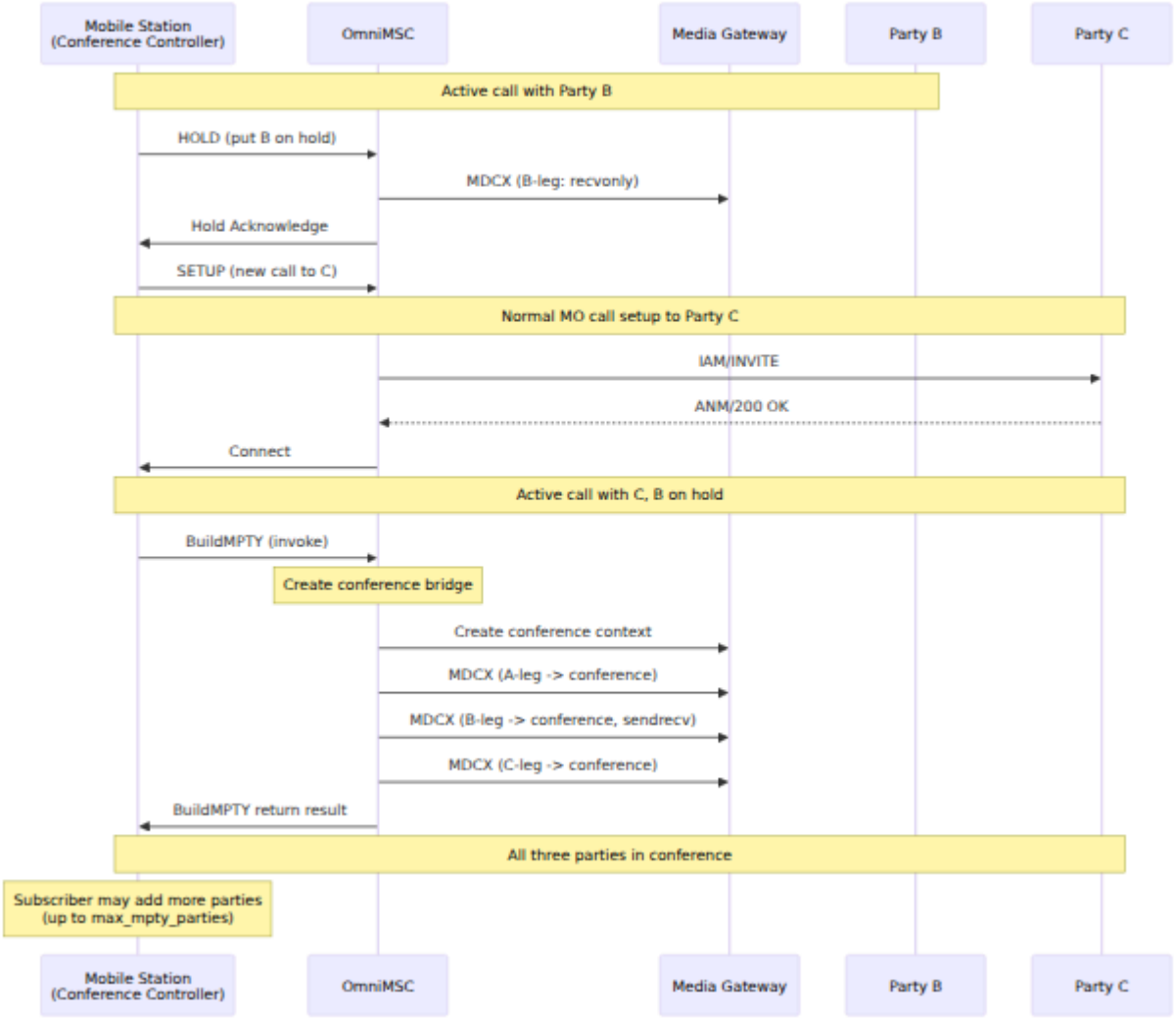
This document describes the advanced circuit-switched call control features implemented in OmniMSC, including multi-party calling, call transfer, call completion, call deflection, multicall, priority services, and charging.

For basic supplementary services (call forwarding, barring, CLIP/CLIR, USSD), see Supplementary Services. For call flow diagrams, see Call Flow Diagrams. For configuration parameters, see Configuration Reference. For general operations, see Operations Guide.

MPTY (Multi-Party / Conference Calling)

MPTY allows a subscriber to establish a conference call with multiple remote parties per 3GPP TS 24.084. The subscriber acts as the conference controller, bridging held and active calls into a single multi-party conversation.

MPTY Sequence



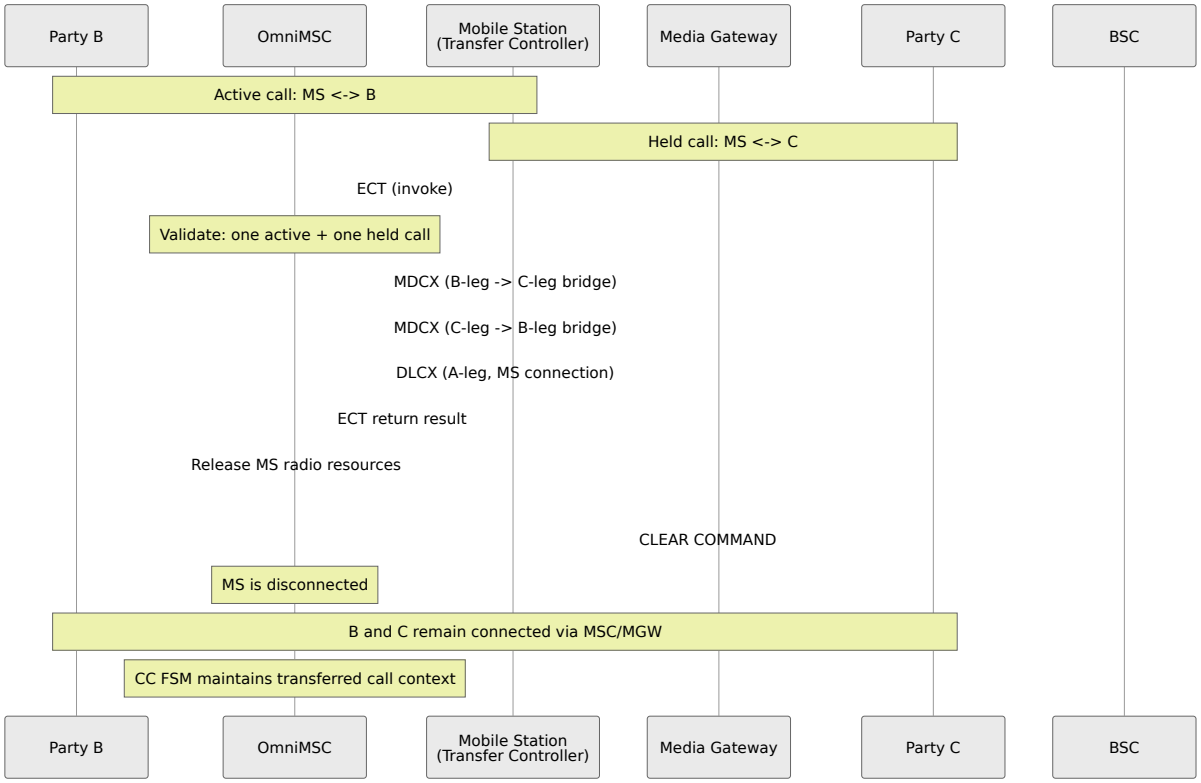
MPTY Configuration

Parameter	Type	Default	Description
`max_mpty_parties`	`integer`	`6`	Maximum number of parties in a single conference (including the controller). Per 3GPP TS 24.084, the minimum required is 3.
`mpty_tone_on_join`	`boolean`	`true`	Play a notification tone when a party joins the conference.
`mpty_tone_on_leave`	`boolean`	`true`	Play a notification tone when a party leaves the conference.

ECT (Explicit Call Transfer)

ECT allows a subscriber to connect two calls together and then drop out of the connection per 3GPP TS 24.091. The subscriber must have one active call and one held call. After ECT, the two remote parties are connected directly through the MSC.

ECT Sequence



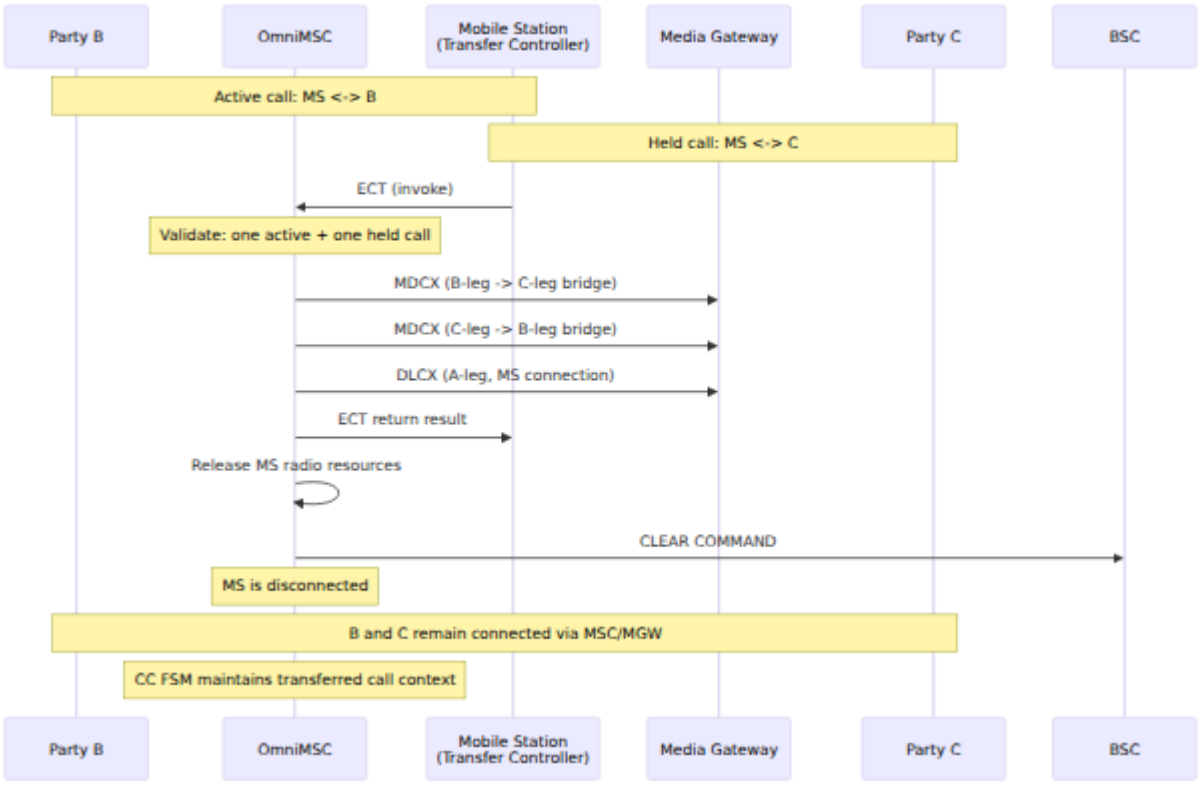
ECT Configuration

Parameter	Type	Default	Description
`ect_alerting_allowed`	`boolean`	`true`	Whether ECT is permitted when the second call is in alerting state (not yet answered). When `false`, both calls must be answered before ECT.

CCBS (Call Completion to Busy Subscriber)

CCBS allows a caller to request automatic callback when a busy called subscriber becomes free, per 3GPP TS 24.093 and 3GPP TS 23.135. The MSC monitors the called subscriber's state and initiates a callback when they become idle.

CCBS Flow



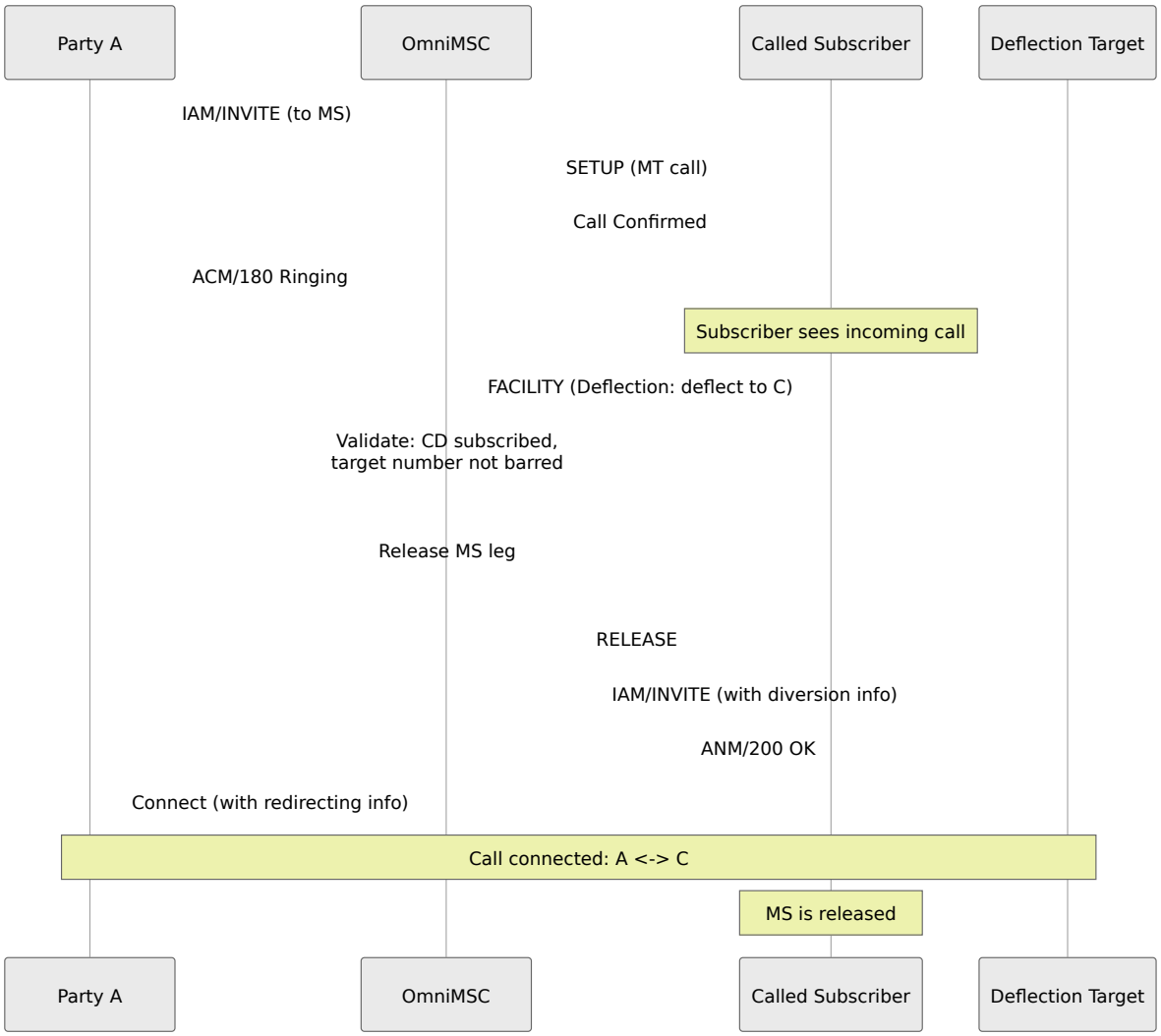
CCBS Configuration

Parameter	Type	Default	Description
`ccbs_queue_size`	`integer`	`5`	Maximum number of pending CCBS requests per subscriber. Additional requests are rejected. Per 3GPP TS 23.135 Sec 4.2.
`ccbs_supervision_timer`	`integer`	`180`	Supervision timer in seconds. If the called subscriber does not become free within this period, the CCBS request expires.
`ccbs_recall_timer`	`integer`	`20`	Time in seconds to wait for the originating subscriber to answer the CCBS recall.
`ccbs_retain_timer`	`integer`	`30`	Time in seconds to retain the CCBS request after the called subscriber becomes free, in case the subscriber becomes busy again before the recall completes.

Call Deflection

Call Deflection (CD) allows a called subscriber to redirect an incoming call to another number before answering, per 3GPP TS 24.072. Unlike call forwarding, CD is an on-demand action initiated by the subscriber during the alerting phase.

Call Deflection Sequence



Call Deflection Configuration

Parameter	Type	Default	Description
`cd_max_redirections`	`integer`	`5`	Maximum number of sequential call deflections to prevent loops.

Multicall Configuration

Multicall allows a subscriber to maintain multiple simultaneous CS calls per 3GPP TS 23.135. This is distinct from MPTY (conference): in multicall, each call has independent audio. The subscriber switches between calls using the HOLD/RETRIEVE mechanism.

Parameter	Type	Default	Description
`max_calls_per_subscriber`	`integer`	`2`	Maximum number of simultaneous CS calls subscriber. One call is active, remaining are
`max_bearers_per_subscriber`	`integer`	`2`	Maximum number of simultaneous radio be Typically matches `max_calls_per_subsc`

---

eMLPP (Enhanced Multi-Level Precedence and Pre-emption)

eMLPP provides priority call handling per 3GPP TS 24.067. Calls are assigned a priority level, and higher-priority calls may pre-empt lower-priority calls when resources are scarce.

Priority Levels

Priority Level	Name	Description
0	A	Highest priority (Flash Override). Reserved for national emergency authorities.
1	B	Flash. Reserved for senior government/military officials.
2	0	Immediate. High-priority government/military.
3	1	Priority. General priority traffic.
4	2	Routine. Standard subscriber calls (default).

eMLPP Configuration

Parameter	Type	Default	Description
`emlpp_enabled`	`boolean`	`false`	Enable eMLPP priority call handling.
`emlpp_default_priority`	`integer`	`4`	Default priority level for calls without explicit priority (Routine).
`emlpp_preemption_enabled`	`boolean`	`true`	Whether higher-priority calls may pre-empt lower-priority active calls when resources are exhausted.
`emlpp_preemption_tone`	`boolean`	`true`	Play a preemption warning tone before disconnecting the pre-empted call.

AoCC (Advice of Charge - Charging)

AoCC provides real-time charge information to the subscriber during a call per 3GPP TS 24.086. The MSC sends charge rate information and accumulated charge indications to the mobile station.

AoCC Configuration

Parameter	Type	Default	Description
`aocc_enabled`	`boolean`	`false`	Enable Advice of Charge - Charging.
`aocc_currency`	`string`	`"EUR"`	Currency code (ISO 4217) for charge display.
`aocc_rate_source`	`atom`	`:camel`	Source of charging rates: `:camel` (from SCP via CAP), `:local` (from local rate table), `:cdr` (from CDR parameters).
`aocc_update_interval`	`integer`	`10`	Interval in seconds between charge update notifications to the subscriber.

3GPP Specification References

Specification	Title	Feature
TS 24.084	Multi-Party (MPTY) Supplementary Service	MPTY / Conference calling
TS 24.091	Explicit Call Transfer (ECT) Supplementary Service	ECT
TS 24.093	Call Completion to Busy Subscriber (CCBS)	CCBS
TS 23.135	Multicall	Multicall, CCBS network procedures
TS 24.072	Call Deflection Supplementary Service	Call Deflection
TS 24.067	enhanced Multi-Level Precedence and Pre-emption (eMLPP)	eMLPP priority levels
TS 24.086	Advice of Charge (AoC) Supplementary Services	AoCC charging
TS 24.083	Call Waiting and Call Hold	Hold/Retrieve for multicall

REST API

This document describes the OmniMSC REST API, which provides programmatic access to subscriber data, active calls, routing configuration, SIP peers, media gateways, RAN connections, and system status. The API listens on port 8444 and serves an auto-generated OpenAPI 3 (OAS3) specification.

For the web-based control panel, see Control Panel. For routing concepts referenced by the route endpoints, see Routing Configuration.

OpenAPI Documentation

OmniMSC automatically generates an OpenAPI 3 specification from the API router. The interactive Swagger UI is available at `http://<host>:8444/schema` and provides a browsable, testable interface for all endpoints.

Endpoints

All endpoints are served under the `/api` path prefix. Request and response bodies use JSON encoding.

Subscribers

Method	Path	Description
GET	/api/subscribers	List VLR subscribers. Supports optional query parameters for filtering by IMSI or MSISDN (partial match).
GET	/api/subscribers/{id}	Retrieve full detail for a single subscriber, including identity, location, authentication state, service profile, and supplementary services.
DELETE	/api/subscribers/{id}	Purge a subscriber record from the VLR. Triggers MAP PurgeMS toward the HLR.
POST	/api/subscribers/{id}/actions	Execute an action on a subscriber, such as initiating a detach, forcing re-authentication, or triggering a location update.
POST	/api/subscribers/{id}/ss	Manage supplementary services for a subscriber, including activation, deactivation, and interrogation of call barring, call forwarding, and other services.

Calls

Method	Path	Description
GET	/api/calls	List all active CC FSM call transactions, including call reference, direction, parties, state, and duration.
GET	/api/calls/{id}	Retrieve full detail for a single call, including timing, codec, serving BSC/RNC, and CC FSM state history.
DELETE	/api/calls/{id}	Release an active call. Initiates network-side disconnect and BSSMAP CLEAR COMMAND.

SMS

Method	Path	Description
GET	/api/sms	List active SMS transactions with transaction ID, direction, subscriber, state, and SMS Centre address.

Routes

Method	Path	Description
GET	/api/routes	List all entries in the route table, including prefix, destination type, priority, and destination-specific parameters.
POST	/api/routes	Add a new route to the route table. The route takes effect immediately without restart.
DELETE	/api/routes	Delete a route from the route table by prefix and destination type.
GET	/api/routes/lookup	Look up the route that would be selected for a given called party number. Useful for verifying routing behaviour without placing a call.

SIP Peers

Method	Path	Description
GET	/api/sip/peers	List all configured SIP peers with address, transport, status, active calls, and capacity.
GET	/api/sip/peers/{name}	Retrieve full detail for a single SIP peer, including codec list, OPTIONS keepalive state, and call statistics.
PUT	/api/sip/peers/{name}	Update a SIP peer's configuration (address, port, transport, codecs, max channels, OPTIONS interval).

Media Gateways

Method	Path	Description
GET	/api/mgw	List configured media gateways with name, address, protocol (MGCP or Megaco), and reachability status.

RAN and STP

Method	Path	Description
GET	/api/ran/connections	List active RAN connections (SCCP connection-oriented sessions) with connection ID, subscriber IMSI, BSC/RNC, and state.
GET	/api/ran/bscs	List known BSCs with point code, global title, cell count, and last BSSMAP RESET timestamp.
GET	/api/stp	Get STP link status, including M3UA ASP state, SCTP association details, and traffic counters.

Paging

Method	Path	Description
POST	/api/paging	Initiate a paging request for a subscriber by IMSI or MSISDN. Sends BSSMAP PAGING to the appropriate BSCs.
GET	/api/paging	List pending paging requests with subscriber identity, target LAC, paging cause, and elapsed time.

Advice of Charge

Method	Path	Description
POST	/api/aoc	Send an Advice of Charge message to a subscriber during an active call. Supports AoCI (information) and AoCE (charging) per 3GPP TS 24.086.

Silent Call and SMS

Method	Path	Description
POST	/api/silent	Initiate a silent call or silent SMS toward a subscriber. Used for lawful intercept and network testing purposes.

Handover Cells

Method	Path	Description
GET	/api/handover/cells	List cells configured as handover targets, including cell identity, LAC, BSC, and handover priority.
POST	/api/handover/cells	Add a cell to the handover target list with cell identity, LAC, and priority parameters.

System

Method	Path	Description
GET	/api/health	Health check endpoint. Returns a simple status indicator suitable for load balancer probes.
GET	/api/status	System status including BEAM VM statistics, memory allocation, supervisor health, active alarm count, and MSC configuration summary.
GET	/metrics	Prometheus metrics scrape endpoint. Returns all OmniMSC telemetry counters and gauges in Prometheus exposition format. See Metrics and Monitoring.

Response Format

All endpoints return JSON responses. Successful requests return the requested data in a top-level object or array. Error responses include a JSON object with an `error` field containing a human-readable message and, where applicable, a `code` field with a machine-readable error identifier.

List endpoints support pagination via `page` and `page_size` query parameters. The response includes a `meta` object with `total`, `page`, and `page_size` fields.

Subscriber and call detail responses include nested objects for related data (identity, location, authentication, timing) matching the structure described in the Control Panel subscriber detail and call detail sections.

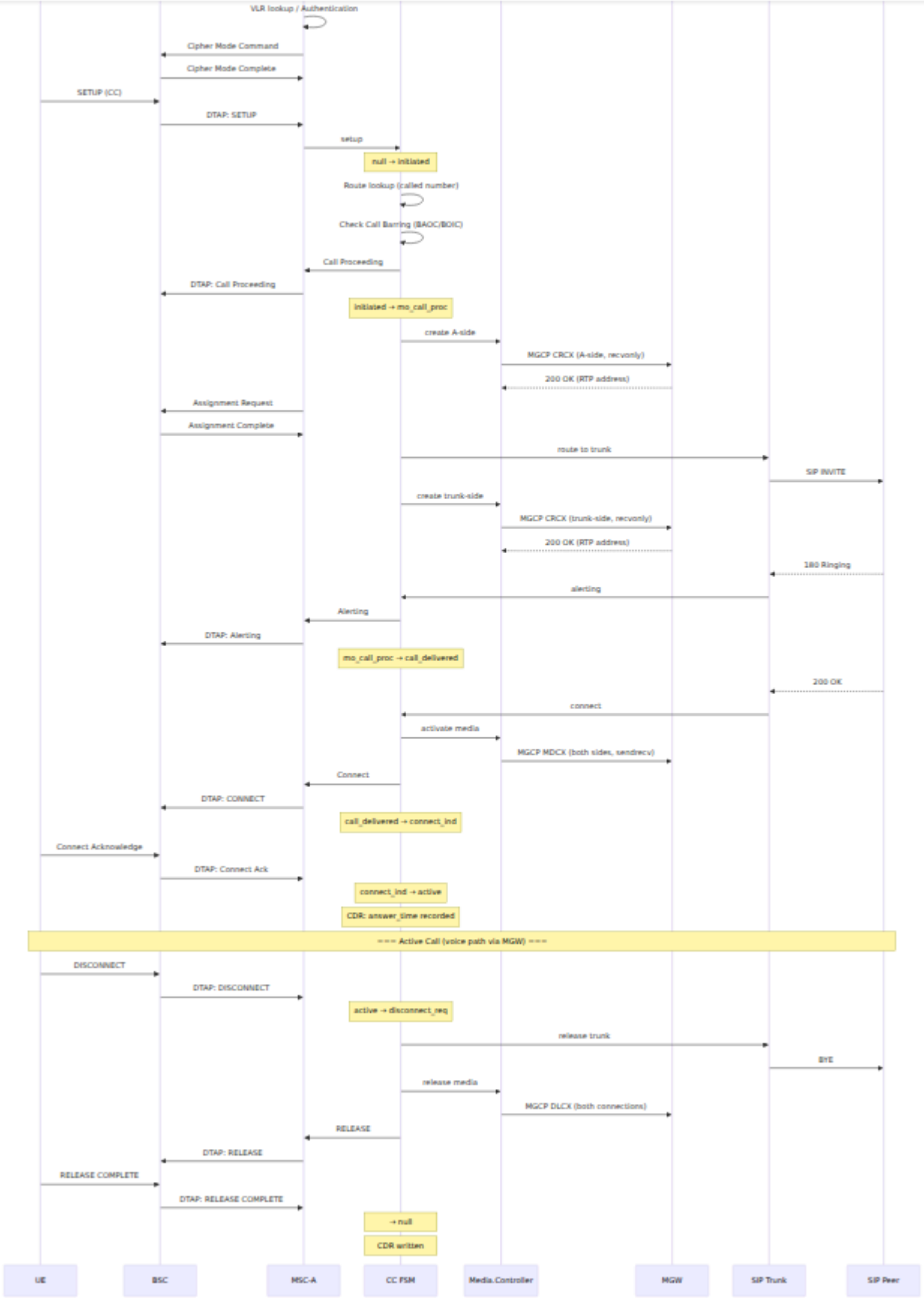
Call Flow Diagrams

This document contains sequence diagrams for all major call scenarios handled by OmniMSC. Each diagram shows the signaling flow between network elements.

For CC FSM state definitions, see Control Panel Guide. For routing configuration, see Routing Configuration. For SIP trunk signaling details, see SIP Trunking. For ISUP trunk signaling, see ISUP Trunking. For media gateway control during call setup, see Media Control. For supplementary service flows (hold, MPTY, ECT), see Supplementary Services.

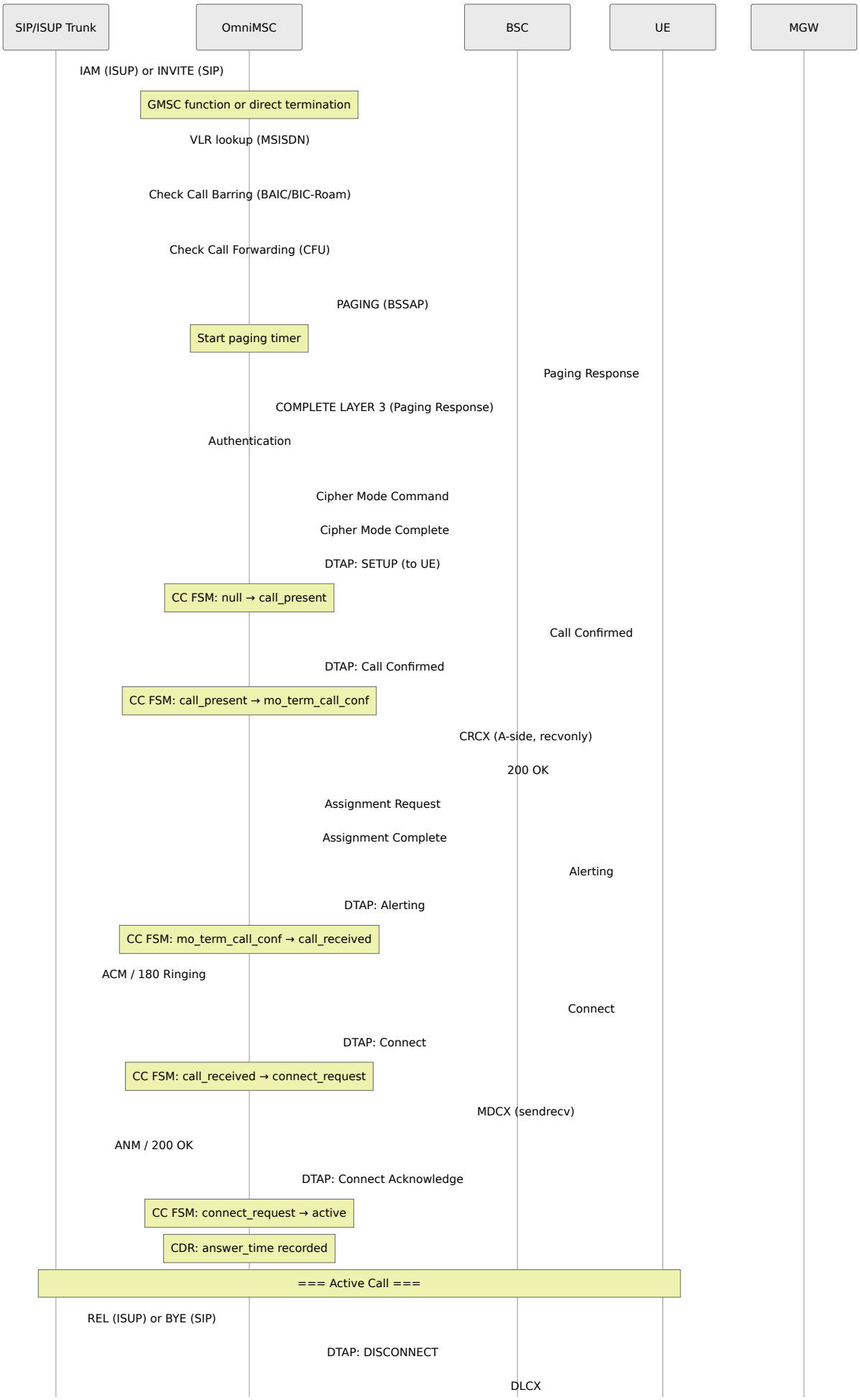
Mobile Originating (MO) Call

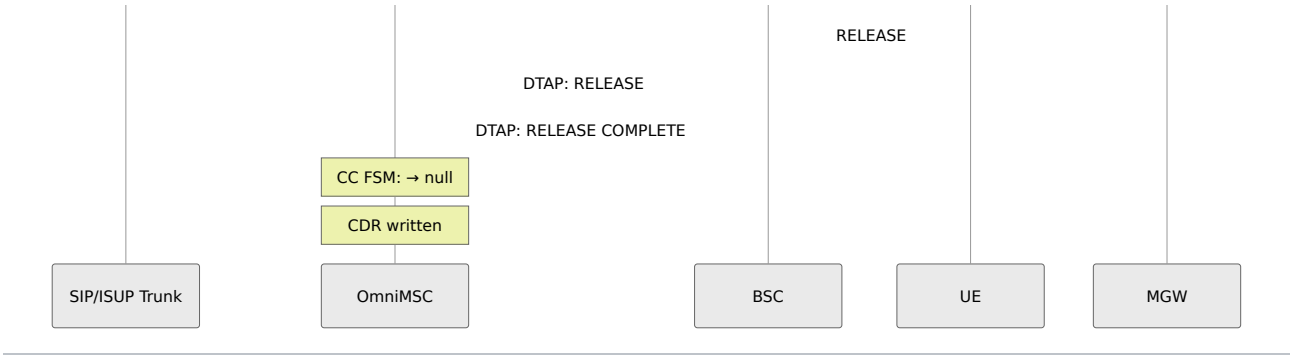
A subscriber initiates an outgoing call. The MSC handles call setup, authentication (if needed), cipher mode, assignment, and trunk allocation.


VLR lookup / Authentication
Cipher Mode Command
Cipher Mode Complete
SETUP (CC)
DTAP: SETUP
setup
null → initiated
Route lookup (called number)
Check Call Barring (BAOC/BOIC)
Call Proceeding
DTAP: Call Proceeding
initiated → mo_call_proc
create A-side
MGCP CRCX (A-side, recvonly)
200 OK (RTP address)
Assignment Request
Assignment Complete
route to trunk
SIP INVITE
create trunk-side
MGCP CRCX (trunk-side, recvonly)
200 OK (RTP address)
180 Ringing
alerting
Alerting
DTAP: Alerting
mo_call_proc → call_delivered
200 OK
connect
activate media
MGCP MDCX (both sides, sendrecv)
Connect
DTAP: CONNECT
call_delivered → connect_ind
Connect Acknowledge
DTAP: Connect Ack
connect_ind → active
CDR: answer_time recorded
=== Active Call (voice path via MGW) ===
DISCONNECT
DTAP: DISCONNECT
active → disconnect_req
release trunk
BYE
release media
MGCP DLCX (both connections)
RELEASE
DTAP: RELEASE
RELEASE COMPLETE
DTAP: RELEASE COMPLETE
→ null
CDR written
UE
BSC
MSC-A
CC FSM
Media Controller
MGW
SIP Trunk
SIP Peer

Mobile Terminating (MT) Call

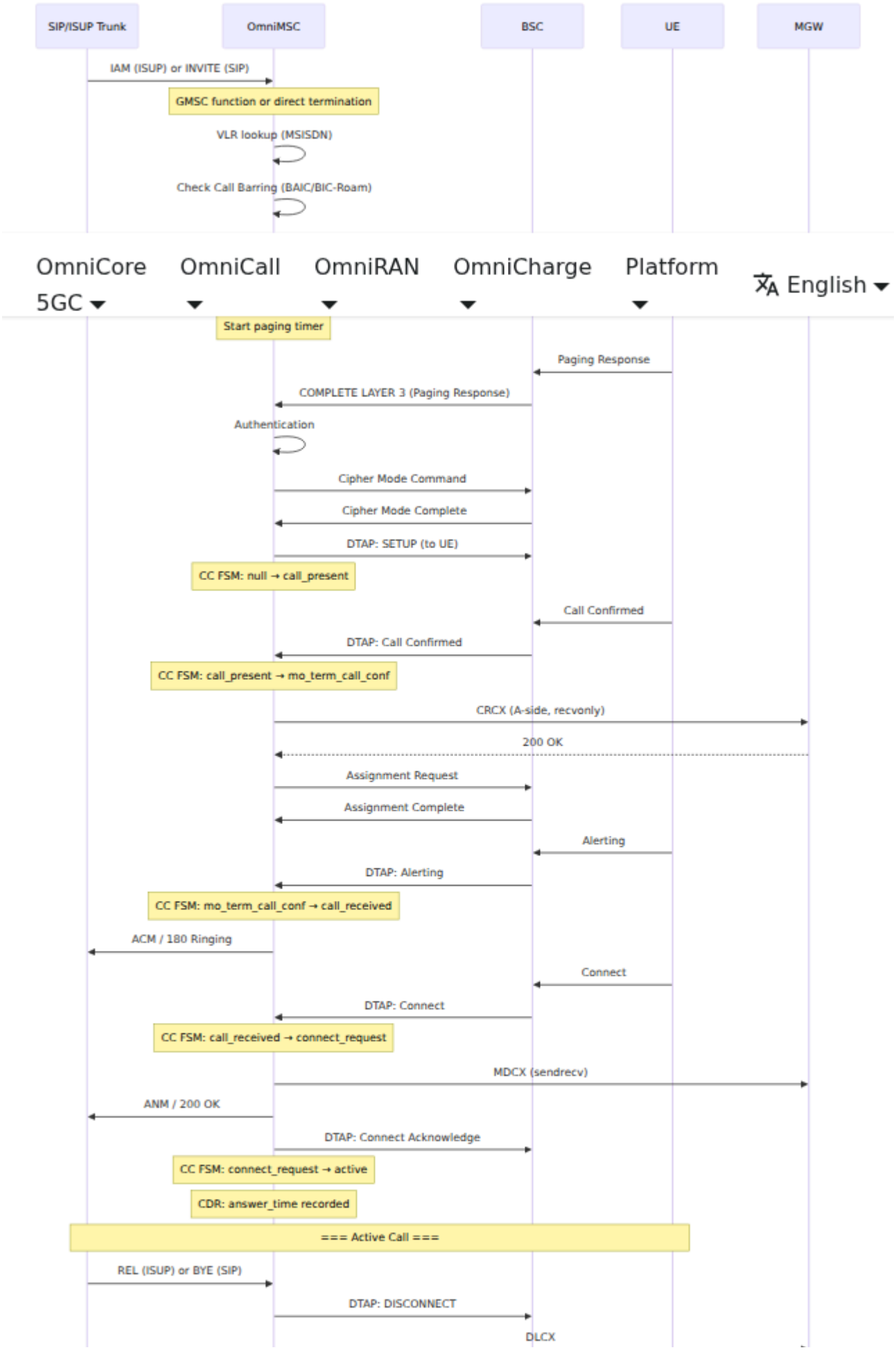
An incoming call arrives from the PSTN or a SIP peer. The MSC pages the subscriber, sets up the radio path, and connects the call. The trunk-side signaling may be SIP or ISUP depending on the route.


SIP/ISUP Trunk
OmniMSC
BSC
UE
MGW
IAM (ISUP) or INVITE (SIP)
GMSC function or direct termination
VLR lookup (MSISDN)
Check Call Barring (BAIC/BIC-Roam)
Check Call Forwarding (CFU)
PAGING (BSSAP)
Start paging timer
Paging Response
COMPLETE LAYER 3 (Paging Response)
Authentication
Cipher Mode Command
Cipher Mode Complete
DTAP: SETUP (to UE)
CC FSM: null → call_present
Call Confirmed
DTAP: Call Confirmed
CC FSM: call_present → mo_term_call_conf
CRCX (A-side, recvonly)
200 OK
Assignment Request
Assignment Complete
Alerting
DTAP: Alerting
CC FSM: mo_term_call_conf → call_received
ACM / 180 Ringing
Connect
DTAP: Connect
CC FSM: call_received → connect_request
MDCX (sendrecv)
ANM / 200 OK
DTAP: Connect Acknowledge
CC FSM: connect_request → active
CDR: answer_time recorded
=== Active Call ===
REL (ISUP) or BYE (SIP)
DTAP: DISCONNECT
DLCX



Call Hold and Retrieve

Call Hold and Retrieve allow a subscriber to suspend and resume an active call per 3GPP TS 24.083. While a call is held, the media gateway places the held leg into receive-only mode, silencing the audio path. The subscriber may establish a second call while the first is held.


SIP/ISUP Trunk
OmniMSC
BSC
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MGW
IAM (ISUP) or INVITE (SIP)
GMSC function or direct termination
VLR lookup (MSISDN)
Check Call Barring (BAIC/BIC-Roam)
OmniCore 5GC
OmniCall
OmniRAN
OmniCharge
Platform
English
Start paging timer
Paging Response
COMPLETE LAYER 3 (Paging Response)
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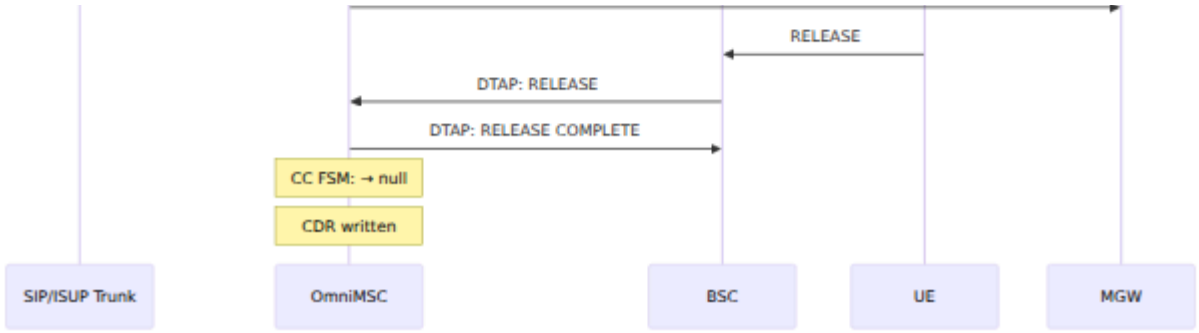
Cipher Mode Complete
DTAP: SETUP (to UE)
CC FSM: null → call_present
Call Confirmed
DTAP: Call Confirmed
CC FSM: call_present → mo_term_call_conf
CRCX (A-side, recvonly)
200 OK
Assignment Request
Assignment Complete
Alerting
DTAP: Alerting
CC FSM: mo_term_call_conf → call_received
ACM / 180 Ringing
Connect
DTAP: Connect
CC FSM: call_received → connect_request
MDCX (sendrecv)
ANM / 200 OK
DTAP: Connect Acknowledge
CC FSM: connect_request → active
CDR: answer_time recorded
=== Active Call ===
REL (ISUP) or BYE (SIP)
DTAP: DISCONNECT
DLCX



The HOLD message from the UE contains no parameters. The MSC responds with HOLD ACK on success or HOLD REJECT if the operation is not permitted (for example, if the subscriber does not have Call Hold provisioned). RETRIEVE follows the same pattern with RETRIEVE ACK or RETRIEVE REJECT.

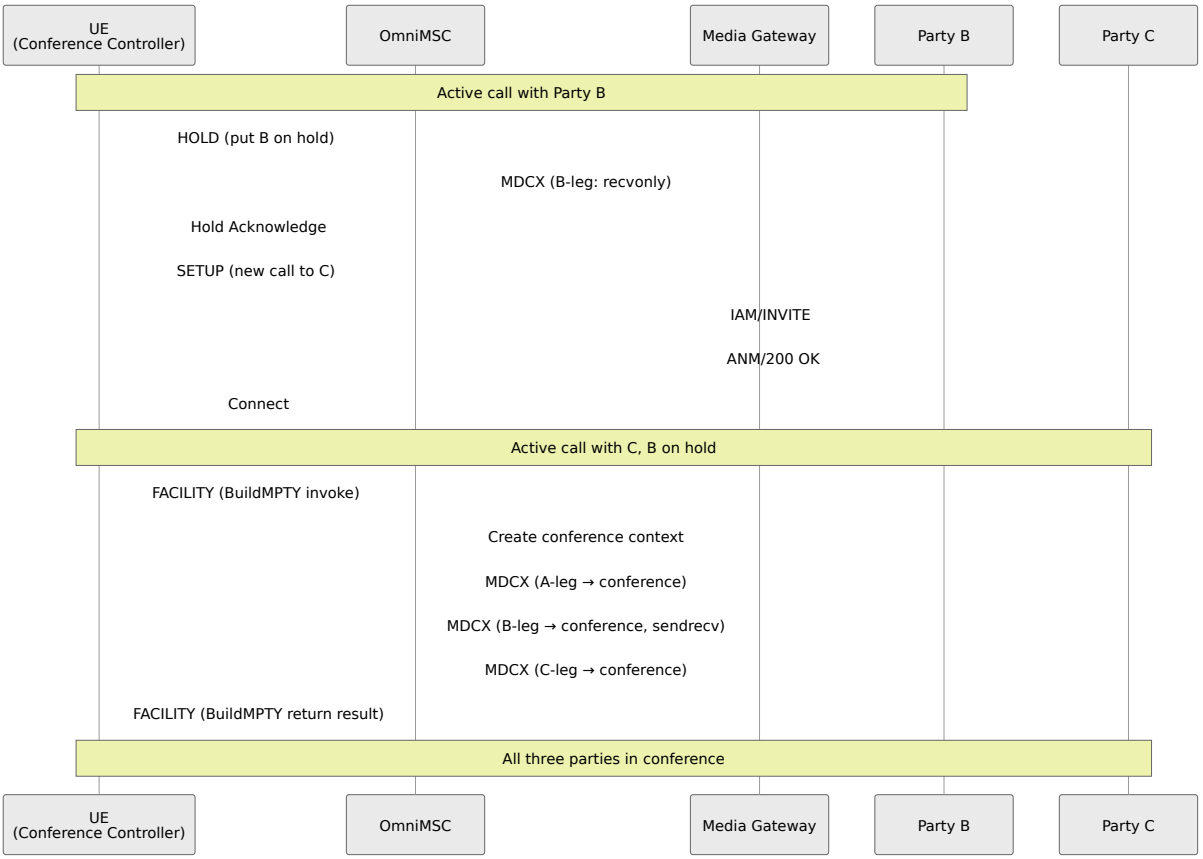
Multi-Party Conference (MPTY)

MPTY allows a subscriber to bridge multiple calls into a conference per 3GPP TS 24.084. The subscriber acts as the conference controller. All conference audio is mixed through a conference bridge on the media gateway.

For configuration, see Advanced Call Features.

BuildMPTY

The subscriber establishes two calls (one active, one held), then invokes BuildMPTY via a CC FACILITY message to merge them into a conference.



HoldMPTY, RetrieveMPTY, SplitMPTY

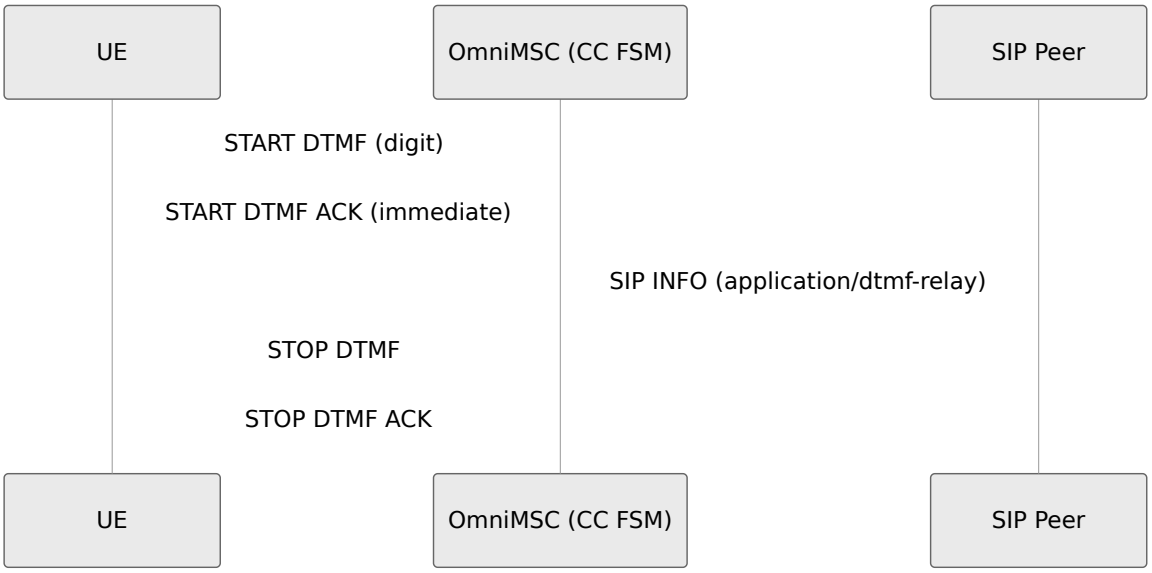
Once a conference is established, the controller has three additional operations available:

Operation	Effect
HoldMPTY	Places the entire conference on hold. All remote parties hear silence. The controller may establish a private call outside the conference.
RetrieveMPTY	Resumes a held conference. All parties are reconnected to the conference bridge.
SplitMPTY	Extracts one party from the conference into a separate held call. The remaining parties continue in conference. If only two parties remain after the split, the conference context is released and the call reverts to a normal point-to-point connection.

Each operation is invoked by the UE via a CC FACILITY message and acknowledged by the MSC with a corresponding return result or error.

DTMF Relay

OmniMSC relays DTMF tones from the radio interface to the trunk side. The UE sends a START DTMF message (3GPP TS 24.008) containing the digit. The CC FSM immediately acknowledges the UE and forwards the digit to the SIP peer.



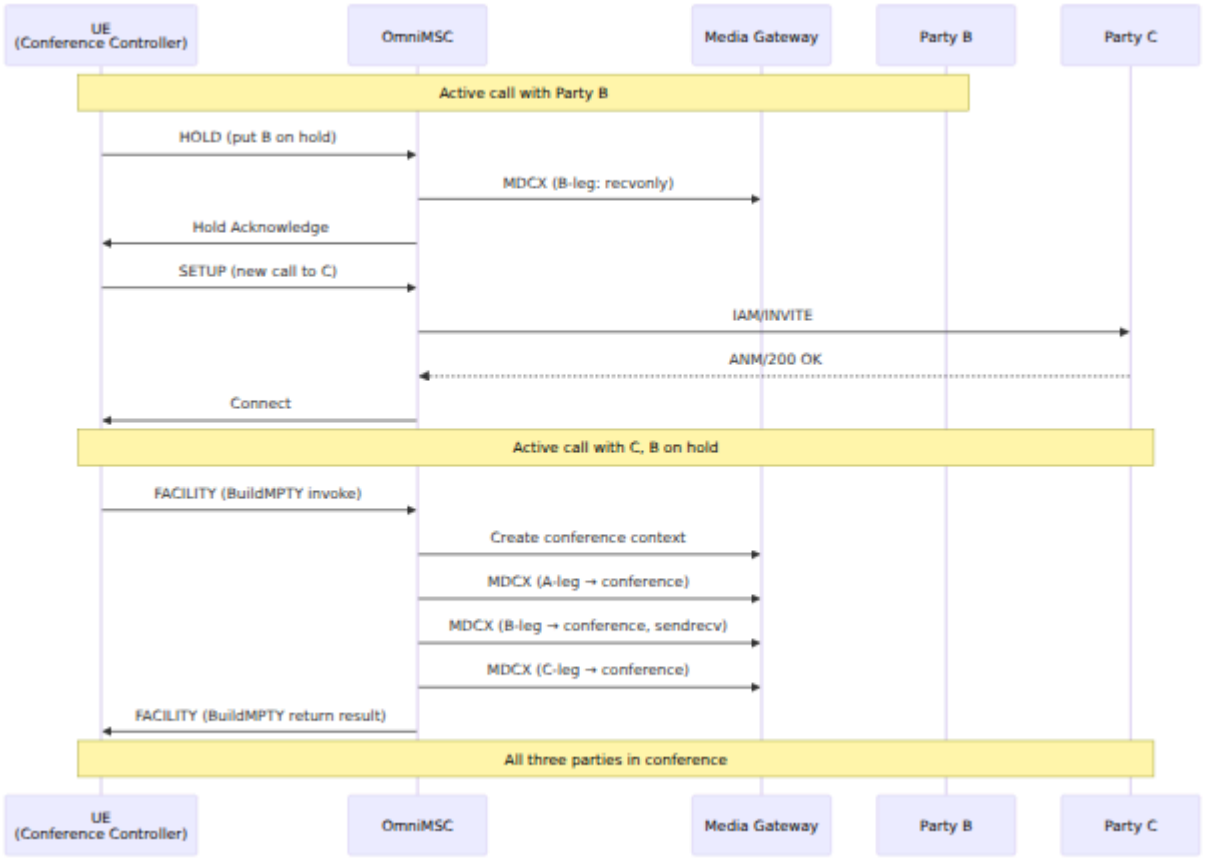
The SIP INFO body uses the `application/dtmf-relay` content type with Signal and Duration fields. The MSC sends START DTMF ACK immediately to the UE without waiting for the SIP peer's response, avoiding audio clipping on the radio path. DTMF relay is handled in any call state where a trunk-side connection exists.

Emergency Calls

Emergency calls receive priority handling. The MSC detects emergency calls by the CC Emergency Setup message type (3GPP TS 24.008 §9.3.8, message type 0x0E) and the CM Service Request type (`:emergency`).

Authentication is attempted normally. If it succeeds, cipher mode is established and the subscriber's MSISDN is used as the calling party. If authentication fails, the call proceeds anyway — cipher mode is skipped and the IMEI is used as the calling party instead. Calls from UEs without a SIM are accepted.

Unlike regular CC Setup, the Emergency Setup message does not carry a Called Party BCD Number IE — the handset sends only an optional Bearer Capability and Emergency Service Category. The MSC uses the configured `psap_address` as the called number for route table lookup and the outgoing SIP INVITE Request-URI. See Configuration Reference for emergency parameters.



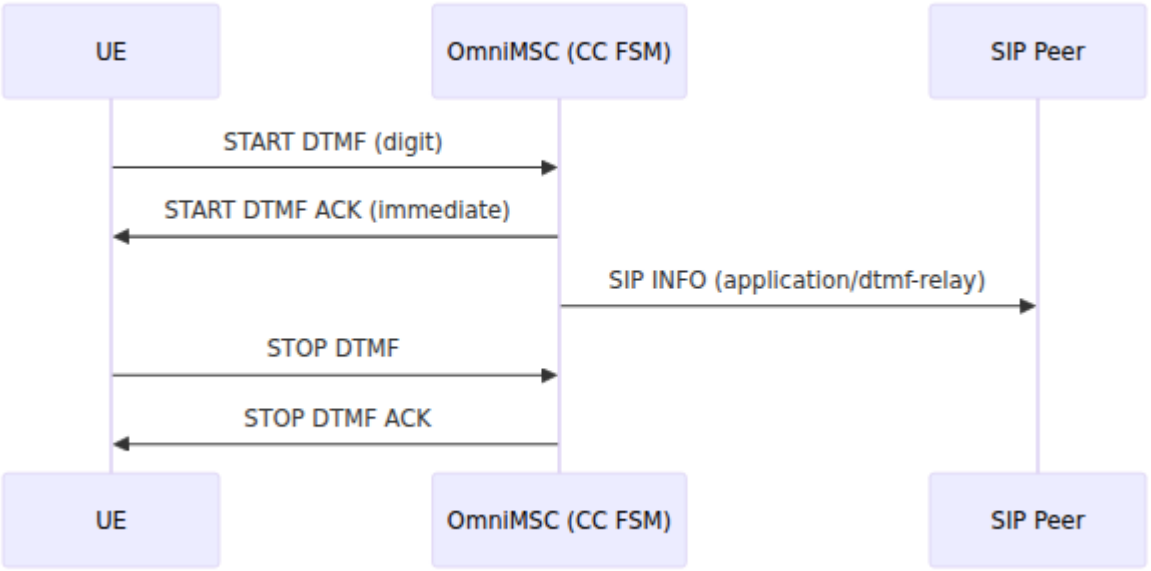
CC FSM State Diagram

The CC FSM (Call Control Finite State Machine) manages the lifecycle of a single circuit-switched call per 3GPP TS 24.008. The MO and MT paths share common states but enter through different transitions. Live call state, duration, codec, and serving BSC/RNC are visible in the Active Calls page of the control panel — see Control Panel Guide.

MO Call States


null
MS SETUP received
initiated
Call Proceeding sent
mo_call_proc
Alerting (remote ringing)
call_delivered
Connect sent to MS
RELEASE COMPLETE
connect_ind
Connect Ack from MS
active
MS DISCONNECT
Network release
disconnect_req
disconnect_ind
RELEASE sent
RELEASE sent
release_req

MT Call States



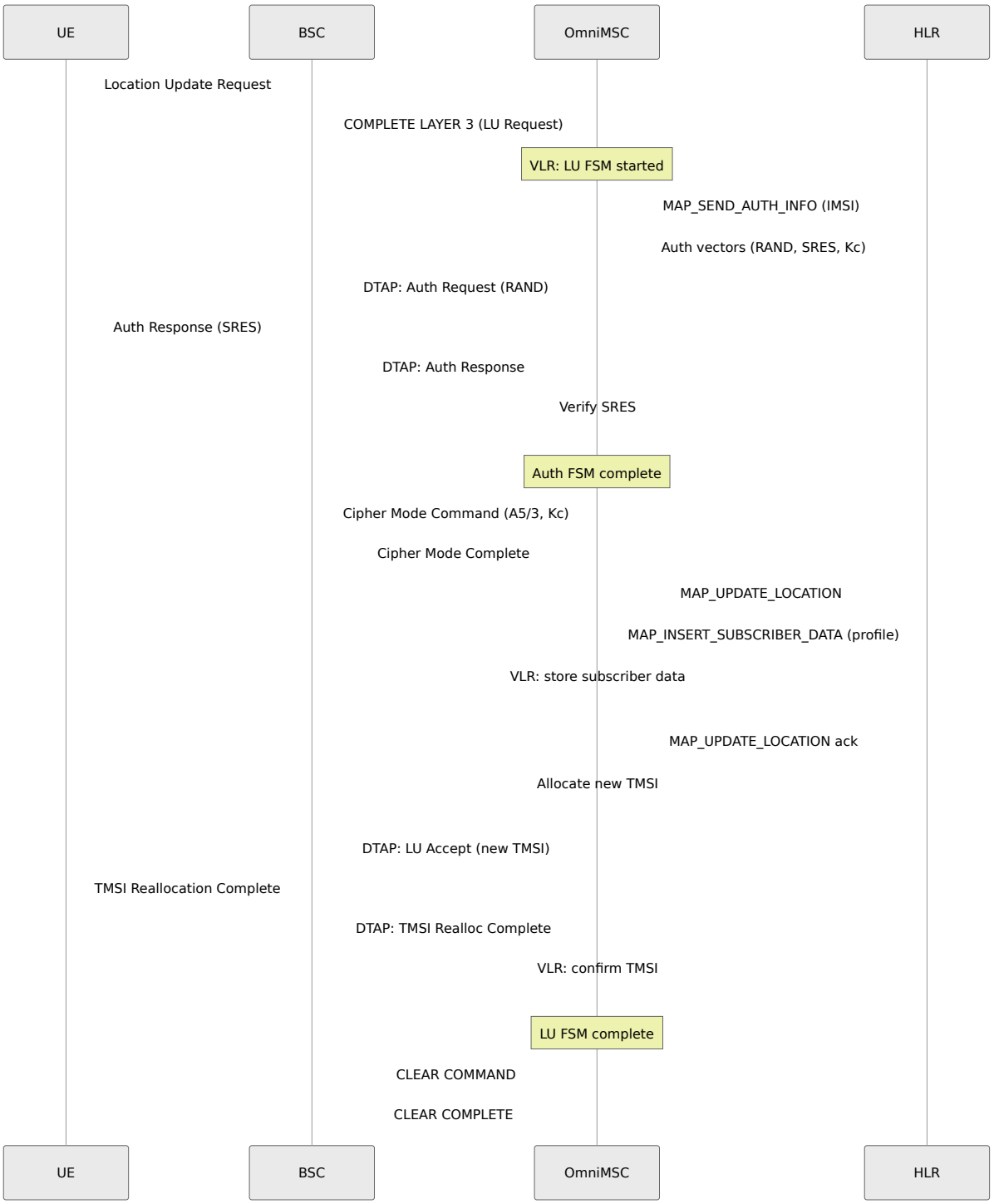
Connection Release

When the A-interface connection is lost (radio link failure, BSC clear, or transport failure), MSC-A sends a `connection_lost` event to all CC FSMs associated with that subscriber. Each CC FSM receiving this event releases its trunk-side resources (SIP BYE or ISUP REL) and media gateway connections (MGCP DLCX), then transitions to the null state and writes the CDR.

This ensures that no trunk or media resources are leaked when the radio path is unexpectedly lost. The CC FSM handles `connection_lost` in any state except null.

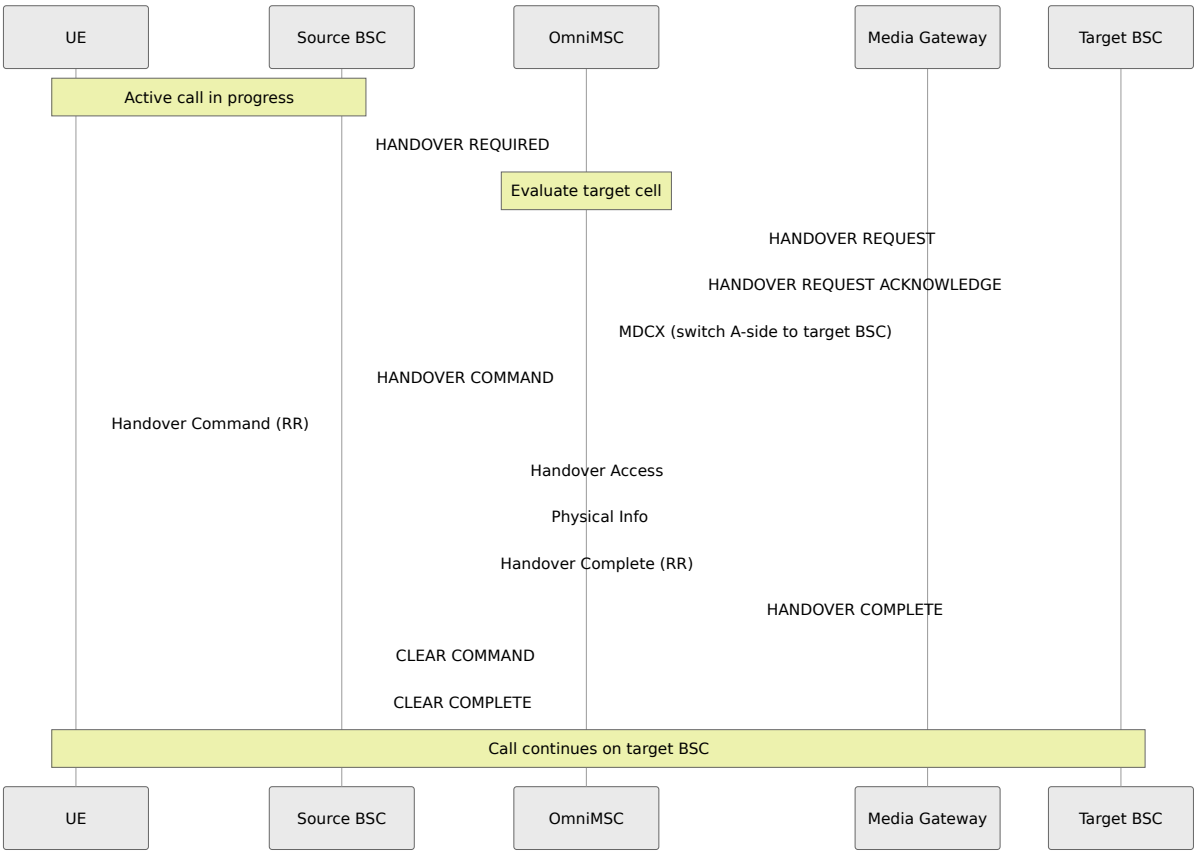
Location Update

Subscriber registers with the MSC. The MSC authenticates the subscriber and updates the HLR.



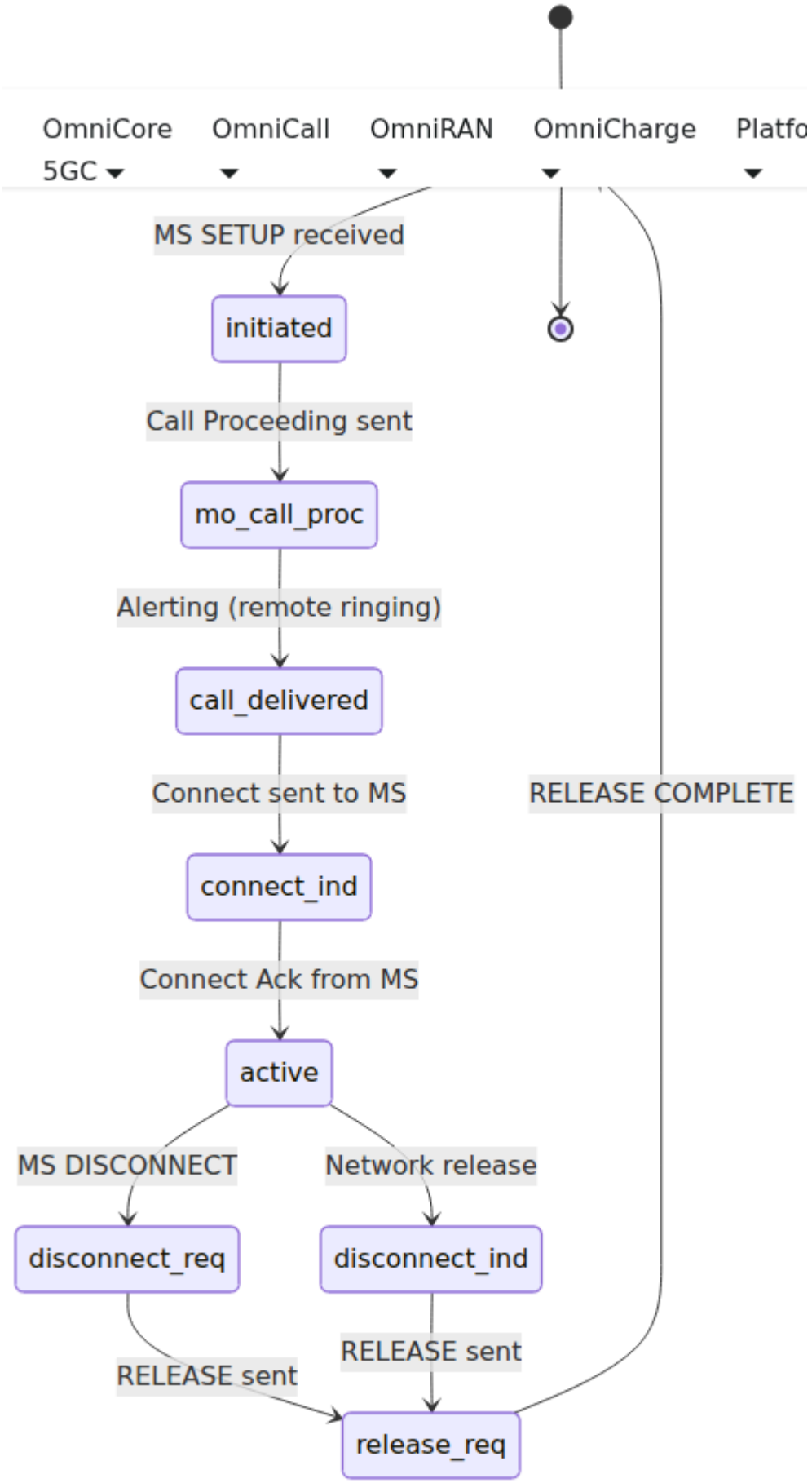
Intra-MSC Handover

Handover of an active call between two BSCs served by the same MSC.



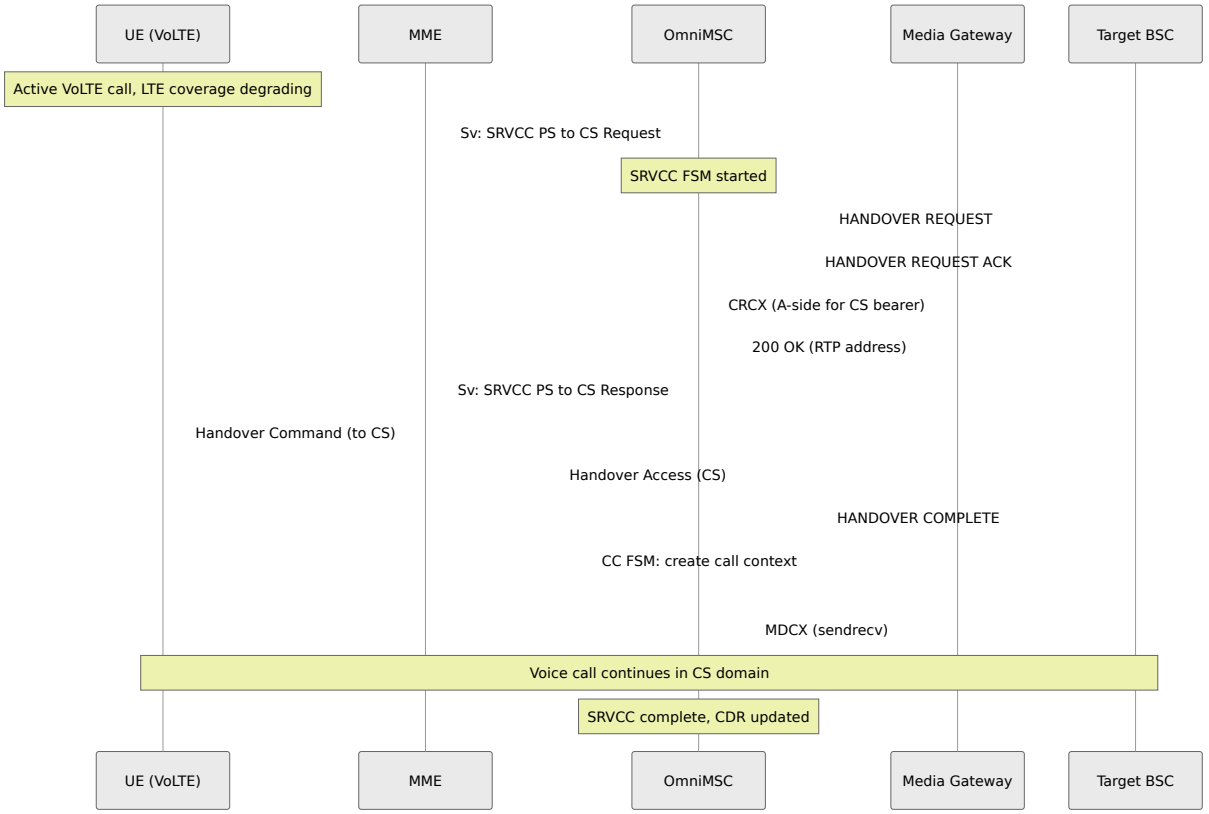
Inter-MSC Handover

Handover of an active call from OmniMSC (MSC-A) to a target MSC (MSC-B).


OmniCore
5GC
OmniCall
OmniRAN
OmniCharge
Platfo
MS SETUP received
initiated
Call Proceeding sent
mo_call_proc
Alerting (remote ringing)
call_delivered
Connect sent to MS
connect_ind
Connect Ack from MS
active
MS DISCONNECT
Network release
disconnect_req
disconnect_ind
RELEASE sent
RELEASE sent
release_req
RELEASE COMPLETE

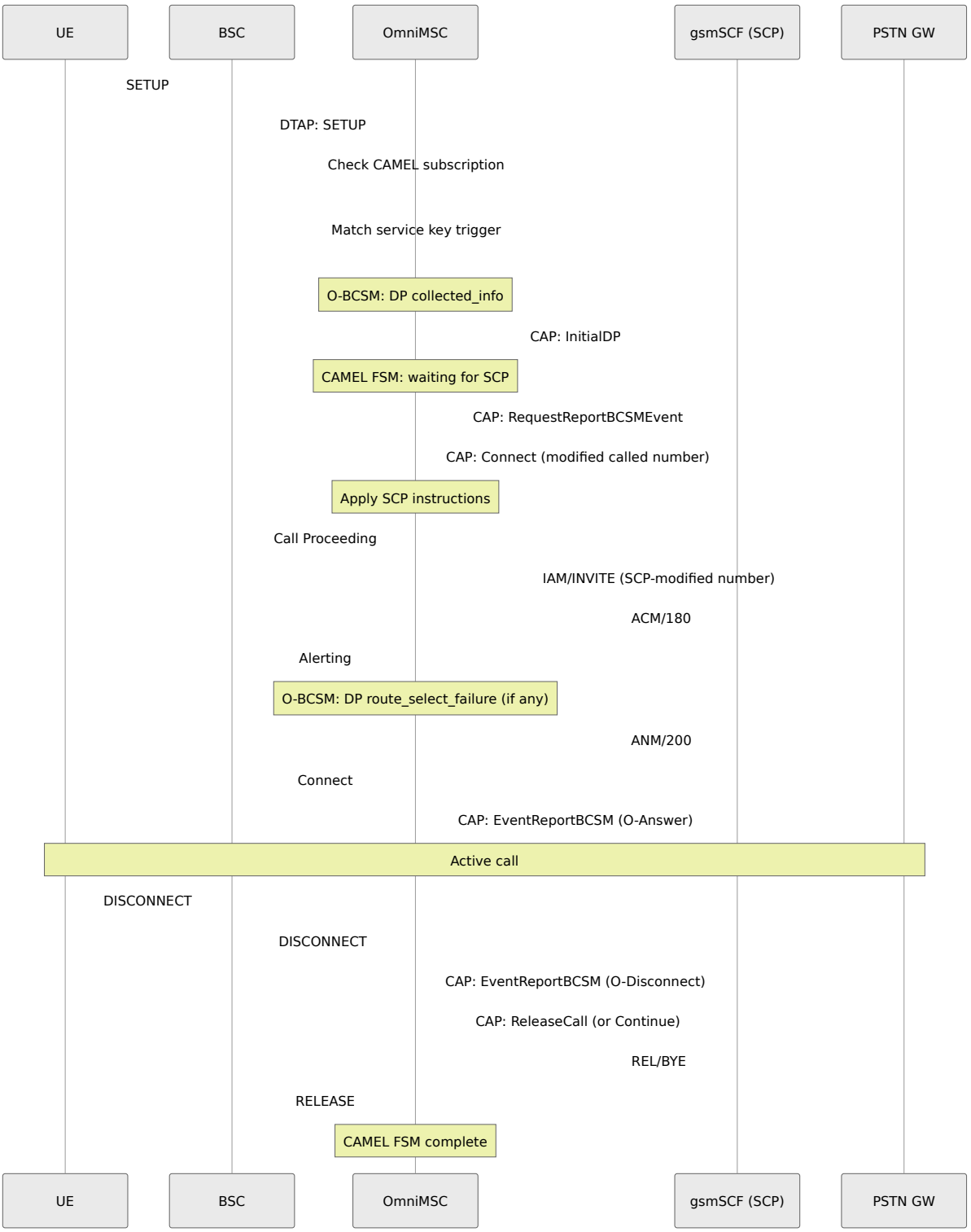
SRVCC (Single Radio Voice Call Continuity)

Handover of a VoLTE call from the IMS/LTE domain to the CS domain per 3GPP TS 23.216.



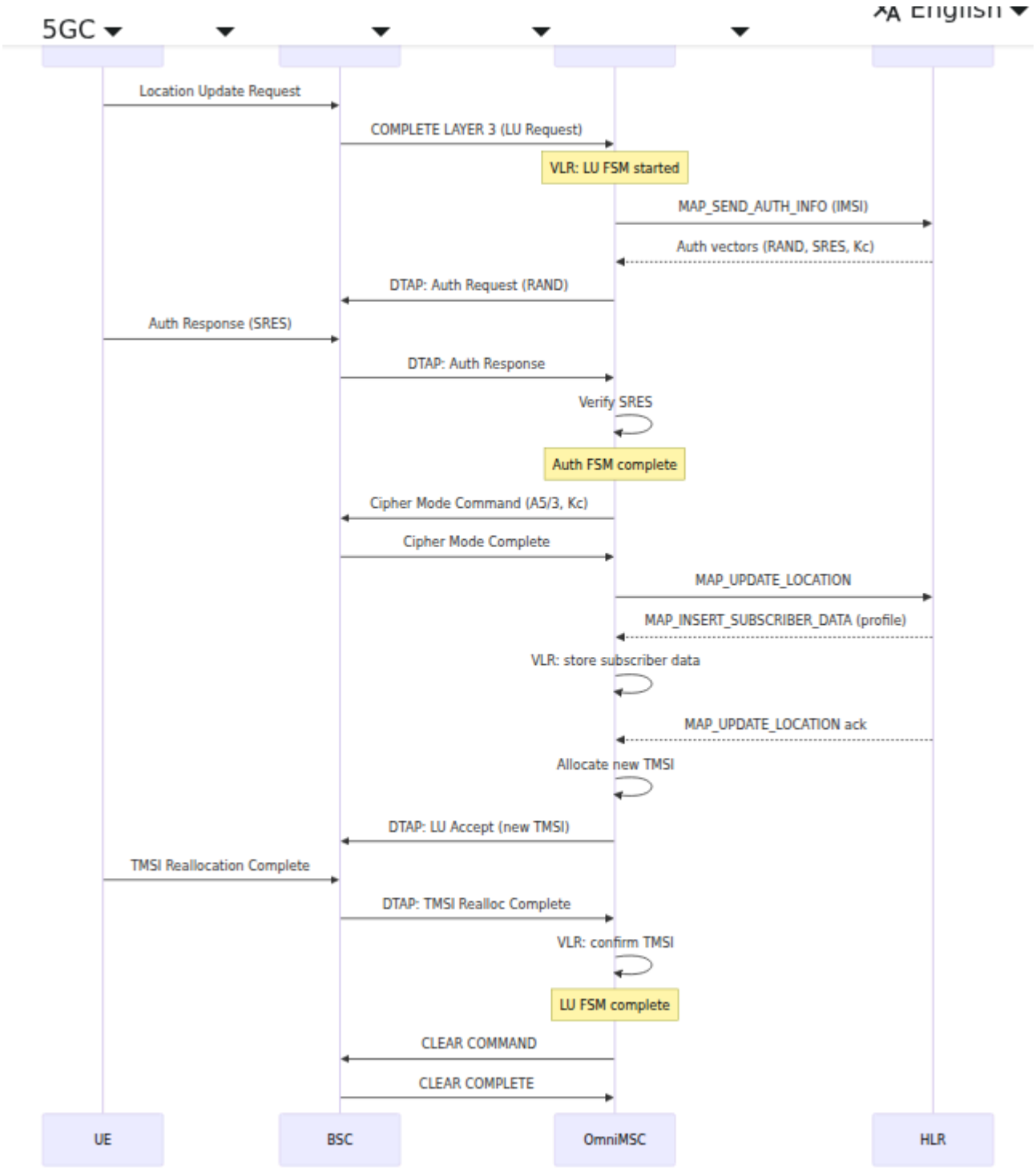
CAMEL-Triggered Call (SCP Interaction)

Call with CAMEL Originating BCSM (O-BCSM) interaction per 3GPP TS 23.078.


UE
BSC
OmniMSC
gsmSCF (SCP)
PSTN GW
SETUP
DTAP: SETUP
Check CAMEL subscription
Match service key trigger
O-BCSM: DP collected_info
CAP: InitialDP
CAMEL FSM: waiting for SCP
CAP: RequestReportBCSMEvent
CAP: Connect (modified called number)
Apply SCP instructions
Call Proceeding
IAM/INVITE (SCP-modified number)
ACM/180
Alerting
O-BCSM: DP route_select_failure (if any)
ANM/200
Connect
CAP: EventReportBCSM (O-Answer)
Active call
DISCONNECT
DISCONNECT
CAP: EventReportBCSM (O-Disconnect)
CAP: ReleaseCall (or Continue)
REL/BYE
RELEASE
CAMEL FSM complete
UE
BSC
OmniMSC
gsmSCF (SCP)
PSTN GW

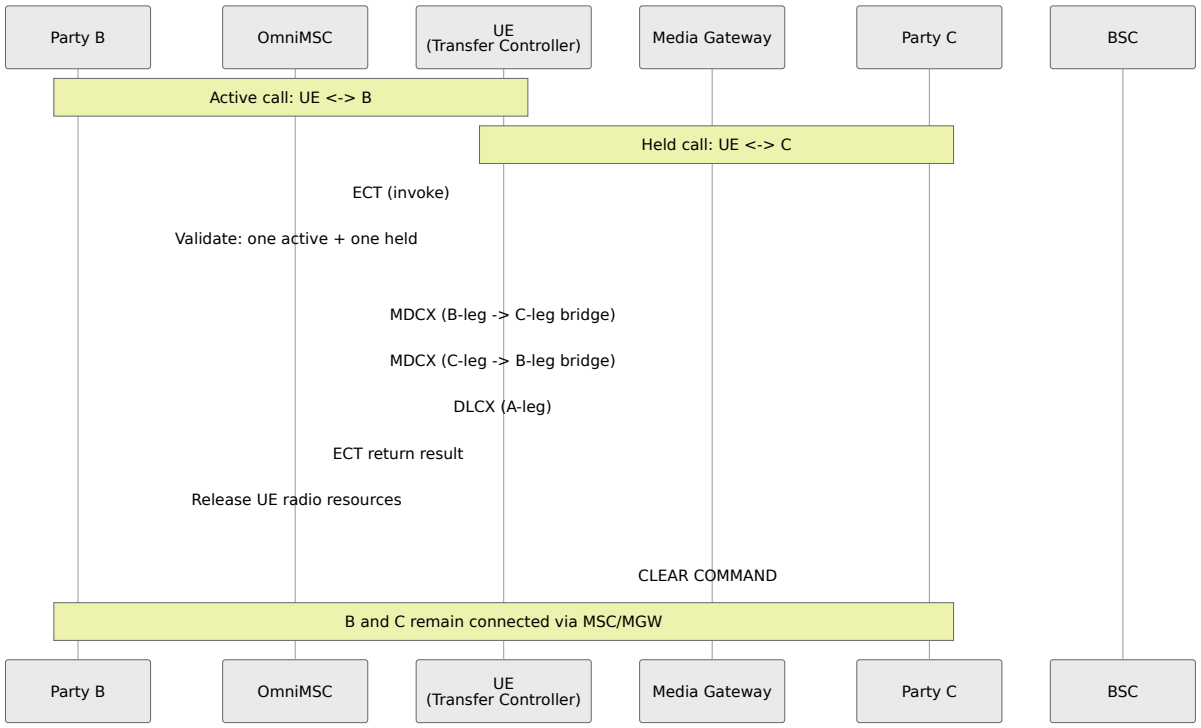
MPTY (BuildMPTY) Sequence Diagram

Multi-party conference call setup. The subscriber establishes two calls, then bridges them into a conference per 3GPP TS 24.084. For configuration, see Advanced Call Features.



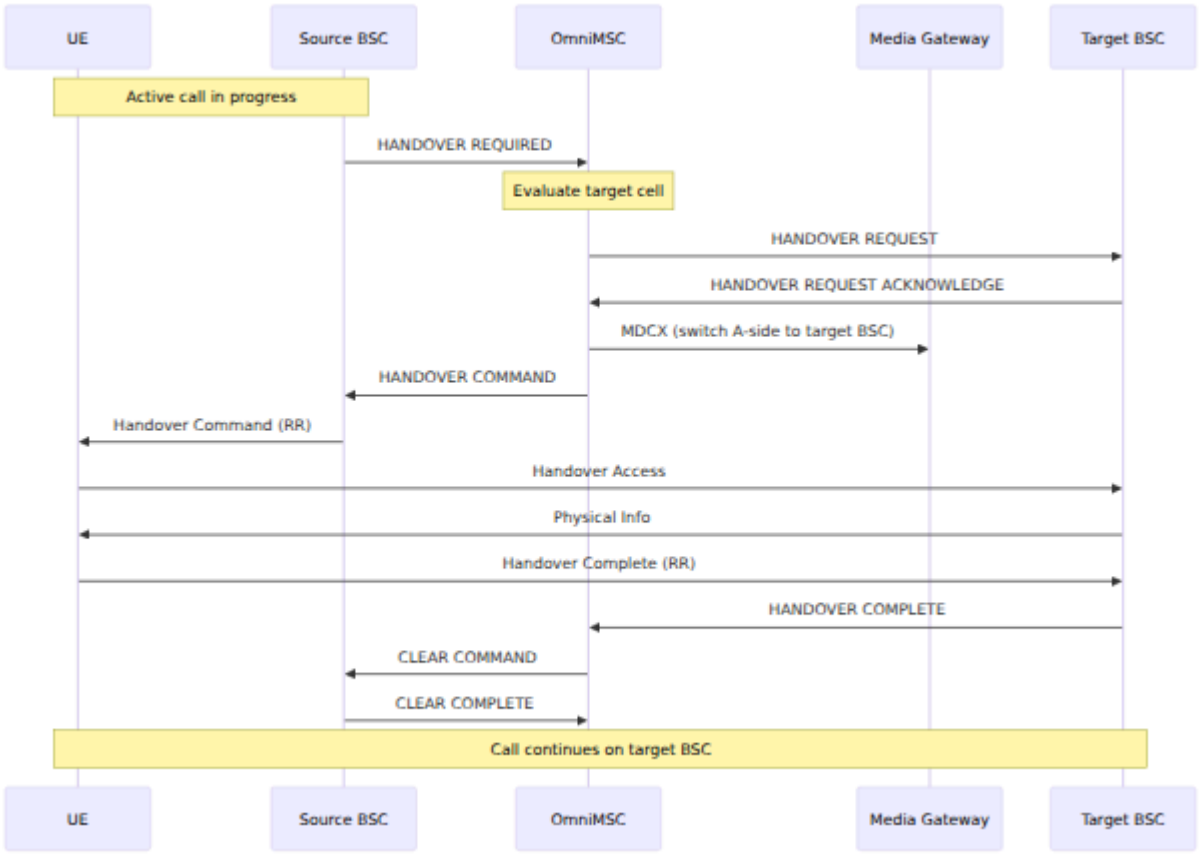
ECT (Explicit Call Transfer) Sequence Diagram

Explicit Call Transfer connects two remote parties and releases the transferring subscriber per 3GPP TS 24.091. For configuration, see Advanced Call Features.



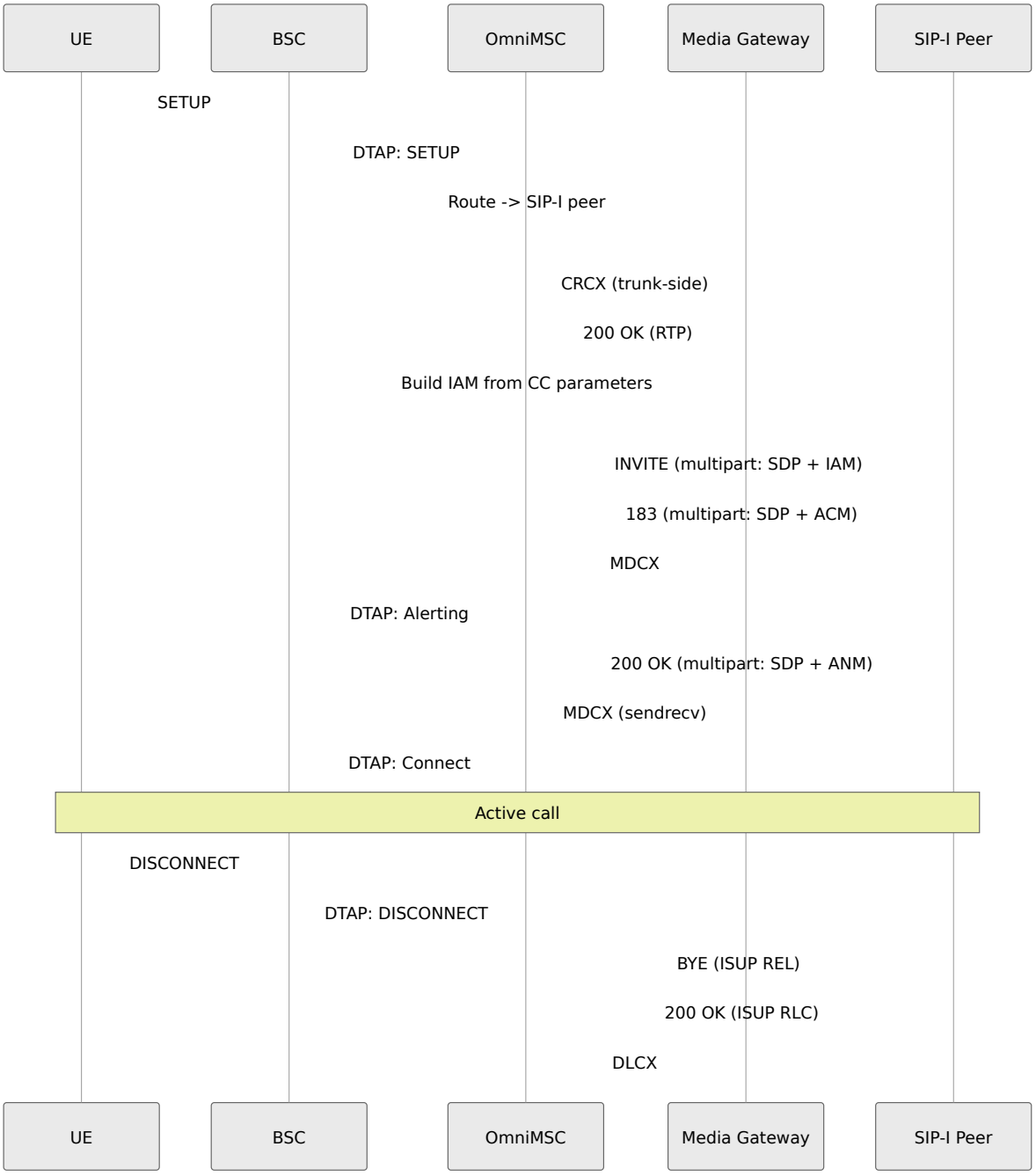
CSFB MT Call Sequence Diagram

Mobile-terminating call to an LTE-attached subscriber via Circuit-Switched Fallback per 3GPP TS 23.272. The MSC pages via the SGs interface, the UE falls back to 2G/3G, and the call proceeds over the A-interface. For SGs details, see SGs Interface & CSFB.



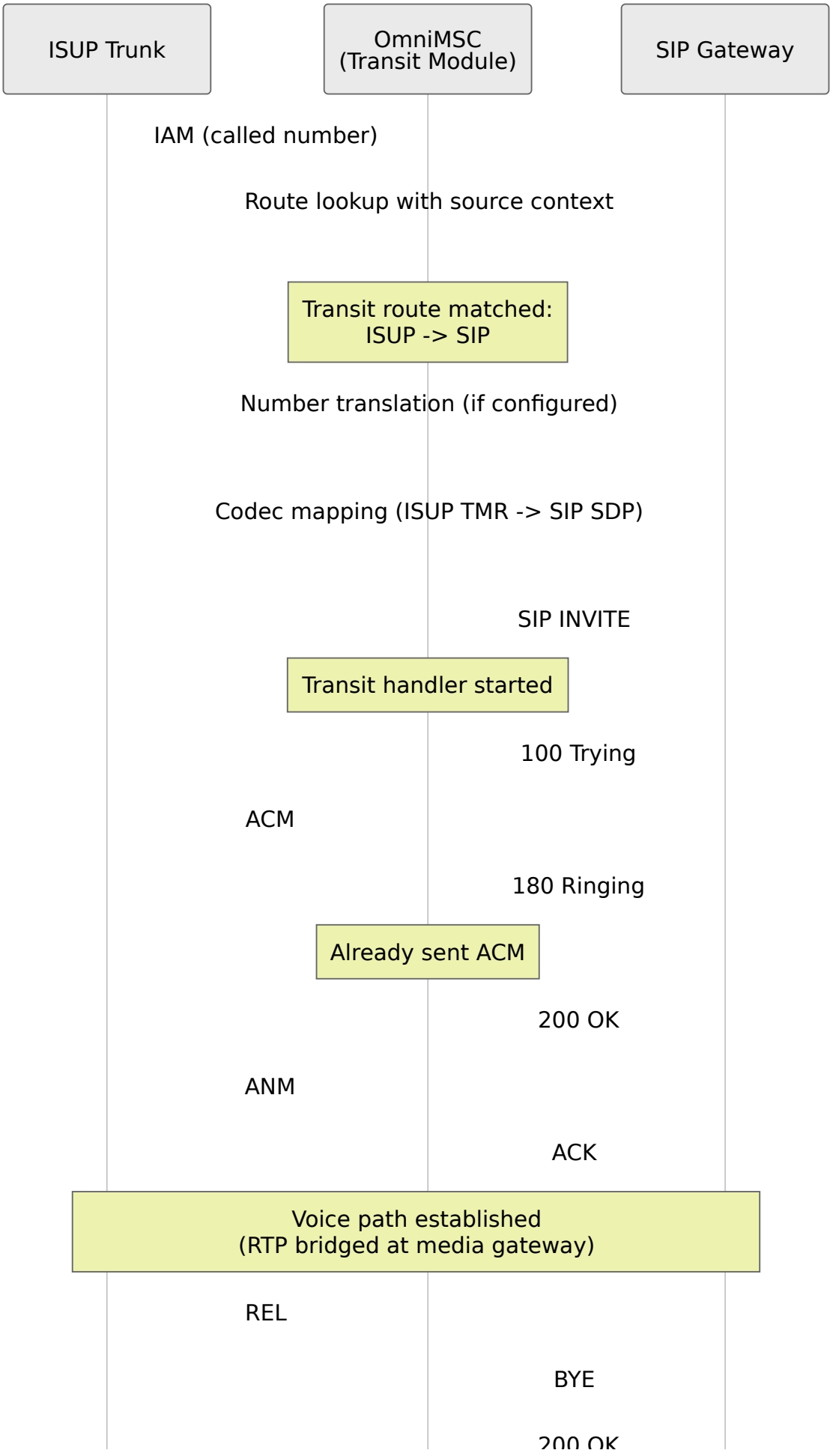
SIP-I Call Sequence Diagram

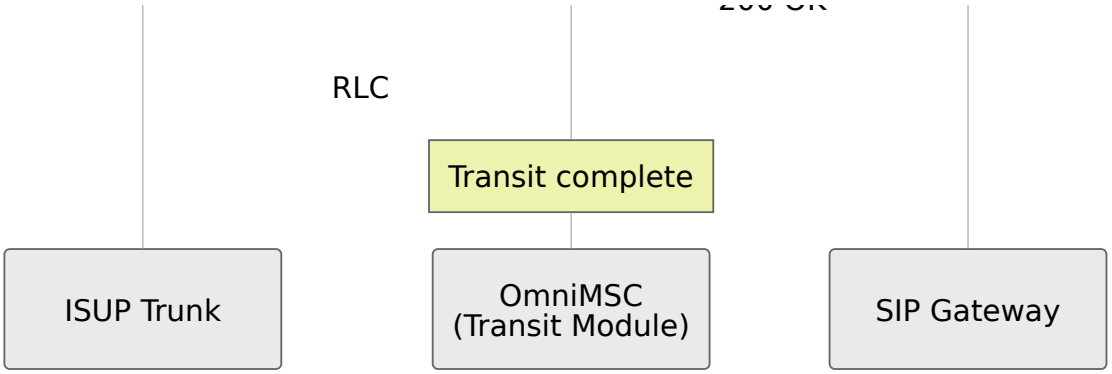
Outgoing call via SIP-I trunk with encapsulated ISUP per ITU-T Q.1912.5. The INVITE carries a multipart body containing SDP and the ISUP IAM. For SIP-I details, see SIP-I Trunking.



ISUP-to-SIP Transit Call

Transit call interworking between an ISUP trunk and a SIP peer, bypassing the CC FSM.


ISUP Trunk
OmniMSC
(Transit Module)
SIP Gateway
IAM (called number)
Route lookup with source context
Transit route matched:
ISUP -> SIP
Number translation (if configured)
Codec mapping (ISUP TMR -> SIP SDP)
SIP INVITE
Transit handler started
100 Trying
ACM
180 Ringing
Already sent ACM
200 OK
ANM
ACK
Voice path established
(RTP bridged at media gateway)
REL
BYE
200 OK


200 OK
RLC
Transit complete
ISUP Trunk
OmniMSC
(Transit Module)
SIP Gateway

CAMEL / CAP

This document describes the CAMEL (Customized Applications for Mobile Enhanced Logic) implementation in OmniMSC, including service key configuration, CAP dialogue handling, BCSM detection points, supported CAP operations, and TCAP transport.

For the full CAMEL-triggered call sequence diagram (InitialDP, Connect, EventReportBCSM), see Call Flow Diagrams. For how CAMEL triggers fit into the routing pipeline (number analysis flow), see Routing Configuration. For CAMEL-related CDR fields (FurnishChargingInformation, cause_for_term), see Call Detail Records. For configuration of CAMEL service keys, see Configuration Reference.

Overview

CAMEL provides a framework for mobile operators to deploy Intelligent Network (IN) services on top of the GSM/UMTS core network. OmniMSC acts as the gsmSSF (GSM Service Switching Function), interacting with an external gsmSCF (GSM Service Control Function, also known as the SCP) to provide real-time call control services such as prepaid charging, number translation, call screening, and virtual private networks.

The gsmSCF controls call processing by sending instructions to OmniMSC at defined detection points within the Basic Call State Model (BCSM). OmniMSC reports call events back to the gsmSCF and executes the received instructions (continue, connect to a different number, release the call, apply charging).

Service Key Configuration

Each subscriber may have one or more CAMEL subscriptions provisioned via MAP INSERT SUBSCRIBER DATA from the HLR. A CAMEL subscription includes a

service key, which identifies the IN service to invoke, and the gsmSCF address (Global Title) to contact.

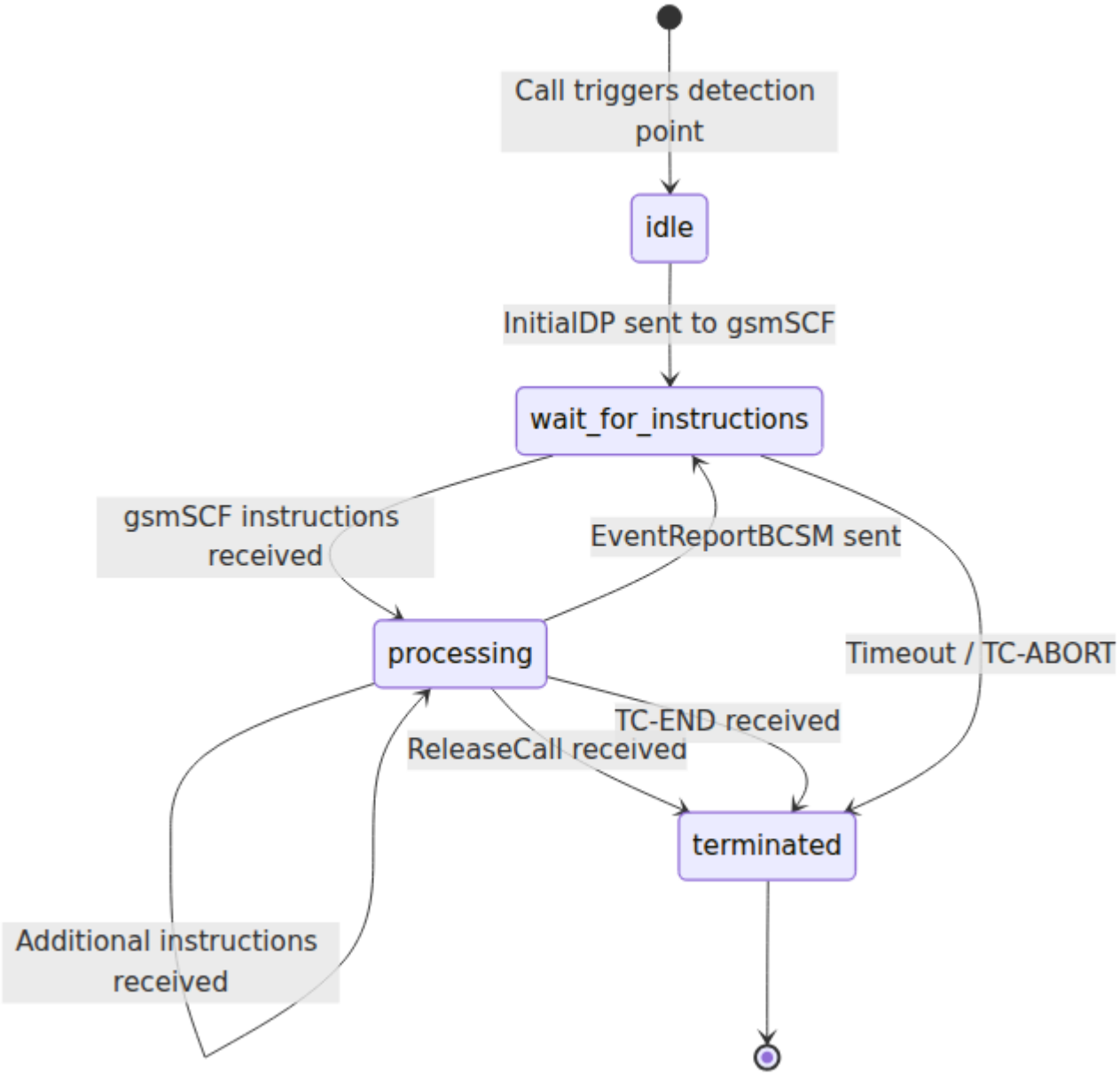
When a call triggers a CAMEL detection point (such as collected_info for MO calls or terminating_attempt_authorized for MT calls), OmniMSC checks the subscriber's CAMEL subscription data. If a matching service key is found, OmniMSC opens a CAP dialogue with the gsmSCF and sends an InitialDP operation.

Subscription Parameter	Description
Service key	Integer identifying the IN service (e.g., 1 for prepaid, 2 for VPN)
gsmSCF address	Global Title of the SCP to contact
Default call handling	Action if the SCP is unreachable: `:continue_call` or `:release_call`
TDP list	List of trigger detection points armed for this subscription
CAMEL phase	Supported CAMEL phase (Phase 1, 2, 3, or 4)

If the gsmSCF is unreachable or the TCAP dialogue fails, OmniMSC applies the default call handling action from the subscriber's CAMEL subscription data.

CAP Dialogue States

Each CAMEL interaction runs as an independent CAP dialogue within a TCAP transaction. The dialogue tracks the state of the SSF-SCF interaction from InitialDP through to termination.



State	Description
idle	Detection point triggered, preparing InitialDP
wait_for_instructions	InitialDP sent, awaiting gsmSCF response
processing	Executing gsmSCF instructions (Continue, Connect, ApplyCharging)
terminated	Dialogue complete, TCAP transaction closed

CAP Operations

OmniMSC supports the following CAP operations for CAMEL Phase 2 and Phase 3 service interaction.

SSF to SCF (OmniMSC to gsmSCF)

Operation	Description
InitialDP	Reports a triggered detection point with call parameters (service key, called/calling number, event type, location information)
EventReportBCSM	Reports a call event at an armed detection point (answer, disconnect, abandon, route select failure)
ApplyChargingReport	Reports the result of a charging operation (call duration, charge units consumed)
CallInformationReport	Reports call information requested by the gsmSCF (call duration, release cause)

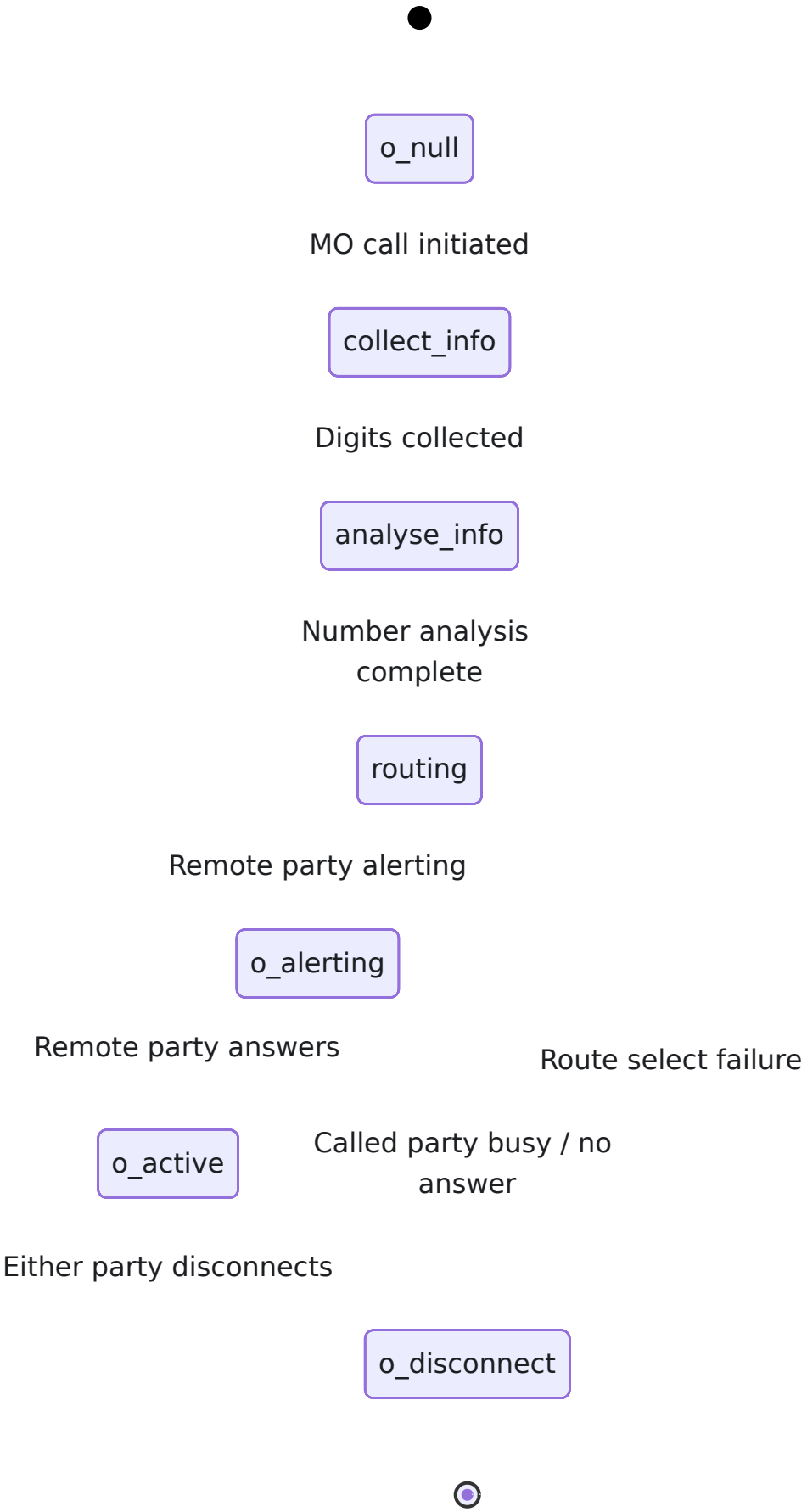
SCF to SSF (gsmSCF to OmniMSC)

Operation	Description
Continue	Resume call processing at the current BCSM state
Connect	Route the call to a different destination number (number translation, VPN routing)
ReleaseCall	Release the call with a specified cause code
RequestReportBCSMEvent	Arm detection points for future event reporting (answer, disconnect, abandon)
ApplyCharging	Apply charging parameters (maximum call duration, charge advice)
FurnishChargingInformation	Provide free-format charging data to be included in the CDR
ResetTimer	Reset the SSF inactivity timer to prevent timeout during long SCP processing
SendChargingInformation	Send advice-of-charge information to the mobile station
CallInformationRequest	Request call information to be reported at call release

---

O-BCSM Detection Points

The Originating Basic Call State Model defines the detection points available for MO calls per 3GPP TS 23.078.

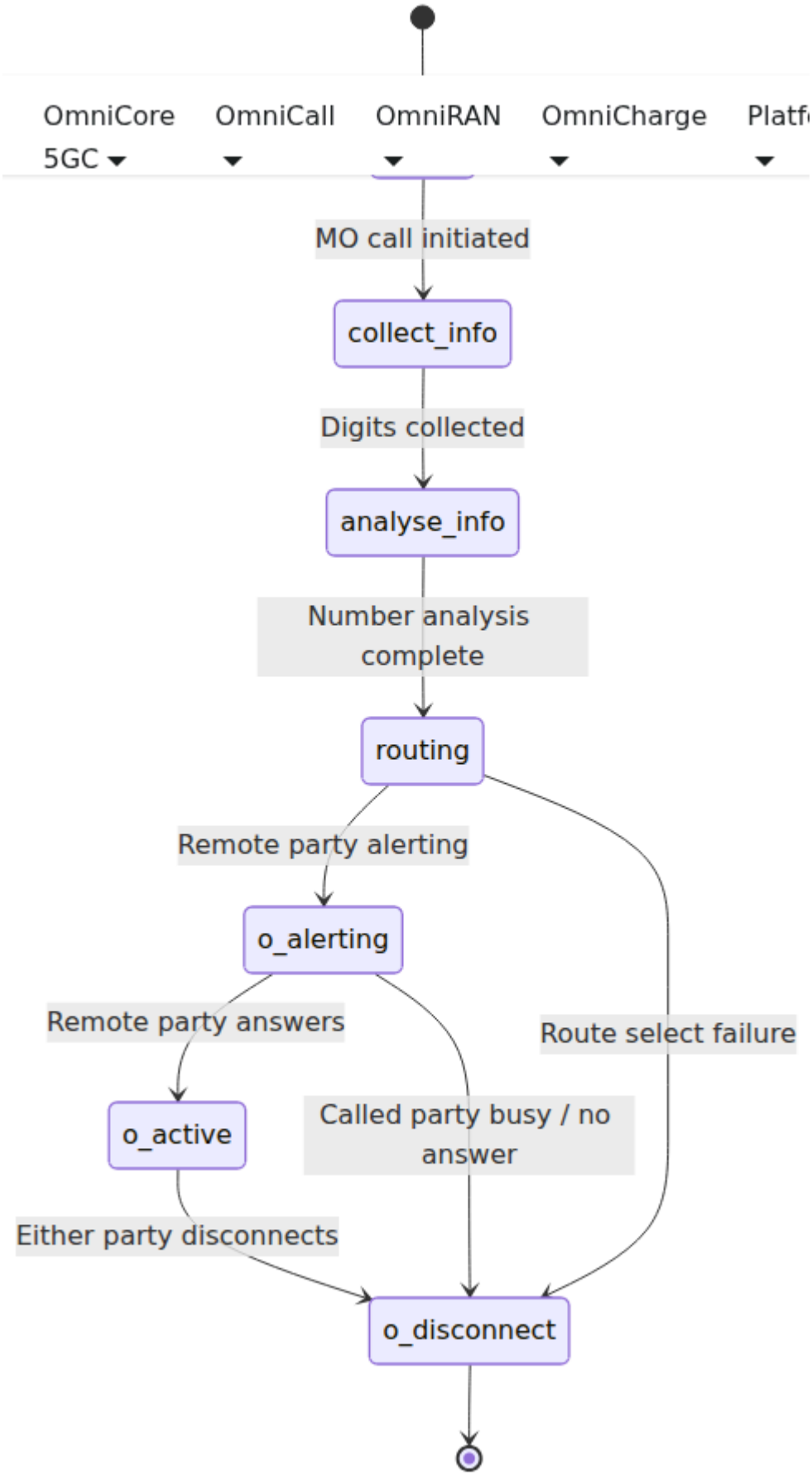

o_null
MO call initiated
collect_info
Digits collected
analyse_info
Number analysis complete
routing
Remote party alerting
o_alerting
Remote party answers
Route select failure
o_active
Called party busy / no answer
Either party disconnects
o_disconnect

Detection Point	BCSM State	Trigger
collected_info (DP 2)	collect_info	Dialed digits available, before number analysis
analysed_info (DP 3)	analyse_info	Number analysis complete, before routing
route_select_failure (DP 4)	routing	Routing fails (no route, trunk busy, peer down)
o_called_party_busy (DP 5)	o_alerting	Called party busy
o_no_answer (DP 6)	o_alerting	Called party does not answer within timer
o_answer (DP 7)	o_active	Called party answers
o_disconnect (DP 9)	o_disconnect	Either party initiates disconnect

---

T-BCSM Detection Points

The Terminating Basic Call State Model defines the detection points available for MT calls.


OmniCore 5GC
OmniCall
OmniRAN
OmniCharge
Platf
MO call initiated
collect_info
Digits collected
analyse_info
Number analysis complete
routing
Remote party alerting
o_alerting
Remote party answers
Route select failure
o_active
Called party busy / no answer
Either party disconnects
o_disconnect

Detection Point	BCSM State	Trigge
terminating_attempt_authorized (DP 12)	terminating_attempt_authorized	MT call received before paging
t_busy (DP 13)	t_alerting	Called subscrib busy
t_no_answer (DP 14)	t_alerting	Called subscrib does no answer within timer
t_answer (DP 15)	t_active	Called subscrib answers
t_disconnect (DP 17)	t_disconnect	Either party initiates disconn

---

TCAP/CAP Transport

CAP operations are carried over TCAP (Transaction Capabilities Application Part) dialogues, which in turn use the SCCP/M3UA/SCTP transport stack. OmniMSC uses a generic BER decoder via the TcapDecoder module to parse incoming TCAP and CAP PDUs.

Transport Parameter	Value
SSN (local, SSF)	146
SSN (remote, SCF)	148
TCAP dialogue type	Structured (TC-BEGIN, TC-CONTINUE, TC-END)
Encoding	ASN.1 BER (Basic Encoding Rules)
Application context	CAP v2: 0.4.0.0.1.0.50.0, CAP v3: 0.4.0.0.1.0.50.1

The gsmSCF address is a Global Title, routed via SCCP Global Title Translation to the SCP node. OmniMSC maintains a TCAP transaction for the lifetime of each CAP dialogue, using TC-CONTINUE to exchange operations within the dialogue and TC-END to close it.

References

Reference	Title	Relevance
3GPP TS 23.078	CAMEL Phase 3 -- Stage 2	CAMEL architecture, BCSM model, detection points
3GPP TS 29.078	CAMEL Application Part (CAP) -- Stage 3	CAP protocol encoding, ASN.1 definitions
ITU-T Q.771-Q.775	Transaction Capabilities	TCAP dialogue and transaction handling
ITU-T Q.711-Q.716	SCCP	Signaling connection for TCAP transport
3GPP TS 22.078	CAMEL -- Service Description	CAMEL service requirements

Call Detail Records

This document describes the Call Detail Record (CDR) subsystem in OmniMSC by Omnitouch. CDRs are generated in compliance with 3GPP TS 32.298 and provide the charging and audit trail for all circuit-switched services handled by the MSC.

For CDR-related configuration parameters, see Configuration Reference. For the CDR Statistics page in the web interface, see Control Panel Guide. For CDR-related Prometheus metrics and alarm events, see Metrics and Monitoring.

Overview

OmniMSC generates CDRs for voice calls, SMS transactions, location updates, and roaming events. Each CDR captures the subscriber identity, service details, timestamps, location, and termination cause for a single transaction. CDRs are collected in memory, buffered, and periodically flushed to files in ASN.1 BER format following the record structures defined in 3GPP TS 32.298.

The CDR subsystem consists of two components:

- **CDR Collector** -- receives events from the Call Control FSM, VLR location update procedures, and SMS handlers. It correlates events for active calls (setup, alerting, answer, release) into complete CDR records and manages per-record-type sequence numbers.
- **CDR Writer** -- writes encoded CDR records to files on disk, handling file rotation based on size, record count, and time interval.

Record Types

OmniMSC supports the following CDR record types, each identified by a unique ASN.1 tag per TS 32.298.

ASN.1 Tag	Record Type	Description
0	MOCallRecord	Mobile Originated voice call. Generated when a subscriber originates a voice call.
1	MTCallRecord	Mobile Terminated voice call. Generated when a subscriber receives a voice call.
5	MOSMSRecord	Mobile Originated SMS. Generated when a subscriber sends a short message.
6	MTSMSRecord	Mobile Terminated SMS. Generated when a subscriber receives a short message.
13	LocUpdateHLRRecord	Location Update (HLR). Generated for location update procedures at the HLR level, tracking MSC/VLR changes.
14	LocUpdateVLRRecord	Location Update (VLR). Generated for location update procedures at the VLR level, tracking location area changes, authentication results, and TMSI allocation.
17	RoamingRecord	Roaming event. Generated for inter-MSC roaming events.

CDR Fields

Voice Call Records (MOCallRecord and

MTCallRecord)

Field	Description
served_imsi	IMSI of the subscriber who originated or received the call
served_msisdn	MSISDN (phone number) of the served subscriber
served_imei	IMEI of the mobile equipment used
calling_number	Calling party number (A-number)
called_number	Called party number (B-number), present in MO records
connected_number	Number of the party actually connected (may differ from called number due to forwarding)
recording_entity	Address of the MSC that generated the CDR
msc_address	E.164 address of the MSC
msc_incoming_tkgp	Incoming trunk group name
msc_outgoing_tkgp	Outgoing trunk group name
location	Subscriber location at time of call, including Location Area Code (LAC) and Cell Identity (CI)
basic_service	Bearer service or teleservice code identifying the service type
seizure_time	UTC timestamp when the call was initiated (setup message received)

Field	Description
answer_time	UTC timestamp when the call was answered (may be nil for unanswered calls)
release_time	UTC timestamp when the call was released
call_duration	Duration of the call in seconds, measured from answer to release. Zero for unanswered calls.
radio_chan_used	Radio channel type used (full rate or half rate)
cause_for_term	Reason for call termination (see Cause for Termination below)
diagnostics	Diagnostic information: GSM 04.08 cause code, MAP error code, or network-specific cause
call_reference	Unique call reference number
sequence_number	Per-record-type sequence number for gap detection by downstream billing systems
ms_classmark	Mobile station classmark information
system_type	Access network type: GERAN, UTRAN, or unknown
partial_record_type	For partial CDRs: indicates whether this is an intermediate or last partial record

SMS Records (MOSMSRecord and MTSMSRecord)

Field	Description
served_imsi	IMSI of the subscriber
served_msisdn	MSISDN of the subscriber
served_imei	IMEI of the mobile equipment
service_centre	Address of the SMS Service Centre
recording_entity	Address of the recording MSC
location	Subscriber location (LAC/CI)
message_reference	SMS message reference number (MO only)
destination_number	SMS destination number (MO only)
originating_number	SMS sender number (MT only)
origination_time	Timestamp of SMS origination (MO) or delivery (MT)
sms_result	Result of SMS delivery: success, delivery failure, or forwarded

Location Update Records

Field	Description
served_imsi	IMSI of the subscriber
served_msisdn	MSISDN of the subscriber (VLR records only)
recording_entity	Address of the recording entity
update_time	UTC timestamp of the location update
update_type	Type of update: normal location update, periodic location update, IMSI attach, or IMSI detach
old_location / new_location	Previous and new location information (VLR records: LAC/CI)
old_msc / new_msc	Previous and new MSC addresses (HLR records)
old_vlr / new_vlr	Previous and new VLR addresses (HLR records)
vlr_result / hlr_result	Outcome of the location update procedure
authentication_result	Authentication outcome: success, failure (no vectors), failure (auth mismatch), or not performed (VLR records only)
tmsi_allocated	New TMSI value if one was allocated during the procedure (VLR records only)

Cause for Termination

The cause_for_term field records why a call was terminated. The following values are defined per TS 32.298.

Cause	Integer Value	Description
normal_release	0	Normal call clearing by either party
partial_record	1	Partial CDR generated for a long-duration call (intermediate record)
partial_record_call_reestablishment	2	Partial record due to call re-establishment
unsuccessful_call_attempt	3	Call setup failed before answer (busy, no answer, routing failure)
abnormal_release	4	Abnormal release due to radio link failure, protocol error, or system error
CAMEL_init_call_release	5	Call released by the CAMEL service (SCP initiated release)
management_intervention	52	Call released by operator intervention

---

CDR Collector

The Collector is a GenServer that acts as the central collection point for CDR events. It receives event notifications from the Call Control FSM, VLR, and SMS handlers, and correlates them into complete CDR records.

Event Flow

For voice calls, the Collector receives a sequence of events over the call lifetime:

1. Call setup -- records the subscriber identity, called/calling numbers, direction (MO or MT), seizure time, and service type.
2. Call alerting -- logged for diagnostics but does not generate a CDR field.
3. Call answer -- records the answer timestamp and starts the partial CDR timer for long calls.
4. Call release -- calculates the call duration, selects the termination cause, generates the final CDR record, and buffers it for writing.

For SMS and location updates, the Collector generates a CDR record immediately from a single event notification.

Buffering and Flushing

CDR records are accumulated in an in-memory buffer. The buffer is flushed to the Writer under two conditions:

- A periodic flush timer fires (default interval: 5000 ms).
- The buffer reaches its maximum size (default: 1000 records), triggering an immediate flush.

Partial CDR Generation

For long-duration calls, the Collector generates intermediate partial CDR records at a configurable interval (default: 3600 seconds / 1 hour). Each partial CDR captures the call state up to that point. The final CDR record upon call release is marked as the last partial if any intermediate partials were generated. This ensures that downstream billing systems can reconstruct the complete call duration even if the MSC fails before the call ends.

Sequence Numbers

The Collector maintains independent sequence number counters for each record type (MO call, MT call, MO SMS, MT SMS, roaming, HLR location update, VLR location update). Sequence numbers increment monotonically and wrap at 10000. Downstream billing systems use sequence numbers to detect gaps indicating lost CDR records.

CDR File Naming

CDR files follow a naming convention that includes the MSC identity, timestamp, and sequence number:

<NodeID><*Date*><Time>_<SeqNum>.dat

Where:

- NodeID is the MSC name (from the recording_entity configuration).
- Date is in YYYYMMDD format.
- Time is in HHMMSS format.
- SeqNum is a zero-padded 4-digit sequence number (wraps at 10000).

For example: MSC01_20260329_143022_0001.dat

Files are written in ASN.1 BER format containing a sequence of CDR records as defined in TS 32.298.

File Rotation

CDR files are rotated (closed and a new file opened) when any of the following conditions are met:

- The file exceeds the configured maximum size (default: 10 MB).
- The file contains the configured maximum number of records (default: 100,000).

- The configured time interval has elapsed since the file was opened (default: 3600 seconds).
- An explicit rotation is triggered via the API.

Configuration

The CDR subsystem is configured through the Collector and Writer startup parameters.

Collector Parameters

Parameter	Default	Description
recording_entity	(required)	Address or name of the recording MSC, written into every CDR record
msc_address	Same as recording_entity	E.164 MSC address included in call records
flush_interval	5000 ms	Interval between periodic buffer flushes to the Writer
buffer_size	1000	Maximum number of CDR records held in the buffer before a forced flush
partial_cdr_interval	3600 seconds	Interval for generating intermediate partial CDRs on long-duration calls

Writer Parameters

Parameter	Default	Description
output_dir	(required)	Directory where CDR files are written. Created automatically if it does not exist.
node_id	(required)	Network element identifier used in CDR file names
extension	.dat	File extension for CDR files
max_file_size	10,000,000 bytes (10 MB)	Maximum file size before rotation
max_records	100,000	Maximum number of records per file before rotation
rotation_interval	3600 seconds	Maximum time a file remains open before rotation. Set to nil to disable time-based rotation.

CDR Web UI

The CDR Statistics page in the Control Panel displays real-time information about the CDR subsystem.

CDR Statistics page showing writer status, buffer depth, active tracked calls, and per-type sequence numbers.

Field	Description
Records in File	Number of CDR records written to the current output file
Pending in Buffer	Number of CDR records buffered in memory awaiting the next flush
Active Calls Tracked	Number of calls with open CDR state (between setup and release)
Current File	Path of the current CDR output file, or "No file open" if idle
Sequence Numbers	Per-record-type sequence counters showing the next sequence number for each CDR type

The page auto-refreshes every 5 seconds via WebSocket.

3GPP Specification References

Specification	Title	Relevance
TS 32.298	Charging Data Record Encoding Rules	CDR record types, ASN.1 structure, field definitions
TS 32.205	Charging Data Description for CS Domain	CS domain charging principles and CDR content requirements
TS 32.015	Charging and Billing	Overall charging architecture context

Configuration Reference

This document covers every configuration parameter for OmniMSC. Configuration is specified in Elixir config files (`config.exs`, `dev.exs`, `runtime.exs`) and can be overridden at runtime via environment variables.

For a quick start example, see the Operations Guide.

MSC Identity

`config :omnimsc, :msc`

Defines the MSC's SS7 network identity, used for SCCP addressing, MAP operations, Location Area identification, and CDR generation. The active MSC identity parameters are visible in the System page of the control panel — for more, see Control Panel Guide.

```
config :omnimsc, :msc,
  point_code: 500,
  global_title: "14155550100",
  name: "OMNIMSC01",
  msc_number: "14155550100",
  vlr_number: "14155550100",
  mcc: 313,
  mnc: 380,
  lac: 0x1092,
  allowed_a5: [:a5_1, :a5_3]
```

Parameter	Type	Required	Default	Descriptio
`point_code`	`integer` or `[integer, integer, integer]`	Yes	`0`	SS7 point cod May be specifi as a flat integ in 14-bit ITU format `[a, b,` (encoded as `a*2048 + b*8 c`).
`global_title`	`string`	Yes	`"00000000000"`	SCCP Global T (E.164 numbe for MAP routin HLR, SMSc, an peer nodes.
`name`	`string`	Yes	`"OMNIMSC01"`	Logical MSC name. Used in CDR `recording_en` field, alarm descriptions, a log messages.
`msc_number`	`string`	Yes	--	E.164 address the MSC, sent the HLR in MA Update Locati and used for M call routing.
`vlr_number`	`string`	Yes	--	E.164 address the co-located VLR, sent to th HLR in MAP Update Locati

Parameter	Type	Required	Default	Descriptio
				Typically the s as `msc_number`
`mcc`	`integer`	Yes	--	Mobile Countr Code (3 digits) Combined with `mnc` and `lac` t form the Locat Area Identity (broadcast in System Information.
`mnc`	`integer`	Yes	--	Mobile Networ Code (2 or 3 digits).
`lac`	`integer`	Yes	--	Location Area Code (16-bit). Identifies the location area served by this MSC/VLR.
`allowed_a5`	`list(atom)`	No	`[:a5_1, :a5_3]`	Permitted A5 cipher algorith for air-interfac encryption. Va values: `:a5_0` `:a5_1`, `:a5_2` `:a5_3`. Algorit negotiation prefers A5/3 > A5/1 > A5/0 (3 TS 48.008).

HLR

`config :omnimsc, :hlr`

Configures the remote HLR address for MAP operations (Send Authentication Info, Update Location, Insert Subscriber Data, Purge MS).

```
config :omnimsc, :hlr,
  address: "14155550200",
  point_code: [3, 14, 2]
```

Parameter	Type	Required	Default	Description
`address`	`string`	Yes	--	HLR Global Title (E.164 number) for MAP routing.
`point_code`	`integer` or `[integer, integer, integer]`	No	--	HLR SS7 point code for direct MTP3 routing when Global Title Translation is not used. May be a flat integer or ITU 14-bit format `[a, b, c]`.

VLR

`config :omnimsc, :vlr`

Controls Visitor Location Register behavior including authentication policy, TMSI management, and lab/guest modes.

```
config :omnimsc, :vlr,
  hlr_adapter: Omnimsc.VLR.HLR.Live,
  auth_required: true,
  tmsi_realloc: true,
  num_auth_vectors: 1
```

Parameter	Type	Required	Default	
`hlr_adapter`	`module`	No	`Omnimsc.VLR.HLR.Live`	HLR a `Omni` send: opera `Omni` provi simu testir
`auth_required`	`boolean`	No	`true`	Whet GSM authe grant `fals` Upda witho Auth
`tmsi_realloc`	`boolean`	No	`true`	Whet a nev succe Upda subs confi
`num_auth_vectors`	`integer`	No	`1`	Numl authe to re HLR Auth -4 pe 29.0 reduc at th

Parameter	Type	Required	Default	
`lab_mode`	`boolean`	No	`false`	Wher any S authe respo lab te cards not n
`guest_mode`	`boolean`	No	`false`	Wher subs to th with assig Usefu demo netw

M3UA / STP

`config :omnimsc, :m3ua_asp`

Configures the M3UA ASP (Application Server Process) connection to a Signalling Transfer Point. All SS7 signalling (A-interface, MAP to HLR/SMSc, ISUP) is routed through this link.

```
config :omnimsc, :m3ua_asp,
  enabled: true,
  local_ip: {10, 5, 198, 200},
  local_port: 0,
  remote_ip: {10, 179, 4, 10},
  remote_port: 2905,
  routing_context: 10,
  point_code: 500,
  network_indicator: :international,
  receive_watchdog: false
```

Parameter	Type	Required	Default	Descrip
`enabled`	`boolean`	No	`false`	Whether to the M3UA A client. Whe `false`, no connection established
`local_ip`	`tuple`	No	`{0, 0, 0, 0}`	Local SCTP IP address Erlang tupl
`local_port`	`integer`	No	`0`	Local SCTP port. Use `0` let the OS an epheme port.
`remote_ip`	`tuple`	Yes	--	STP SCTP I address as Erlang tupl
`remote_port`	`integer`	No	`2905`	STP SCTP p Port 2905 i IANA-assig M3UA port
`routing_context`	`integer`	No	--	M3UA rout context va Must matc STP's AS configurati this ASP.
`point_code`	`integer`	No	--	Local SS7 p code anno

Parameter	Type	Required	Default	Descrip
				to the STP during ASP Active. Sho match `:ms` `point_cod`
`network_indicator`	`atom`	No	`:international`	MTP3 netw indicator: `:internat` `:national` `:reserved` `:spare`.
`receive_watchdog`	`boolean`	No	`true`	Whether to enable the heartbeat watchdog. `true`, the A monitors fo missing BE Ack and tri link recove

Direct SCTP listeners (for BSC connections without an STP) are configured under `config :omnimsc, :sctp`:

```
config :omnimsc, :sctp,
  listeners: [
    [name: :a_interface, ip: {0, 0, 0, 0}, port: 2905, ppid: 3]
  ]
```

Parameter	Type	Required	Default	Description
`name`	`atom`	Yes	--	Logical listener name for SCTP association lookups and control panel display.
`ip`	`tuple`	No	`{0, 0, 0, 0}`	Bind IP address. Use `{0, 0, 0, 0}` for all interfaces.
`port`	`integer`	No	`2905`	SCTP bind port.
`ppid`	`integer`	No	`3`	SCTP Payload Protocol Identifier. Value `3` indicates M3UA (RFC 4666).

Runtime override: Set `SCTP_LISTEN_IP` and `SCTP_LISTEN_PORT` environment variables.

SIP

`config :omnimsc, :sip`

Configures the SIP listener and SIP peer gateways for VoIP interconnection.

```
config :omnimsc, :sip,
  signaling_address: "10.5.198.200",
  listen_ip: {0, 0, 0, 0},
  listen_port: 5060,
  transport: :udp,
  peers: [
    [name: "Default-GW", address: "10.1.1.50", port: 5060,
     transport: :udp, codecs: [:pcmu, :pcma],
     max_channels: 100, options_interval: 60],
    [name: "International-GW", address: "10.1.1.51", port: 5062,
     transport: :udp, codecs: [:pcmu, :pcma, :amr, :amr_wb]]
  ]
```

SIP Listener Parameters

Parameter	Type	Required	Default	Description
`signaling_address`	`string`	No	--	IP address used in SIP Contact headers and SDP `c=` lines. Must be reachable by SIP peers. Falls back to the SCTP listener address if not set.
`listen_ip`	`tuple`	No	`{0, 0, 0, 0}`	SIP listener bind address.
`listen_port`	`integer`	No	`5060`	SIP listener port.
`transport`	`atom`	No	`:udp`	Default transport protocol. One of `:udp`, `:tcp`, `:tls`.

SIP Peer Parameters

Each peer in the `peers` list accepts the following:

Parameter	Type	Required	Default	Descriptio
`name`	`string`	Yes	--	Logical peer nam Referenced in ro table entries with `:sip`.
`address`	`string`	Yes	--	Peer IP address o hostname.
`port`	`integer`	No	`5060`	Peer SIP port.
`transport`	`atom`	No	`:udp`	Transport for this `:udp`, `:tcp`, or `:
`codecs`	`list(atom)`	No	`[:pcmu, :pcma]`	Supported audio codecs. Valid val `:pcmu`, `:pcma`, `: :amr_wb`.
`max_channels`	`integer`	No	`100`	Maximum concu calls to this peer. calls are rejected `max_channels_re` when the limit is
`options_interval`	`integer` or `nil`	No	`nil`	Interval in secon SIP OPTIONS kee probes. Peer stat transitions to `:d` OPTIONS respons cease. Set to `nil` disable.

For routing configuration that references these peers, see Routing Configuration. For SIP OPTIONS keepalive behavior and peer health states, see SIP Trunking.

MGCP / Media

`config :omnimsc, :mgcp` and `config :omnimsc, :media`

MGCP (Media Gateway Control Protocol, RFC 3435) is used to control media gateways for bearer path setup. The MSC acts as the MGCP Call Agent, issuing CRCX, MDCX, and DLCX commands to gateways. The `:media` key selects which media control protocol to use.

```
config :omnimsc, :mgcp,
  listen_port: 2727,
  gateways: [
    %{name: "MGW-01", address: "10.1.1.50", port: 2427, domain:
"mgw"}
  ]

config :omnimsc, :media,
  gateway: "MGW-01",
  mode: :mgcp
```

MGCP Parameters

Parameter	Type	Required	Default	Description
`listen_port`	`integer`	No	`2727`	Local UDP port for the MGCP Call Agent (RFC 3435 Sec 2.2). Set to `0` to disable the MGCP transport (e.g., in test).
`gateways`	`list(map)`	No	`[]`	List of managed media gateways.

Gateway Parameters

Parameter	Type	Required	Default	Description
`name`	`string`	Yes	--	Logical gateway identifier used for lookups and control panel display.
`address`	`string`	Yes	--	IP address of the media gateway.
`port`	`integer`	No	`2427`	MGCP port on the gateway.
`domain`	`string`	No	--	Gateway domain name used in endpoint naming (e.g., `aaln/1@mgw`).

Media Mode

Parameter	Type	Required	Default	Description
`gateway`	`string`	No	--	Name of the default gateway (must match a gateway `name` from the MGCP or Megaco config).
`mode`	`atom`	No	`:mgcp`	Media control protocol: `:mgcp` for RFC 3435 or `:megaco` for ITU-T H.248.

SMSc

`config :omnimsc, :smsc`

Configures the Short Message Service Centre address for MAP MT-ForwardSM and MO-ForwardSM operations.

```
config :omnimsc, :smsc,
  address: "14155550300"
```

Parameter	Type	Required	Default	Description
`address`	`string`	Yes	--	SMSc Global Title (E.164 number) for MAP routing.

CDR

`config :omnimsc, :cdr`

CDR records are generated per 3GPP TS 32.250 by the CDR Collector and written to files in ASN.1 BER format (3GPP TS 32.298) by the CDR Writer. File naming follows the pattern `<NodeID>_<YYYYMMDD>_<HHMMSS>_<SeqNum>.dat`.

```
config :omnimsc, :cdr,
  output_dir: "/var/cdr/omnimsc",
  max_file_size: 10_000_000,
  max_records: 100_000,
  rotation_interval: 3600
```

Parameter	Type	Required	Default	Desc
`output_dir`	`string`	Yes	`"/tmp/omnimsc/cdr"`	Direct CDR o files. M writab BEAM Create autom if it do exist.
`max_file_size`	`integer`	No	`10,000,000`	Maxim file siz bytes (appro 10 MB file is when is exce
`max_records`	`integer`	No	`100,000`	Maxim numbe CDR re per file
`rotation_interval`	`integer`	No	`3600`	Time-b file rot interv secon new C create this pe regard size o count.

Routes

`config :omnimsc, :routes`

Defines prefix-based routing rules that map dialed number prefixes to destinations. The route table uses longest-prefix match with priority as a tiebreaker.

```
config :omnimsc, :routes, [
  %{prefix: "000", type: :sip, peer: "Default-GW", priority: 100},
  %{prefix: "04", type: :local, priority: 50},
  %{prefix: "02", type: :local, priority: 50},
  %{prefix: "001", type: :sip, peer: "International-GW", priority:
10},
  %{prefix: "", type: :sip, peer: "Default-GW", priority: 1}
]
```

Parameter	Type	Required	Default	Description
`prefix`	`string`	Yes	--	Number prefix to match. Empty string `""` acts as a catch-all default route.
`type`	`atom`	Yes	--	Destination type: `:local`, `:sip`, `:isup`, `:transit`, `:gmsc`, `:sip_i`, or `:sip_with_failover`.
`priority`	`integer`	No	`10`	Route priority. Higher values take precedence when multiple routes match the same prefix.
`peer`	`string`	Conditional	--	SIP peer name (required when `type` is `:sip`, `:sip_i`, or `:sip_with_failover`). Must match a peer `name` from the SIP configuration.
`trunk_group`	`string`	Conditional	--	ISUP trunk group name (required when `type` is `:isup`).
`point_code`	`[integer, integer, integer]`	Conditional	`[0, 0, 0]`	Destination point code for ISUP routing.
`cic_range`	`{integer, integer}`	No	`{1, 31}`	Inclusive range of Circuit Identification

Parameter	Type	Required	Default	Description
				Codes for ISUP trunks.
`transport`	`atom`	No	`:udp`	SIP transport override for this route.

Routes can also be managed at runtime via the REST API (`POST /routes`, `DELETE /routes`) and the Web UI. For detailed routing examples, see Routing Configuration.

MM Information

`config :omnimsc, :mm_info`

Controls the MM INFORMATION message sent to the mobile station after Location Update Accept (3GPP TS 24.008 section 9.2.15a). Contains the network name, time, and timezone.

```
config :omnimsc, :mm_info,
  network_name: "Omnitouch",
  short_name: "OT",
  timezone_offset: 0
```

Parameter	Type	Required	Default	Description
`network_name`	`string`	No	`"Omnitouch"`	Full network name displayed on the handset. Encoded in GSM 7-bit default alphabet (3GPP TS 24.008 10.5.3.5a).
`short_name`	`string` or `nil`	No	`nil`	Short network name. Omitted from the MM INFORMATION message when `nil`.
`timezone_offset`	`integer`	No	`0`	UTC offset in quarter-hours. For example, UTC+5:30 (India) is `22`, UTC-5 (US Eastern) is `-20`. Encoded in BCD per 3GPP TS 24.008 10.5.3.8.

MSC Pool

`config :omnimsc, :pool`

Configures MSC-in-Pool operation per 3GPP TS 23.236. Pool mode enables multiple MSC instances to share BSCs via A-Flex, providing load distribution and resilience.

```
config :omnimsc, :pool,
  enabled: true,
  pool_id: "POOL-01",
  nri_bitlength: 10,
  nri_values: [1, 2],
  members: [
    %{name: "MSC-02", nri_values: [3, 4], address: "10.1.1.2",
port: 2905},
    %{name: "MSC-03", nri_values: [5, 6], address: "10.1.1.3",
port: 2905}
  ]
```

Parameter	Type	Required	Default	Description
`enabled`	`boolean`	No	`false`	Whether to enable MSC pool operation. When `false`, the MSC operates in standalone mode.
`pool_id`	`string`	Conditional	`nil`	Pool area identifier. Required when `enabled` is `true`.
`nri_bitlength`	`integer`	No	`10`	Number of bits for the Network Resource Identifier field extracted from the TMSI. Must be identical across all pool members.
`nri_values`	`list(integer)`	Conditional	`[]`	NRI values owned by this MSC instance. Must not

Parameter	Type	Required	Default	Description
				overlap with other pool members. Required when `enabled` is `true`.
`null_nri`	`integer`	No	`0`	NRI value indicating an unassigned TMSI. Triggers NRI-based re-routing to the correct pool member.
`members`	`list(map)`	No	`[]`	Other MSC instances in the pool. Each member has `name`, `nri_values`, `address`, and `port`.

For pool architecture and NRI bit layout details, see MSC Pool & NRI.

Overload

`config :omnimsc, Omnimsc.Overload`

Overload protection thresholds. When any threshold is exceeded, new service requests (calls, paging, Location Updates) are rejected with GSM cause 42 (switching equipment congestion). The `admit?/0` function provides lock-free reads via `persistent_term` for minimal overhead in the hot path.

```
config :omnimsc, Omnimsc.Overload,
  max_calls: 10_000,
  max_subscribers: 50_000,
  max_process_count: 500_000,
  max_paging_rate: 1_000,
  check_interval: 5_000
```

Parameter	Type	Required	Default	Description
`max_calls`	`integer`	No	`10,000`	Maximum concurrent active calls before entering overload state.
`max_subscribers`	`integer`	No	`50,000`	Maximum VLR registered subscribers before overload.
`max_process_count`	`integer`	No	`500,000`	Maximum BEAM VM process count before overload. Monitors total VM process pressure.
`max_paging_rate`	`integer`	No	`1,000`	Maximum paging requests per second before overload.
`check_interval`	`integer`	No	`5,000`	Interval in milliseconds between overload threshold checks.

Overload state transitions emit telemetry events `[:omnimsc, :overload, :state_change]` for external monitoring. See Metrics Reference.

SGs / CSFB

`config :omnimsc, :sgs`

Configures the SGs-AP interface for Circuit-Switched Fallback (CSFB) and SMS over SGs with LTE MMEs per 3GPP TS 29.118.

```
config :omnimsc, :sgs,
  listen_port: 29118,
  vlr_name: "vlr.omnimsc.local"
```

Parameter	Type	Required	Default	Description
`listen_port`	`integer`	No	`29118`	SCTP listen port for SGs AP connections from MMEs. Port 29118 is the 3GPP-assigned default. Set to `0` to disable SGs
`vlr_name`	`string`	No	`"vlr.omnimsc.local"`	VLR name (FQDN) sent to MMEs in SGs-AP Location Update Accept. The MME uses this to identify and route to this VLR instance.

For SGs protocol details, association states, and CSFB call flows, see SGs / CSFB.

USSD

`config :omnimsc, :ussd`

Configures external USSD gateways for routing Unstructured Supplementary Service Data requests. Each gateway handles specific USSD service codes (e.g. `*100#` for balance). A gateway with `codes: :all` acts as the default fallback for unmatched codes.

```
config :omnimsc, :ussd,
  gateways: [
    %{name: "Balance", address: "14155550300", ssn: 147, codes:
["*100"]},
    %{name: "Recharge", address: "14155550301", ssn: 147, codes:
["*123"]},
    %{name: "Default", address: "14155550302", ssn: 147, codes:
:all}
  ]
```

USSD Gateway Parameters

Each gateway in the `gateways` list accepts the following:

Parameter	Type	Required	Default	Description
`name`	`string`	No	`"unnamed"`	Logical gateway name for logging and control panel display.
`address`	`string`	Yes	--	Gateway Global Title (E.164 number) for MAP USSD routing.
`ssn`	`integer`	No	`147`	SCCP Subsystem Number for the gateway. SSN 147 is the standard USSD SSN.
`codes`	`list(string)` or `:all`	No	`:all`	USSD service codes handled by this gateway (e.g. `["*100", "*101"]`). Set to `:all` for a catch-all default gateway.

For USSD protocol details and relay behavior, see USSD.

Emergency

`config :omnimsc, Omnimsc.Emergency`

Configures emergency number detection, service category classification, and PSAP routing per 3GPP TS 22.101.

Emergency Setup messages (3GPP TS 24.008 §9.3.8) do not carry a Called Party BCD Number IE — unlike regular CC SETUP, the handset does not include the dialed digits. OmniMSC uses the configured `psap_address` as the called number for route table lookup and the outgoing SIP INVITE Request-URI. This value must match a prefix in the route table so that the call can be routed to the appropriate SIP peer or trunk.

```
config :omnimsc, Omnimsc.Emergency,
  numbers: ["112", "911", "999", "000", "110", "119"],
  psap_address: "000",
  allow_without_sim: true
```

Parameter	Type	Required	Default	Description
`numbers`	`list(string)`	No	`["112", "911", "999", "000", "110", "119"]`	Recognized emergency numbers. Cal to these numbers bypass authenticatio ciphering, an call barring.
`psap_address`	`string`	No	`"112"`	Called numbe used for emergency call routing. Because Emergency Setup messages carry no calle number, this value is used as the called party for rout table lookup and the outgoing trun (SIP INVITE Request-URI ISUP IAM Called Party Number). Set this to a number that matches an emergency

Parameter	Type	Required	Default	Description
				route prefix in the route table.
`allow_without_sim`	`boolean`	No	`true`	Whether to allow emergency calls from mobile stations with no SIM inserted (IMSI absent). Per 3GPP TS 22.101, networks should permit this.

Web UI

`config :omnimsc, OmnimscWeb.Endpoint`

The web control panel is served by a Phoenix endpoint with LiveView. It provides real-time dashboards for subscribers, calls, connections, routing, and alarms.

```
config :omnimsc, OmnimscWeb.Endpoint,
  http: [ip: {0, 0, 0, 0}, port: 4000],
  url: [host: "localhost"],
  secret_key_base: "generate-with-mix-phx-gen-secret",
  server: true,
  pubsub_server: Omnimsc.PubSub,
  live_view: [signing_salt: "oMnImScLv"]
```

Parameter	Type	Required	Default	De
`http.ip`	`tuple`	No	`{0, 0, 0, 0}`	HTTP addre `{127` to res locall
`http.port`	`integer`	No	`4000`	HTTP for th pane
`url.host`	`string`	No	`"localhost"`	Hostn URL Set t hostn produ
`secret_key_base`	`string`	Yes	--	Phoe signi Gene `mix phx.g` Requ produ from `SECR` envir varia
`server`	`boolean`	No	`true`	Whet the H Set t disab contr

Parameter	Type	Required	Default	De
`check_origin`	`boolean`	No	`true` (prod)	Whet WebS head `fals` deve
`pubsub_server`	`atom`	No	`Omnimsc.PubSub`	PubS name LiveV broa
`live_view.signing_salt`	`string`	No	`"oMnImScLv"`	LiveV signi

Runtime override: Set `SECRET_KEY_BASE`, `PHX_HOST`, and `PORT` environment variables. In production, HTTPS with port 443 is configured automatically.

REST API

`config :api_ex`

The REST API is served by api_ex on a separate port, providing programmatic access to subscribers, calls, routes, SIP peers, connections, and system health.

```
config :api_ex,
  api: %{
    port: 8444,
    listen_ip: "0.0.0.0",
    product_name: "Omnitouch MSC",
    title: "API - Omnitouch MSC",
    hostname: "localhost",
    enable_tls: false
  }
```

Parameter	Type	Required	Default	Description
`port`	`integer`	No	`8444`	HTTP listen port for the REST API.
`listen_ip`	`string`	No	`"0.0.0.0"`	Bind IP address for the API listener.
`product_name`	`string`	No	`"Omnitouch MSC"`	Product name displayed in the Swagger UI.
`title`	`string`	No	`"API - Omnitouch MSC"`	Page title for the Swagger UI.
`hostname`	`string`	No	`"localhost"`	Hostname for API URL generation.
`enable_tls`	`boolean`	No	`false`	Whether to enable TLS for the API endpoint.

Available API Endpoints

Path	Methods	Description
`GET /subscribers`	GET, DELETE	List or remove VLR subscribers.
`POST /subscribers/:id/actions`	POST	Trigger subscriber actions (paging, detach).
`GET /calls`	GET, DELETE	List or release active calls.
`GET /sms`	GET	List SMS transactions.
`GET /routes`	GET, POST, DELETE	Manage the route table.
`GET /routes/lookup`	GET	Look up a route by dialed number.
`GET /sip/peers`	GET, PATCH	List or update SIP peer configuration.
`GET /mgw`	GET	List media gateway status.
`GET /ran/connections`	GET	List active RAN (A-interface) connections.
`GET /ran/bscs`	GET	List connected BSCs.
`GET /stp`	GET	Show STP connection status.
`GET /health`	GET	System health check.
`GET /status`	GET	System status summary.

Path	Methods	Description
`POST /paging`	POST	Trigger a paging request.
`POST /silent`	POST	Initiate a silent call or silent SMS.

Complete Production Configuration

Example

```
# config/runtime.exs
import Config

config :omnimsc, :msc,
    point_code: 500,
    global_title: "14155550100",
    name: "OMNIMSC01",
    msc_number: "14155550100",
    vlr_number: "14155550100",
    mcc: 313,
    mnc: 380,
    lac: 0x1092,
    allowed_a5: [:a5_1, :a5_3]

config :omnimsc, :hlr,
  address: "14155550200",
  point_code: [3, 14, 2]

config :omnimsc, :vlr,
  hlr_adapter: Omnimsc.VLR.HLR.Live,
  auth_required: true,
  tmsi_realloc: true,
  num_auth_vectors: 1

config :omnimsc, :m3ua_asp,
  enabled: true,
  local_ip: {10, 5, 198, 200},
  local_port: 0,
  remote_ip: {10, 179, 4, 10},
  remote_port: 2905,
  routing_context: 10,
  point_code: 500,
  network_indicator: :international,
  receive_watchdog: true

config :omnimsc, :sip,
  signaling_address: "10.5.198.200",
  listen_ip: {0, 0, 0, 0},
  listen_port: 5060,
  transport: :udp,
```

```
  peers: [
    [name: "Default-GW", address: "10.1.1.50", port: 5060,
     transport: :udp, codecs: [:pcmu, :pcma],
     max_channels: 100, options_interval: 60],
    [name: "International-GW", address: "10.1.1.51", port: 5062,
     transport: :udp, codecs: [:pcmu, :pcma, :amr, :amr_wb],
     max_channels: 500]
  ]

config :omnimsc, :mgcp,
  listen_port: 2727,
  gateways: [
    %{name: "MGW-01", address: "10.1.1.50", port: 2427, domain:
"mgw"}
  ]

config :omnimsc, :media,
  gateway: "MGW-01",
  mode: :mgcp

config :omnimsc, :smsc,
  address: "14155550300"

config :omnimsc, :cdr,
  output_dir: "/var/cdr/omnimsc",
  max_file_size: 10_000_000,
  max_records: 100_000,
  rotation_interval: 3600

config :omnimsc, :routes, [
  %{prefix: "000", type: :sip, peer: "Default-GW", priority: 100},
  %{prefix: "04", type: :local, priority: 50},
  %{prefix: "02", type: :local, priority: 50},
  %{prefix: "001", type: :sip, peer: "International-GW", priority:
10},
  %{prefix: "", type: :sip, peer: "Default-GW", priority: 1}
]

config :omnimsc, :mm_info,
  network_name: "Omnitouch",
  short_name: "OT",
  timezone_offset: 0

config :omnimsc, Omnimsc.Overload,
```

```
  max_calls: 10_000,
  max_subscribers: 50_000,
  max_process_count: 500_000,
  max_paging_rate: 1_000,
  check_interval: 5_000

config :omnimsc, Omnimsc.Emergency,
  numbers: ["112", "911", "999", "000", "110", "119"],
  psap_address: "000",
  allow_without_sim: true

config :omnimsc, :sgs,
  listen_port: 29118,
  vlr_name: "vlr.omnimsc.local"

config :omnimsc, :ussd,
gateways: []

config :omnimsc, :pool,
enabled: false
```

Control Panel

This document describes the OmniMSC web-based control panel, a real-time monitoring and management interface built with Phoenix LiveView. The control panel is accessible at `http://<host>:4000` and provides live visibility into subscribers, calls, connections, routing, CDRs, and system health.

All pages auto-refresh every 5 seconds via WebSocket push. No manual page reload is required. The auto-refresh toggle on each page allows pausing updates when inspecting a particular record.

For endpoint configuration (bind address, port), see Configuration Reference. For the REST API, see API Reference.

Dashboard

The dashboard is the main landing page, providing an at-a-glance summary of the entire MSC.

Summary Cards

The top row displays six real-time counters:

Card	Description
Subscribers	Number of subscribers currently registered in the VLR
Active Calls	Number of active CC FSM call transactions
Active SMS	Number of in-progress SMS transactions
RAN Connections	Count of BSCs and RNCs with ESTABLISHED SCTP associations
STP Link Status	M3UA ASP state toward the STP (ACTIVE, INACTIVE, DOWN)
System Uptime	Elapsed time since application start

SS7 Links Table

Displays the status of all configured SS7 signalling links, including M3UA ASP state and traffic mode.

Known BSCs Table

Lists each BSC by name, point code, cell count, and SCTP association state.

SIP Peers Table

Column	Description
Name	Peer logical name
Address	Peer IP address and SIP port
Calls	Current active call count on this peer
Status	Health state badge (Up, Down, Unknown)

Media Gateways Table

Lists configured media gateways with name, address, protocol (MGCP or Megaco), and reachability status.

Recent Events Feed

A scrolling feed of the most recent telemetry events, each tagged with an event type badge (CALL, LU, PEER, SMS, INFO) and UTC timestamp. Events are pushed in real time as they occur.

Subscribers

The Subscribers page provides a searchable list of all VLR subscriber records. Enter an IMSI or MSISDN (partial match supported) in the search box to filter the list in real time.

Subscriber List Columns

Column	Description
IMSI	International Mobile Subscriber Identity
MSISDN	Mobile Station ISDN Number
TMSI	Temporary Mobile Subscriber Identity allocated by the VLR
LAC	Location Area Code of the subscriber's current cell
State	VLR registration state
Auth	Authentication status
LU	Location Update completion status

Expandable Subscriber Detail

Clicking a subscriber row expands a detailed view organised into the following sections.

Identity

Field	Description
IMSI	International Mobile Subscriber Identity
MSISDN	Mobile Station ISDN Number
TMSI	Temporary Mobile Subscriber Identity
IMEI	International Mobile Equipment Identity (if available)
HLR Number	Address of the subscriber's home HLR

Location and State

Field	Description
LAC	Location Area Code
CI	Cell Identity
Serving BSC	Name of the BSC currently serving this subscriber
RAN Type	Radio access type (GERAN-A, UTRAN-Iu, or SGs)
Usage Count	Number of active use count tokens held by this subscriber's MSC-A
Expires	VLR record expiry timestamp
State	Current subscriber state in the VLR
Roaming	Whether the subscriber is marked as roaming
SGs	SGs association state (if registered via MME for CSFB)

Authentication

Field	Description
Algorithm	Authentication algorithm in use (COMP128v1, COMP128v3, Milenage)
Tuples	Number of remaining authentication triplets
UMTS Quintuplet	Whether UMTS authentication quintuplets are available

Service Profile

Displays the circuit-switched services received from the HLR via MAP Insert Subscriber Data, including bearer service and teleservice subscriptions.

Supplementary Services

Lists all provisioned supplementary services with status badges:

Service	Description
BAOC	Barring of All Outgoing Calls
BOIC	Barring of Outgoing International Calls
BOIC-exHC	Barring of Outgoing International Calls except to Home Country
BAIC	Barring of All Incoming Calls
BIC-Roam	Barring of Incoming Calls when Roaming
CFU	Call Forwarding Unconditional
CFB	Call Forwarding on Busy
CFNRy	Call Forwarding on No Reply
CFNRc	Call Forwarding on Not Reachable
CW	Call Waiting
HOLD	Call Hold
MPTY	Multi-Party
CLIP	Calling Line Identification Presentation
CLIR	Calling Line Identification Restriction

Each service displays an active/inactive status badge and, where applicable, the forwarded-to number and condition parameters.

MSC-A Connection

Displays the current MSC-A FSM state for any active connection associated with this subscriber, including the state name, active use count tokens, and elapsed time in state.

Connections

The Connections page provides visibility into all signalling links, RAN connections, and SIP peers.

STP Link (M3UA ASP)

Field	Description
Local IP	Local SCTP bind address
Remote	STP remote address and port
Association ID	SCTP association identifier
Status	M3UA ASP state badge (ACTIVE, INACTIVE, DOWN)

All signalling transport uses SCTP (Stream Control Transmission Protocol) for reliable, multi-homed delivery per RFC 4960.

Known BSCs

Column	Description
Point Code	BSC SS7 point code
Global Title	BSC global title address (if configured)
Last Reset	Timestamp of the most recent BSSMAP RESET exchange with this BSC

SIP Peers

Column	Description
Name	Peer logical name
Address	Peer IP address and SIP port
Transport	Protocol in use (UDP, TCP, or TLS)
Active Calls	Current number of active calls to this peer
Capacity	Maximum concurrent channels configured for this peer
Last OPTIONS	Timestamp of the most recent successful OPTIONS keepalive response
Status	Health state badge (Up, Down, Unknown)

Active Calls

The Active Calls page displays a live table of all CC FSM call transactions currently in progress.

Call Table Columns

Column	Description
Call Ref	Unique call reference number
Direction	MO (Mobile Originating) or MT (Mobile Terminating) badge
IMSI	Subscriber IMSI
Calling	Calling party number (A-number)
Called	Called party number (B-number)
State	CC FSM state with colour-coded badge
Duration	Elapsed time since call seizure
Codec	Negotiated speech codec
BSC/RNC	Name of the serving BSC or RNC

Expandable Call Detail

Clicking a call row expands a detailed view with two panels.

Call Details

Field	Description
Call Ref	Unique call reference
Direction	MO or MT
CC FSM State	Current state of the Call Control finite state machine
IMSI	Subscriber IMSI
MSISDN	Subscriber phone number
IMEI	Mobile equipment identity
Calling Party	A-number as presented on the signalling interface
Called Party	B-number as presented on the signalling interface

Timing and Resources

Field	Description
Duration	Elapsed call duration
Seizure Time	UTC timestamp of call setup start
Answer Time	UTC timestamp of call answer (blank if not yet answered)
Basic Service	Bearer service or teleservice associated with the call
Active CC Timer	Currently running CC protocol timer (if any)
BSC/RNC	Name of the serving BSC or RNC handling the radio side

---

Routes and Trunks

The Routes and Trunks page provides three tabbed views for managing call routing configuration at runtime.

Route Table Tab

Displays all prefix-based routing rules with search filtering by prefix or destination type.

Column	Description
Prefix	Number prefix (empty string indicates the catch-all default route)
Destination Type	Badge indicating route type: EMRG, SIP, LOCAL, ISUP, TRANSIT, GMSC
Details	Destination-specific information such as peer name or trunk group
Priority	Numeric priority value (higher values take precedence)
Actions	Edit and Delete buttons for each route

The **Add Route** button opens a modal form supporting all destination types. Route changes take effect immediately without requiring a restart.

For detailed routing concepts and configuration, see Routing Configuration.

ISUP Trunks Tab

Displays configured ISUP trunk groups with circuit availability and call counts.

SIP Peers Tab

Displays SIP peer status with address, transport, codec support, channel capacity, and active call counts.

SMS

The SMS page lists all active SMS transaction FSMs, showing transaction ID, subscriber IMSI, direction (MO or MT), transaction state, and SMS Centre address. Completed transactions are removed from the list automatically.

CDRs

The CDR Statistics page provides visibility into the Charging Data Record subsystem.

Summary Counters

Counter	Description
Records in File	Number of CDR records written to the current output file
Pending in Buffer	Number of CDR records buffered in memory awaiting the next write cycle
Active Calls Tracked	Number of calls with open (not yet finalised) CDR records

Writer Status

Field	Description
Current File Path	Filesystem path of the active CDR output file
Records Written	Total records written to the current file
Buffer Pending	Records queued in the write buffer

Sequence Numbers

Per-record-type sequence counters track the monotonically increasing sequence number for each CDR category:

Record Type	Description
LU HLR	Location Update toward HLR
LU VLR	Location Update in VLR
MO Call	Mobile Originating voice call
MT Call	Mobile Terminating voice call
MO SMS	Mobile Originating short message
MT SMS	Mobile Terminating short message
Roaming	Roaming event records

Pool

The Pool page is accessible when MSC pool mode is enabled. It displays the status of all MSC pool members, including member name, point code, NRI range, health state (Up, Down, or Draining), and last successful health probe timestamp. A subscriber distribution chart shows the NRI allocation across pool members.

For pool configuration and NRI-based routing, see MSC Pool and NRI.

System

The System page provides detailed information about the BEAM VM, memory allocation, MSC identity, transport links, and supervision tree health.

BEAM VM

Field	Description
OTP Release	Erlang/OTP major version
Processes	Current process count and configured limit
Ports	Open port count
Atoms	Atom table size
Schedulers	Number of online schedulers
Uptime	BEAM VM uptime since boot

Memory

Field	Description
Total	Total memory allocated by the BEAM VM
Processes	Memory consumed by Erlang/Elixir processes
ETS	Memory consumed by ETS tables
Binary	Memory consumed by binary reference-counted data
Atom	Memory consumed by the atom table
System	Memory consumed by the runtime system (non-process)

MSC Configuration

Displays the active MSC identity parameters:

Field	Description
Name	MSC logical name
Point Code	Local SS7 point code
Global Title	MSC global title address
Allowed A5	Permitted A5 ciphering algorithms for GERAN

SCTP Transport

Field	Description
Link	Transport link name
Local IP	Local SCTP bind address
Remote	Remote STP address and port
Status	SCTP association state badge

Supervision Tree Health

Lists all children of the top-level OmniMSC supervisor (44 children), each displayed with:

Field	Description
PID	Erlang process identifier
Type	Process type (worker or supervisor)
Status	Health badge (Running, Restarting, or Stopped)

This view is useful for verifying that all subsystems are operational after startup or following a fault recovery event.

ISUP Trunking

This document describes the ISUP (ISDN User Part) trunk interface implemented by OmniMSC, including trunk group management, circuit allocation, message encoding, timers, continuity check support, and integration with the route table and SIP-I.

For ISUP-SIP cause code mapping, see SIP Trunking. For SIP-I (SIP with encapsulated ISUP), see SIP-I Trunking. For routing configuration and the `:isup` route type, see Routing Configuration. For call flow diagrams showing ISUP signaling in context (IAM/ACM/ANM, ISUP-to-SIP transit), see Call Flow Diagrams.

ISUP Trunk Groups

OmniMSC organizes ISUP circuits into trunk groups. Each trunk group represents a bundle of voice circuits to a remote SS7 exchange, identified by a destination point code and a range of Circuit Identification Codes (CICs).

CIC Allocation

When a call is routed to an ISUP trunk group, the circuit manager allocates a free CIC from the configured range using a sequential hunt algorithm. The allocated CIC is included in the outgoing IAM and reserved until the call is released.

Parameter	Type	Description
`trunk_group`	`string`	Unique trunk group identifier, referenced in route table entries
`point_code`	`list`	Destination point code as `[a, b, c]`, encoded as `a*2048 + b*8 + c`
`cic_range`	`{start, end}`	Inclusive range of CICs available for this trunk group

Circuit State Management

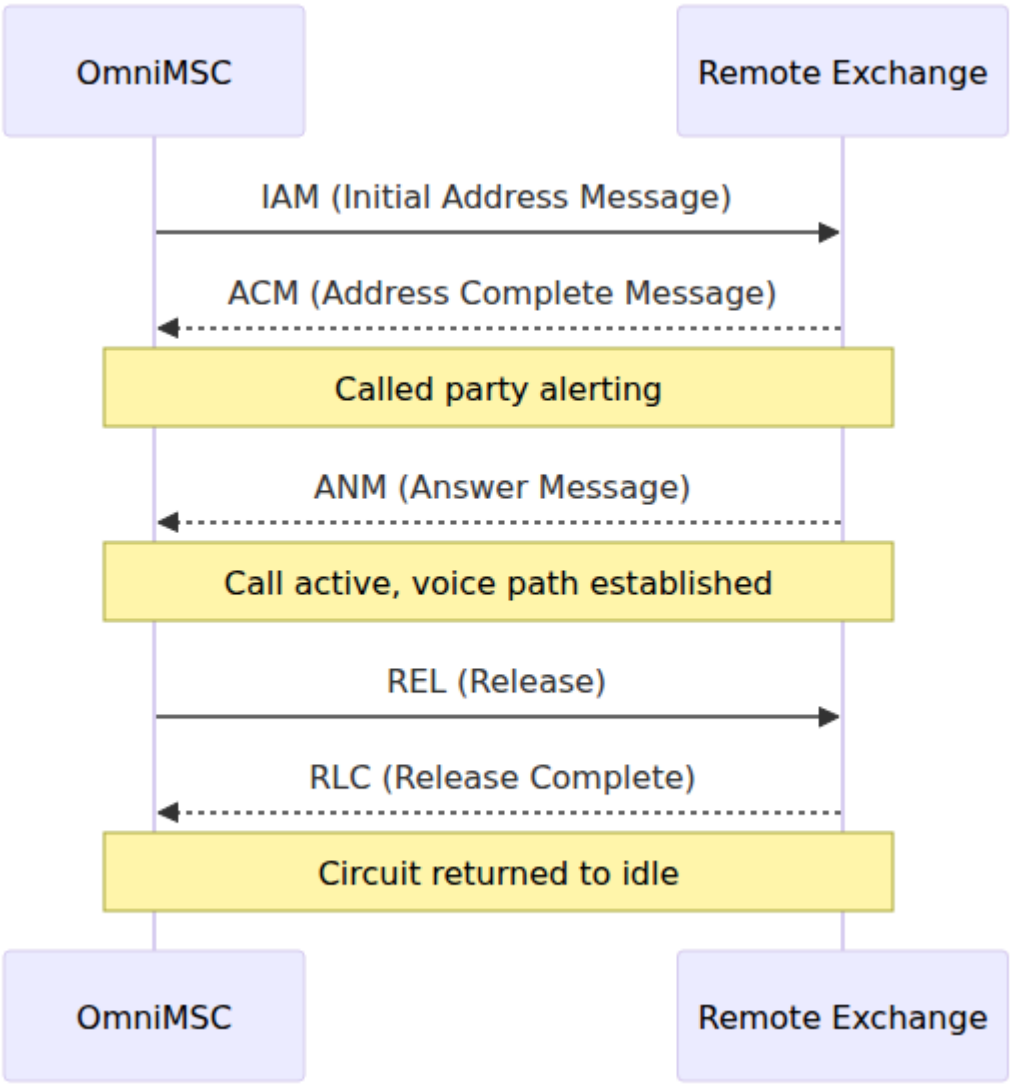
Each circuit within a trunk group tracks an independent state. The circuit manager handles blocking, unblocking, and group reset operations.

Circuit State	Description
Idle	Available for seizure
Seized	Allocated for an outgoing call (IAM sent)
Incoming	Reserved for an incoming call (IAM received)
Active	Call in progress (ANM exchanged)
Blocked (local)	Locally blocked via BLO, unavailable for seizure
Blocked (remote)	Remotely blocked via BLO from far end
Unequipped	CIC exists in range but is not provisioned

Circuit blocking (BLO) and unblocking (UBL) can be performed per-circuit or in groups (CGB/CGU). Group reset (GRS/GRA) resets all circuits in a range to idle and clears any blocking state.

ISUP Message Flow

The standard ISUP signaling sequence for a successful call follows the IAM-ACM-ANM-REL-RLC pattern.

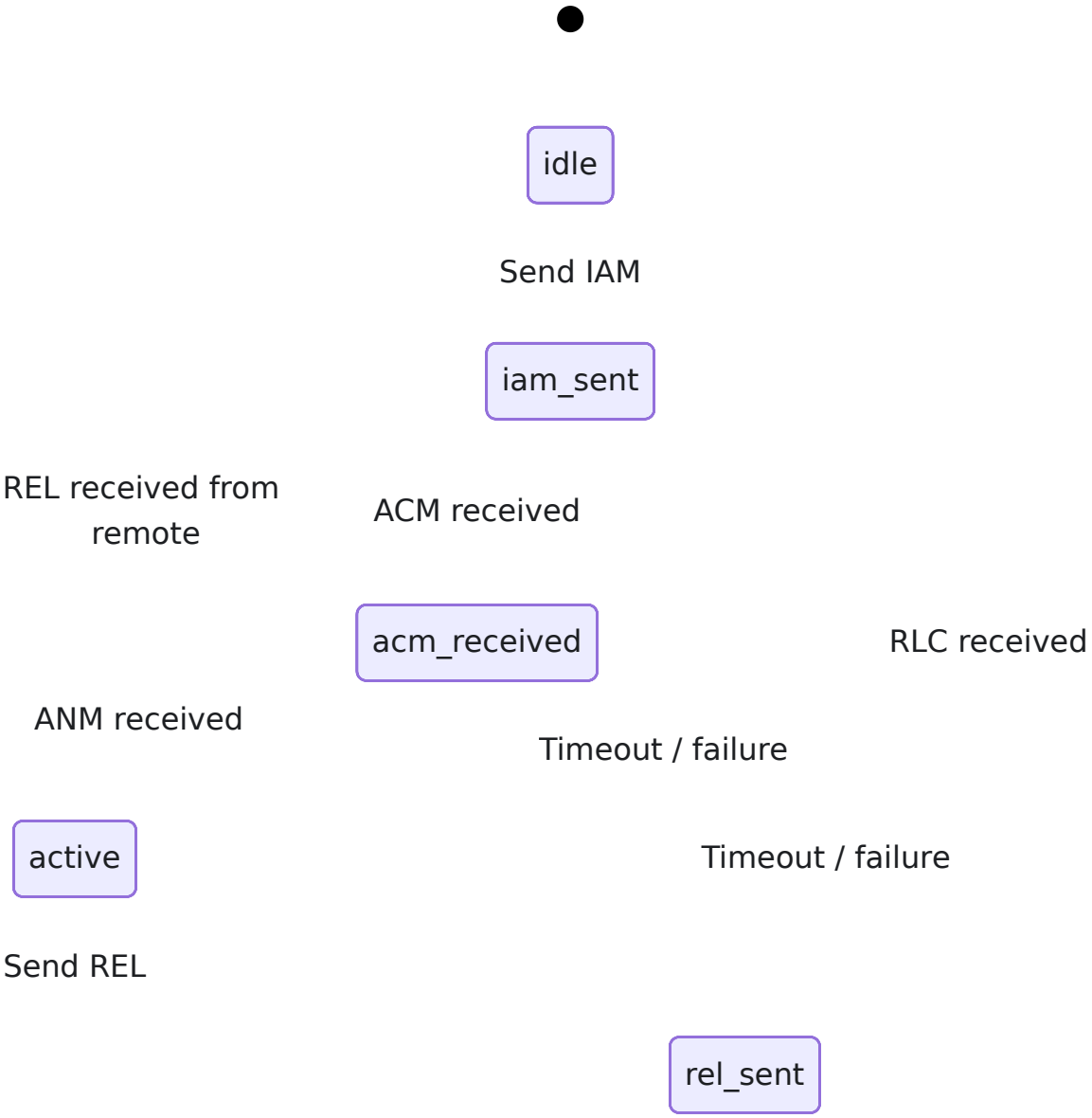


For incoming calls, the directions are reversed: IAM arrives from the remote exchange, and OmniMSC sends ACM and ANM as the call progresses through alerting and answer.

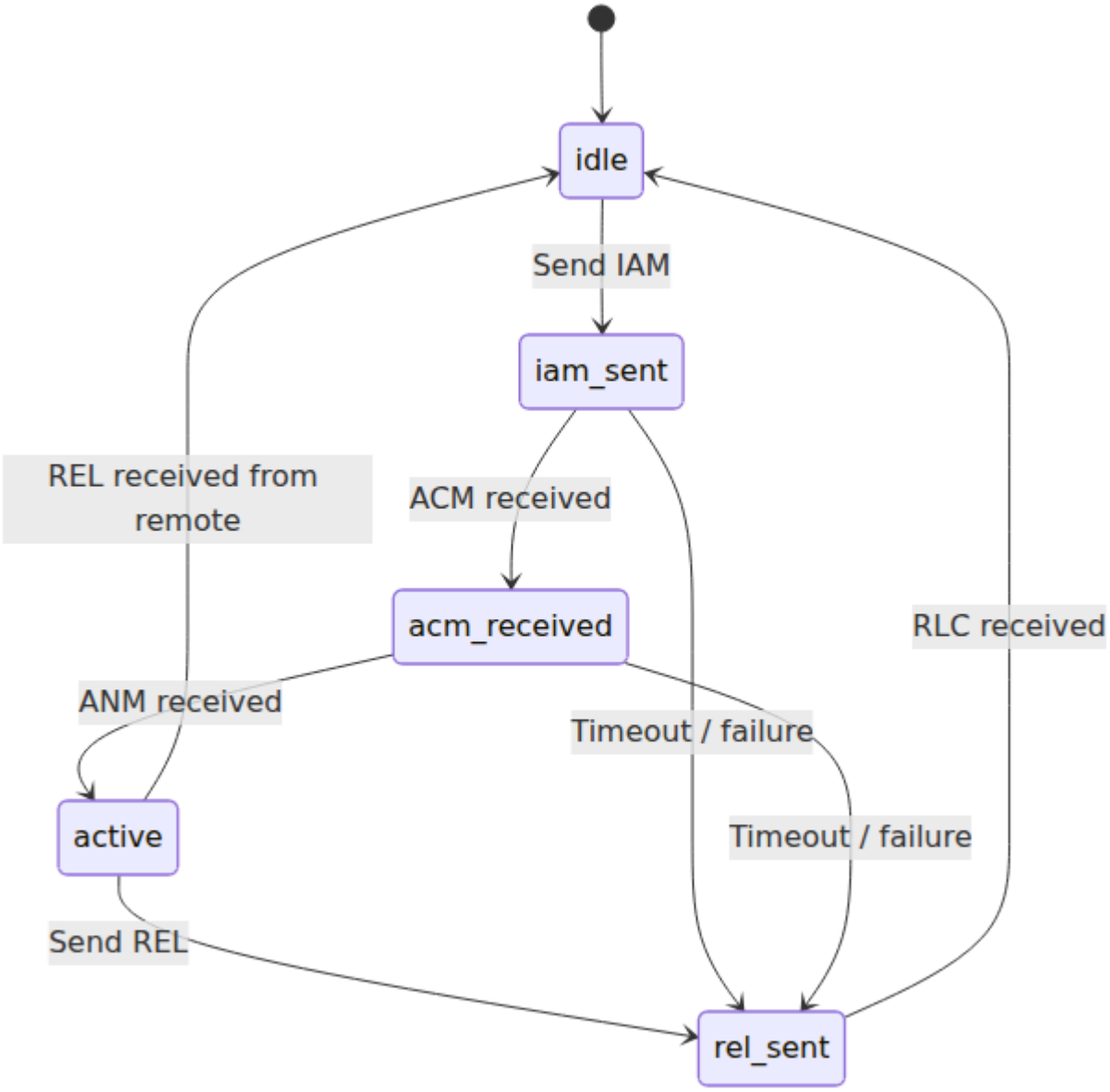
ISUP Handler States

The ISUP handler maintains per-call state that tracks the signaling progress for each circuit.

Outgoing Call States



Incoming Call States



ISUP Timers

OmniMSC implements the standard ISUP timers defined in ITU-T Q.764. These timers guard against signaling failures and ensure circuits are not left in indeterminate states.

Timer	Duration	Started After	Expires When	Action on Expiry
T1	20s	REL sent	RLC not received	Retransmit REL
T5	300s	T1 first expiry	RLC still not received	Send maintenance alert, reset circuit
T7	25s	IAM sent	ACM not received	Release call, send REL
T9	180s	ACM received	ANM not received	Release call, send REL

When T7 expires without receiving an ACM, OmniMSC sends a REL with cause 102 (recovery on timer expiry) and returns the circuit to idle. When T9 expires without an ANM, OmniMSC sends a REL with cause 19 (no answer from user).

ISUP Message Encoding

OmniMSC implements a codec for encoding and decoding the five core ISUP message types used in call setup and release. All messages follow the ITU-T Q.763 format with mandatory fixed, mandatory variable, and optional parameter sections.

Message	Type Code	Direction	Key Parameters
IAM	`0x01`	Originating	Nature of Connection, Forward Call Indicators, Calling Party Category, Transmission Medium Requirement, Called Party Number, Calling Party Number
ACM	`0x06`	Terminating	Backward Call Indicators, Optional Backward Call Indicators, Cause Indicators
ANM	`0x09`	Terminating	Backward Call Indicators
REL	`0x0C`	Either	Cause Indicators
RLC	`0x10`	Either	(none -- acknowledgement only)

Additional messages supported for circuit maintenance:

Message	Type Code	Purpose
BLO	`0x13`	Block a circuit (local maintenance)
UBL	`0x14`	Unblock a circuit
GRS	`0x17`	Group reset of a circuit range
GRA	`0x29`	Group reset acknowledgement
COT	`0x05`	Continuity check result

Continuity Check

OmniMSC supports the ISUP continuity check procedure for verifying the bearer path before connecting a call. When the Forward Call Indicators in an outgoing IAM request a continuity check, the following sequence occurs:

1. OmniMSC seizes the circuit and sends IAM with the continuity check indicator set.
2. A loopback is applied at the far end of the circuit.
3. OmniMSC sends a test tone and verifies the return.
4. On success, OmniMSC sends COT (continuity check successful) and the call proceeds.
5. On failure, OmniMSC sends COT (check failed) and may re-attempt on an alternate circuit.

For incoming calls, when OmniMSC receives an IAM with the continuity check indicator, it applies a loopback on the specified circuit and waits for the COT message before proceeding with call setup.

Route Table Integration

Routes with type `:isup` in the route table direct calls to an ISUP trunk group. The route entry specifies the trunk group name, destination point code, and CIC range.

Route Parameter	Description
`type`	`:isup`
`trunk_group`	Trunk group name matching a configured trunk group
`point_code`	Destination point code as `[a, b, c]`
`cic_range`	CIC range as `{start, end}`

When routing selects an ISUP destination, the trunk router requests a free circuit from the circuit manager. If no circuits are available in the primary trunk group, the system attempts overflow trunk groups to the same destination point code.

For route configuration examples, see Routing Configuration.

SIP-I Support

SIP-I (SIP with encapsulated ISUP) provides an IP-based transport for ISUP messages. Routes with type `:sip_i` carry the full ISUP message (IAM, ACM, ANM, REL) as an `application/ISUP` MIME body within SIP signaling per ITU-T Q.1912.5 and RFC 3204.

SIP-I peers are configured separately from pure SIP peers. The ISUP encoder/decoder used for native ISUP trunks is shared with SIP-I for encoding and decoding the encapsulated ISUP bodies.

For SIP-I configuration, call flows, and header mapping, see SIP-I Trunking.

SIP with ISUP Failover

Routes with type `:sip_with_failover` attempt the call via a SIP peer first. If the SIP route fails (peer unreachable, 5xx response, or timeout), the trunk router automatically retries via the configured ISUP trunk group.

Failover Trigger	Description
Peer status `:down`	SIP peer is unreachable (OPTIONS keepalive failed)
SIP 5xx response	Server error from the SIP peer
SIP timeout	No response within the SIP transaction timer
`max_channels` exceeded	SIP peer has no available capacity

On failover, the trunk router allocates a CIC from the ISUP trunk group and sends an IAM. The CC FSM remains in the same state throughout the failover -- the retry is transparent to the radio-side signaling.

For failover route configuration, see Routing Configuration.

References

Reference	Title	Relevance
ITU-T Q.761	Functional Description of the ISUP	ISUP overview and architecture
ITU-T Q.762	General Function of Messages and Signals of the ISUP	Message definitions
ITU-T Q.763	Formats and Codes of the ISUP	Message encoding and parameter formats
ITU-T Q.764	Signaling Procedures of the ISUP	Call setup/release procedures, timer definitions
ITU-T Q.850	Usage of Cause and Location in ISDN	Cause code definitions used in REL
RFC 3204	MIME Media Type for ISUP and QSIG Objects	ISUP encapsulation in SIP-I
ITU-T Q.1912.5	Interworking between SIP and BICC or ISUP	SIP-I protocol definition

MAP Operations

This document describes the MAP (Mobile Application Part) operations implemented by OmniMSC, including dialogue management, subscriber location procedures, authentication, SMS forwarding, USSD, and error handling. For configuration, see Configuration Reference. For the SMS-layer view of MO-ForwardSM and MT-ForwardSM, including CP/RP protocol details, see SMS. For authentication vector handling and the Auth FSM, see Security. For how InsertSubscriberData populates the subscriber profile visible in the control panel, see Control Panel Guide.

MAP Client

OmniMSC operates a MAP client that manages dialogues with remote MAP peers (HLR, SMSc, USSD gateway). The MAP client handles dialogue lifecycle, TCAP transaction correlation, and dialogue timeout supervision.

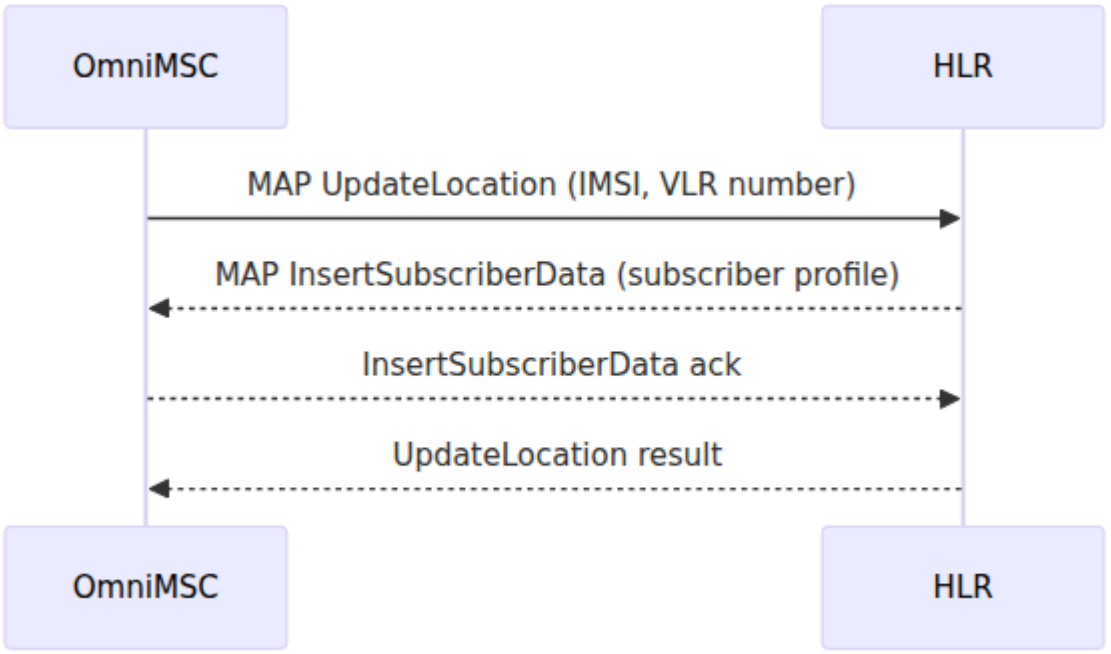
When a subsystem (VLR, SMS handler, SS handler) needs to invoke a MAP operation, it requests a new dialogue from the MAP client. The client allocates a local transaction ID (otid), opens a TCAP BEGIN towards the peer, and monitors the dialogue for a configurable timeout period. If the peer does not respond within the timeout, the client aborts the dialogue and notifies the requesting subsystem of the failure.

For incoming MAP dialogues initiated by remote peers (such as InsertSubscriberData from the HLR or MT-ForwardSM from the SMSc), the MAP client accepts the TCAP BEGIN, correlates it with the remote transaction ID (dtid), spawns a handler process, and supervises the dialogue until completion.

UpdateLocation

The MSC sends MAP UpdateLocation to the HLR when a subscriber performs a Location Update. The message carries the subscriber IMSI and the VLR number

(the E.164 address of this MSC/VLR). The HLR uses the VLR number to route future MT transactions (calls, SMS, USSD) to the correct MSC.



On receiving the UpdateLocation result, the VLR marks the Location Update as complete and the LU FSM proceeds to TMSI allocation. If the HLR returns an error (unknown subscriber, roaming not allowed), the MSC rejects the Location Update towards the mobile station with the appropriate cause.

InsertSubscriberData

The HLR sends MAP InsertSubscriberData to the MSC during the UpdateLocation procedure, and also proactively when the subscriber profile changes (e.g., supplementary service activation via the HLR provisioning interface). The message carries the subscriber profile including:

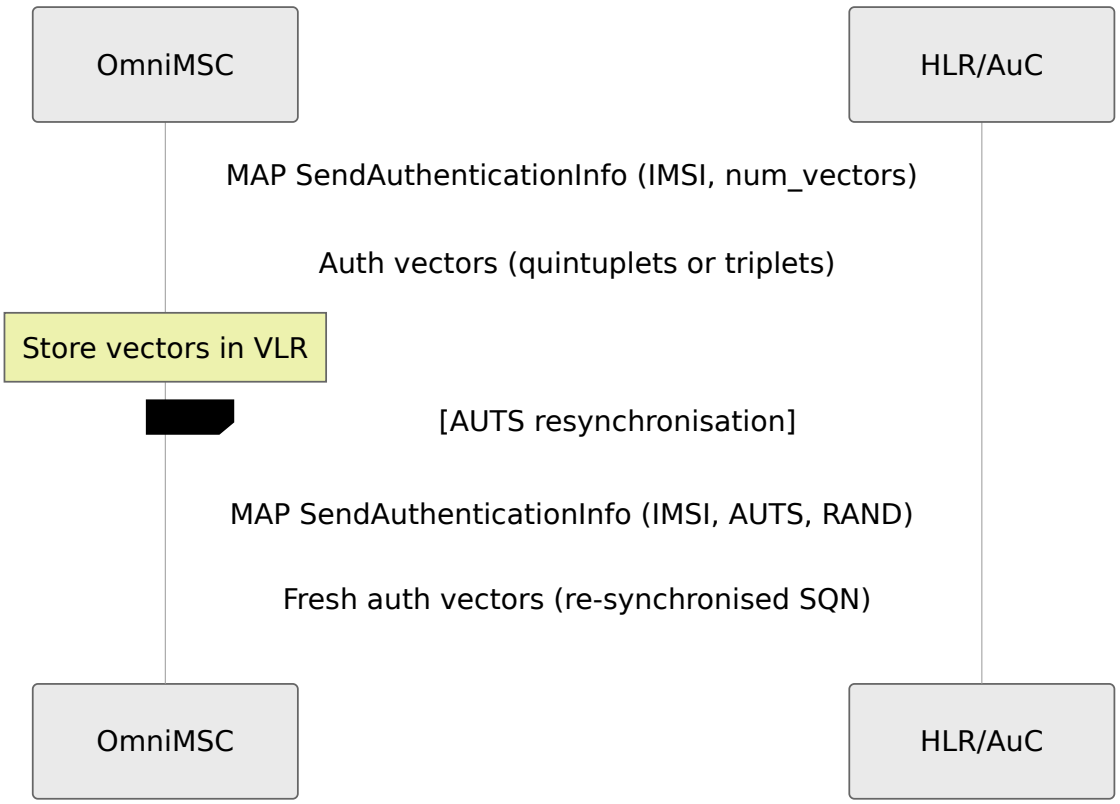
- MSISDN (the subscriber's directory number)
- CS bearer services and teleservices
- Operator Determined Barring (ODB) categories
- Supplementary service data (call forwarding numbers, barring status, CLIR mode, CW status)
- CAMEL subscription information (service keys, gsmSCF addresses)

The VLR stores this data in the subscriber record. Subsequent call setup, SMS delivery, and SS operations reference this locally cached profile to avoid round-trips to the HLR.

SendAuthenticationInfo

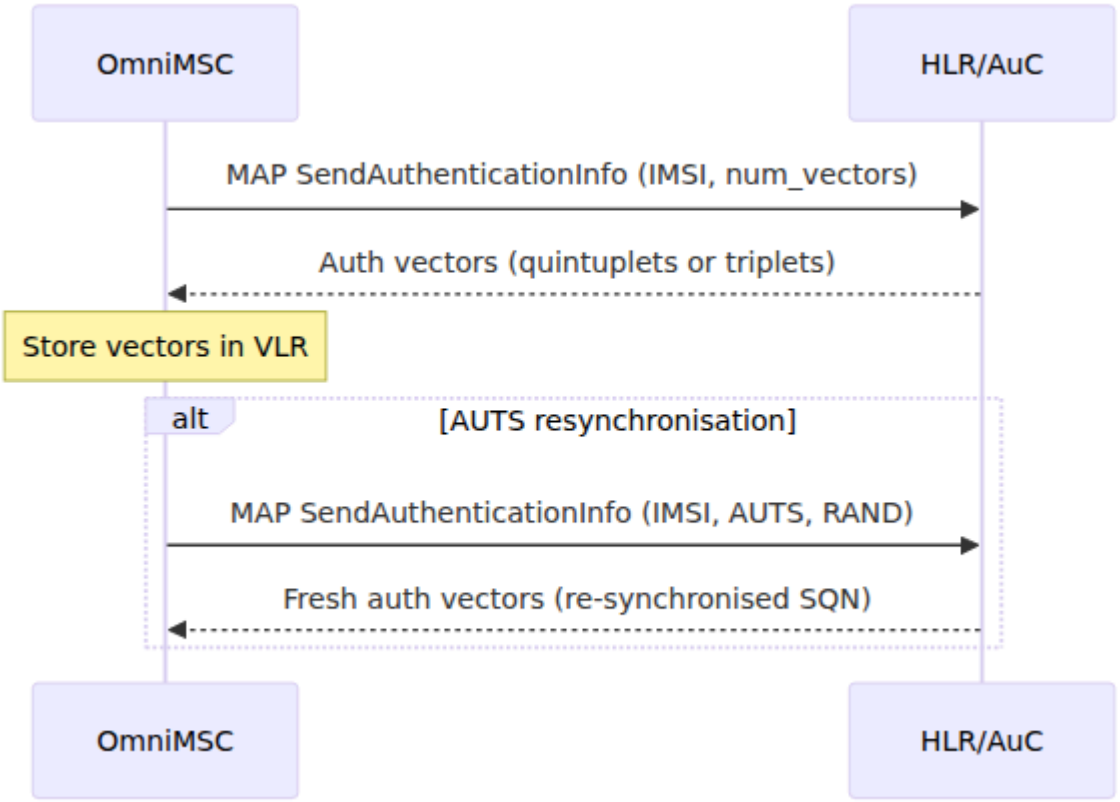
The MSC sends MAP SendAuthenticationInfo to the HLR to fetch authentication vectors for a subscriber. The request carries the IMSI, the number of requested vectors, and optionally re-synchronisation data (AUTS) if the UE reported a sequence number failure.

The HLR returns authentication vectors from the AuC: quintuplets (RAND, XRES, CK, IK, AUTN) for UMTS-capable subscribers or triplets (RAND, SRES, Kc) for GSM-only subscribers. The MSC stores the vectors in the VLR and uses them for subsequent authentication attempts without contacting the HLR again until the vector supply is exhausted.



PurgeMS

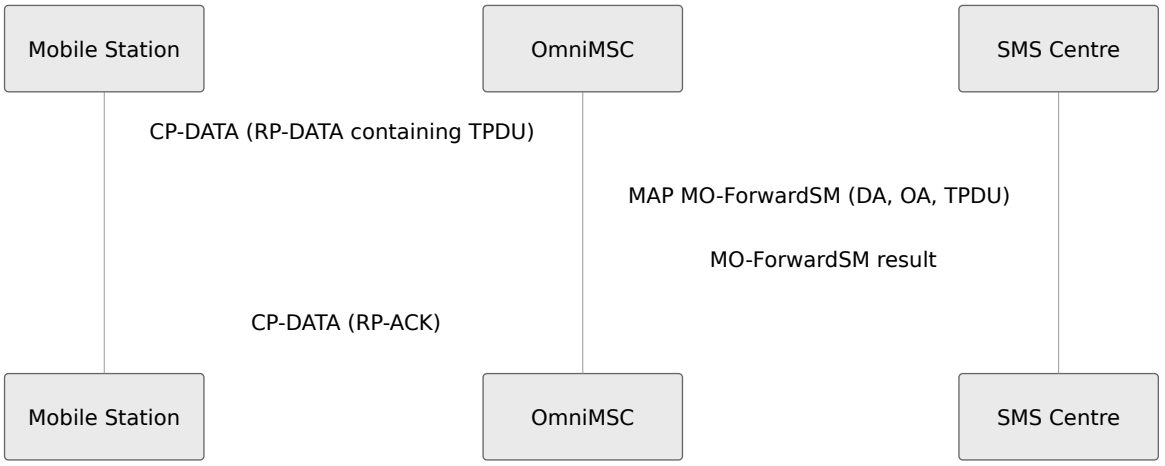
The MSC sends MAP PurgeMS to the HLR when a subscriber performs an IMSI detach. The message carries the IMSI and the VLR number. On receiving PurgeMS, the HLR clears the VLR address from the subscriber record. This ensures correct T-ADS (Terminating Access Domain Selection) routing: without a valid VLR address, the HLR knows the subscriber is not reachable via CS domain and can route MT services accordingly (e.g., trigger the MNRF flag for SMS, return absent subscriber for MT calls).



MO-ForwardSM

The MSC sends MAP MO-ForwardSM to the SMS Centre for mobile-originated short messages. The message carries the SM-RP-DA (destination, typically the SMSc address), SM-RP-OA (originator, the subscriber MSISDN), and the SM-RP-UI (the TPDU containing the SMS payload).

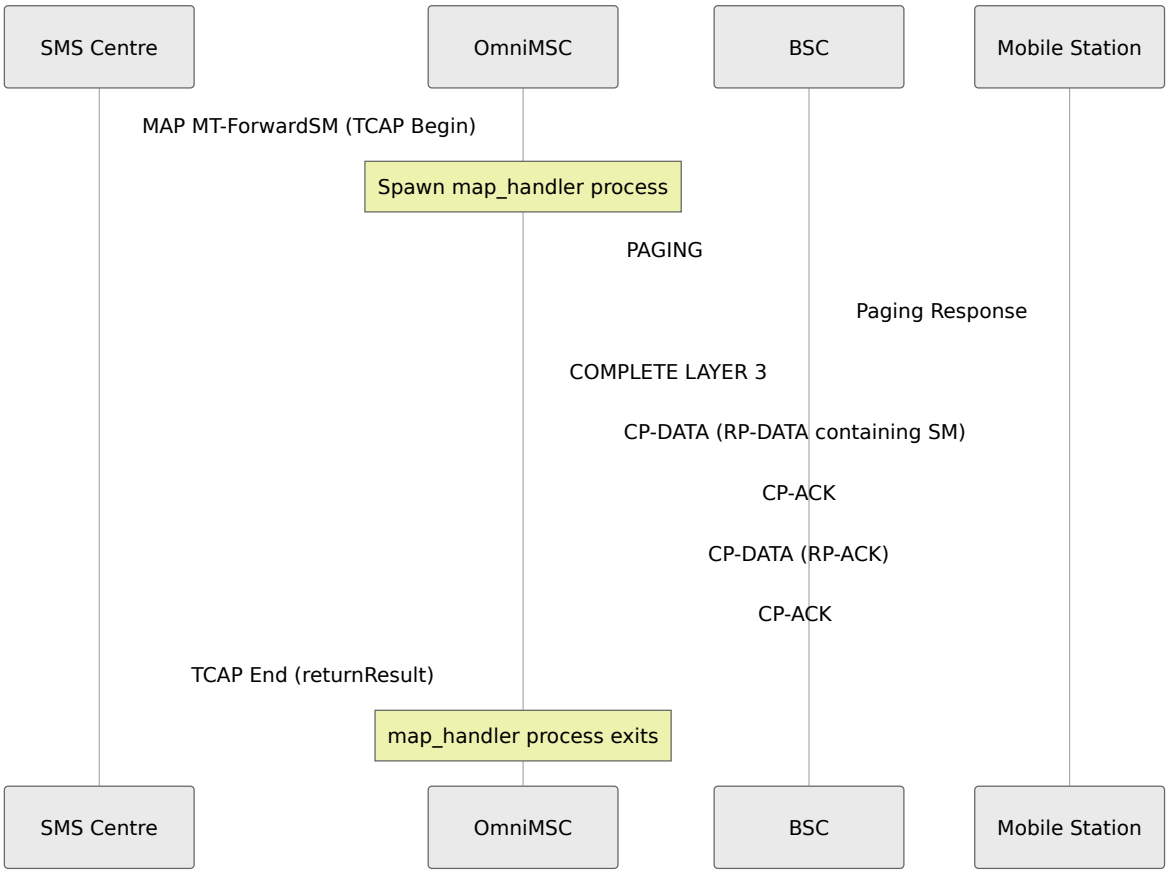
The SMSc acknowledges with a return result on successful acceptance, or a return error if the message cannot be processed (e.g., SMSc congestion, invalid destination).



MT-ForwardSM

The SMS Centre sends MAP MT-ForwardSM to the MSC for mobile-terminated short message delivery. On receiving this operation, OmniMSC spawns a dedicated map_handler process that manages the delivery attempt and holds the MAP dialogue open until the outcome is known.

The handler looks up the subscriber in the VLR, pages if necessary, and delivers the SM via DTAP. Once the mobile station acknowledges delivery (RP-ACK) or reports failure (RP-ERROR), the handler sends a TCAP End containing either a returnResult (successful delivery) or a returnError (delivery failure with cause) back to the SMSc.



If delivery fails (subscriber absent, memory exceeded, protocol error), the handler returns the appropriate MAP error:

Error	Cause
Absent Subscriber	Subscriber not reachable (not registered or no paging response)
SM Delivery Failure	MS returned RP-ERROR (memory full, unspecified error)
System Failure	Internal processing error

ProcessUnstructuredSS-Request

OmniMSC relays USSD requests to the HLR via MAP ProcessUnstructuredSS-Request when the USSD string targets a service that requires HLR processing

(e.g., supplementary service management, balance queries). The MSC forwards the USSD string and DCS (data coding scheme) to the HLR, which processes the request and returns a response string to be displayed on the mobile station.

For operator-defined USSD codes that are handled by an external USSD gateway rather than the HLR, the MSC routes the MAP dialogue to the configured gateway address instead.

TCAP Transaction Management

MAP dialogues are carried within TCAP transactions. OmniMSC maintains a transaction table that correlates local and remote transaction identifiers across the dialogue lifecycle.

TCAP Message	MAP Dialogue Phase	Description
TC-BEGIN	Opening	Initiator sends first invoke; carries originating transaction ID (otid)
TC-CONTINUE	Active	Both parties exchange components; response carries destination transaction ID (dtid) matching the peer's otid
TC-END	Closing	Final message; carries result or error components; dialogue terminated
TC-ABORT	Abort	Abnormal termination; protocol error or timeout

The MAP client tracks each active dialogue by its otid/dtid pair. When a TCAP CONTINUE or END arrives, the client matches the dtid to a local otid to locate the corresponding handler process. This correlation is essential for multiplexing many concurrent MAP dialogues over a single SCCP connection.

Dialogue Timeout

Each MAP dialogue has a configurable timeout. If the remote peer does not respond within the timeout period, the MAP client sends a TC-ABORT (locally initiated) and notifies the requesting subsystem. This prevents resource leaks from unresponsive peers.

MAP Error Handling

OmniMSC handles MAP errors at both the component level (return error within a dialogue) and the dialogue level (abort, provider error).

Common MAP Errors

Error	Typical Operation	MSC Handling
Absent Subscriber	MT-ForwardSM, SendRoutingInfo	Report delivery failure to SMSc; set MNRF in VLR
SM Delivery Failure	MT-ForwardSM	Forward TP failure cause to SMSc
Unknown Subscriber	UpdateLocation	Reject Location Update
Roaming Not Allowed	UpdateLocation	Reject Location Update with appropriate cause
System Failure	Any	Retry or report failure to requesting subsystem
Data Missing	SendAuthenticationInfo	Authentication cannot proceed; reject service
Unexpected Data Value	InsertSubscriberData	Log and reject the offending parameter

Dialogue-Level Failures

When a TCAP ABORT is received or a dialogue times out, the MAP client determines which subsystem initiated the dialogue and delivers a failure notification. The subsystem then applies its own recovery logic (e.g., the LU FSM rejects the Location Update, the SMS handler returns a delivery failure to the SMSc).

Point Code Routing

When OmniMSC receives an incoming MAP dialogue (e.g., MT-ForwardSM from the SMSc, InsertSubscriberData from the HLR), the M3UA layer records the originating point code from the incoming message's routing information. When the MSC sends the response (TCAP Continue or End), it uses this recorded point code as the destination point code (DPC) for the outgoing M3UA message.

This mechanism ensures that responses are routed back to the correct originating node, even when multiple HLRs or SMSCs with different point codes are reachable via the same STP. The routing_info from the incoming M3UA association is stored per-dialogue and used for all subsequent messages within that dialogue.

3GPP Specification References

Specification	Title	Relevance
TS 29.002	Mobile Application Part (MAP) Specification	All MAP operations, error codes, dialogue procedures
ITU-T Q.771-Q.775	Transaction Capabilities Application Part (TCAP)	TCAP transaction management, component handling
ITU-T Q.711-Q.716	Signalling Connection Control Part (SCCP)	SCCP addressing and routing for MAP dialogues
RFC 4666	MTP3 User Adaptation Layer (M3UA)	M3UA transport, point code routing

Media Control

This document describes media gateway control in OmniMSC, covering MGCP and Megaco/H.248 gateway protocols, the Media Controller state machine, codec negotiation, conference bridging, and SDP generation.

For call flow diagrams showing media setup in context, see Call Flow Diagrams. For SIP trunk codec configuration, see SIP Trunking. For media gateway configuration parameters, see Configuration Reference. For MPTY conference bridging from the supplementary services perspective, see Supplementary Services.

MGCP Gateway Control

OmniMSC controls media gateways using the Media Gateway Control Protocol (MGCP) per RFC 3435. The MGCP Call Agent issues three primary commands to manage RTP connections on the gateway.

Commands

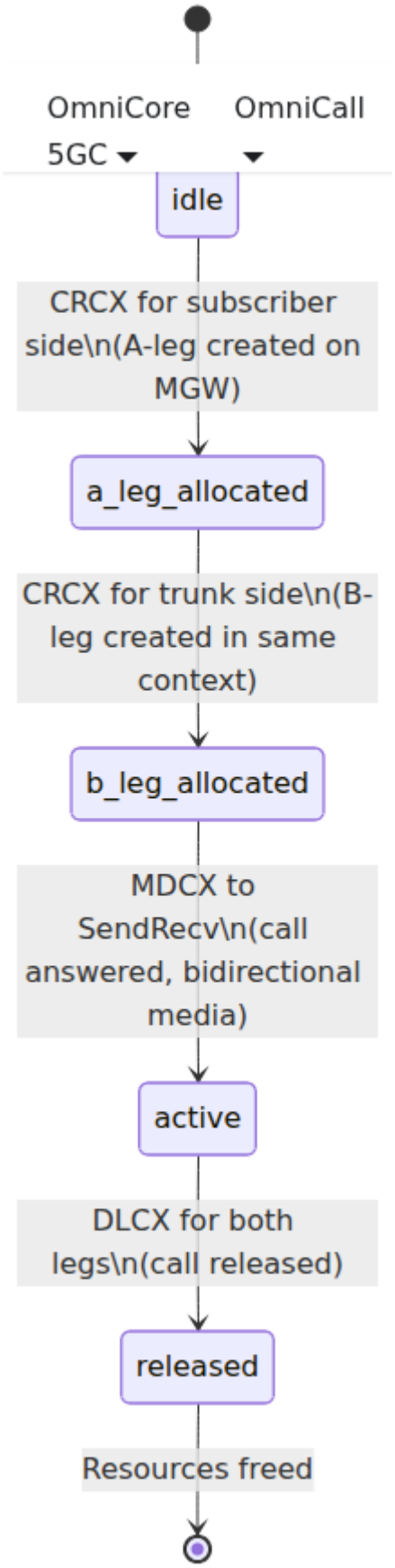
Command	Purpose	Description
CRCX	Create Connection	Allocates a new RTP endpoint on the media gateway and returns the local IP address, port, and SDP for that endpoint.
MDCX	Modify Connection	Updates an existing connection's parameters, such as switching the media mode from receive-only to send-and-receive when a call is answered.
DLCX	Delete Connection	Tears down an RTP connection and releases the associated endpoint resources on the gateway.

Endpoint Naming

Media gateway endpoints follow the naming convention `rtpbridge/N@mgw`, where `N` is a sequentially assigned endpoint number and `mgw` is the gateway domain name. Each call leg occupies one endpoint within a connection context on the gateway.

Media Controller States

The Media Controller manages the lifecycle of media resources for each call. It progresses through the following states as the call is established and released.


OmniCore 5GC
OmniCall
idle
CRCX for subscriber side\n(A-leg created on MGW)
a_leg_allocated
CRCX for trunk side\n(B-leg created in same context)
b_leg_allocated
MDCX to SendRecv\n(call answered, bidirectional media)
active
DLCX for both legs\n(call released)
released
Resources freed

A-Leg (Subscriber Side)

The A-leg represents the media path between the media gateway and the subscriber's radio access bearer via the BSC or RNC.

When a call is initiated, the Media Controller sends a CRCX command to the media gateway to allocate the A-leg endpoint. The initial connection mode is set to **RecvOnly**, meaning the gateway accepts incoming media from the subscriber side but does not transmit. This prevents clipping and echo during the alerting phase.

When the call is answered, the Media Controller sends an MDCX command to change the A-leg mode to **SendRecv**, enabling bidirectional media flow.

B-Leg (Trunk Side)

The B-leg represents the media path between the media gateway and the remote trunk (SIP peer or ISUP circuit).

The B-leg is allocated via a second CRCX command within the same MGW connection context as the A-leg. By placing both legs in the same context, the media gateway internally bridges the two RTP streams, enabling voice path continuity without hairpinning media through the MSC.

Codec Negotiation

OmniMSC negotiates speech codecs based on the capabilities reported by the BSC in the BSSMAP Assignment Request response (speech version list) and the codec preferences of the outbound trunk.

Supported Codecs

Codec	RTP Payload	Description
AMR-FR	dynamic	Adaptive Multi-Rate Full Rate, octet-aligned mode (octet-align=1)
GSM-EFR	dynamic	GSM Enhanced Full Rate (6.60 kbit/s)
GSM-FR	3	GSM Full Rate (13.0 kbit/s)
AMR-HR	dynamic	Adaptive Multi-Rate Half Rate
GSM-HR	dynamic	GSM Half Rate (5.60 kbit/s)

The codec offer list is constructed from the intersection of the BSC's reported speech version capabilities and the MSC's configured codec preferences. The resulting SDP is included in the outbound SIP INVITE to the trunk peer, allowing the remote side to select the preferred codec from the offered set.

Conference Bridge (Multi-Party)

OmniMSC supports multi-party (MPTY) conference calls per 3GPP TS 24.083 by leveraging the media gateway's conference bridging capability.

When a subscriber invokes the MPTY supplementary service, the Media Controller allocates a conference context on the media gateway. Individual call legs are then added to or removed from the conference as participants join or leave.

Conference Operations

Operation	Description
Create Conference	Allocates a new conference context on the media gateway with a dedicated mixing endpoint.
Add to Conference	Moves an existing call leg into the conference context via MDCX, connecting the participant to the conference mixer.
Remove from Conference	Moves a call leg out of the conference context back to a point-to-point connection.

The conference mixer on the media gateway handles media mixing for all participants. The MSC controls conference membership through signalling only and does not process media directly.

SDP Generation

OmniMSC generates SDP (Session Description Protocol) for outbound SIP messages using the following conventions:

SDP Field	Value
Origin (o=)	`OmniMSC` as the session originator username
Connection (c=)	IP address returned by the media gateway in the CRCX response
Media (m=)	RTP port returned by the media gateway, with codec payload types from the negotiated offer
Attributes (a=)	Codec-specific parameters such as `fmtp` for AMR octet-align mode

The connection address and port in the SDP always reflect the media gateway's allocated endpoint, ensuring that RTP flows directly between the media gateway and the remote peer without traversing the MSC signalling path.

Megaco/H.248 Support

As an alternative to MGCP, OmniMSC supports media gateway control via the Megaco/H.248 protocol per ITU-T Recommendation H.248. The choice between MGCP and Megaco is configurable per media gateway via the media mode setting.

Megaco uses a transaction-based command model with Add, Modify, Subtract, and Move commands that map conceptually to the MGCP CRCX, MDCX, and DLCX operations. The Megaco Controller manages H.248 transactions, context allocation, and termination lifecycle on the gateway.

Both MGCP and Megaco transports use UDP by default. The Media Controller abstracts the protocol differences so that the CC FSM and other call-handling components interact with a unified media control interface regardless of the underlying gateway protocol.

Gateway Configuration

Each media gateway is defined with the following identity parameters:

Parameter	Description
Name	Logical gateway name used for identification in logs and the control panel
Address	IP address of the media gateway
Port	MGCP or Megaco listening port on the gateway
Domain	Gateway domain name used in MGCP endpoint addressing (the `@mgw` portion of endpoint names)

For the full set of media configuration parameters, see Configuration Reference.

References

- **RFC 3435** -- Media Gateway Control Protocol (MGCP) Version 1.0
- **ITU-T H.248** -- Gateway Control Protocol (Megaco)
- **3GPP TS 24.083** -- Call Waiting and Call Hold supplementary services (MPTY)
- **RFC 4566** -- Session Description Protocol (SDP)

Metrics and Monitoring

This document describes the telemetry, metrics, alarms, and health endpoints provided by OmniMSC. For overload threshold configuration, see Configuration Reference. For troubleshooting alert conditions, see Troubleshooting Guide. For the real-time dashboard view of active calls and subscriber count, see Control Panel Guide.

Telemetry Overview

OmniMSC emits Erlang/Elixir telemetry events for all significant operational activities. These events are exported as Prometheus metrics, available at the `/metrics` endpoint on the Phoenix HTTP port. All metric names are namespaced under `omnimsc_` to avoid collisions with other applications. The System page in the control panel provides a real-time view of BEAM VM statistics including process count, memory, and scheduler load — see Control Panel Guide.

Metric definitions are declared in `Omnimsc.Telemetry.Metrics.Prometheus.metrics/0`. Any Prometheus-compatible scraper (Prometheus, Grafana Agent, Datadog, Victoria Metrics) can collect these metrics at the standard scrape interval.

Metrics Reference

Metric	Type	Labels	
`omnimsc_active_calls_count`	Gauge	--	Cu voi
`omnimsc_vlr_subscribers_count`	Gauge	--	Sul reg
`omnimsc_sccp_connections_count`	Gauge	--	Act (A/
`omnimsc_sms_sent_count`	Counter	--	Tot ser
`omnimsc_location_update_complete_count`	Counter	`type`	Loc cor nor
`omnimsc_auth_failure_count`	Counter	`reason`	Aut (m syr
`omnimsc_auth_skipped_count`	Counter	--	Aut exi cor
`omnimsc_handover_attempt_count`	Counter	`type`	Ha (int int
`omnimsc_paging_attempt_count`	Counter	`result`	Pag (di tim

Metric	Type	Labels	
`omnimsc_peer_status`	Gauge	`peer`	SIP (1=
`omnimsc_ss_operation_count`	Counter	`operation`, `ss_service`	Sup ope
`omnimsc_ss_error_count`	Counter	`reason`	SS
`omnimsc_ussd_request_count`	Counter	`routing`	US hlr
`omnimsc_map_dialogue_duration`	Histogram	`operation`	MA tim
`omnimsc_call_release_count`	Counter	`type`	Cal

Label Values

omnimsc_location_update_complete_count -- the `type` label distinguishes location update types per 3GPP TS 24.008:

Value	Description
`imsi_attach`	IMSI attach (subscriber powering on)
`normal`	Normal location update (subscriber moved to new location area)
`periodic`	Periodic location update (T3212 timer expiry)

omnimsc_auth_failure_count -- the `reason` label identifies the failure cause:

Value	Description
`mac_failure`	SRES/RES mismatch -- MS response does not match expected value
`sync_failure`	SQN out of range, resynchronization needed
`timeout`	Authentication timer (T3260) expired without response

omnimsc_paging_attempt_count -- the `result` label tracks paging outcomes:

Value	Description
`dispatched`	Paging sent to BSC(s)
`success`	Subscriber responded to paging
`timeout`	Max retries exhausted without response

omnimsc_peer_status -- the `peer` label identifies the remote peer by its configured name (e.g., Default-GW, International-GW, MSC-02).

omnimsc_ss_operation_count -- the `operation` label identifies the SS operation (register, erase, activate, deactivate, interrogate) and the `ss_service` label identifies the target service (cfu, cfb, cfnry, cfnrc, cw, clip, clir, baoc, baoic).

omnimsc_ussd_request_count -- the `routing` label distinguishes between locally handled SS requests and those relayed to the HLR:

Value	Description
`local_ss`	Request handled locally by the MSC
`hlr_relay`	Request relayed to the HLR via MAP

omnimsc_call_release_count -- the `type` label distinguishes call direction:

Value	Description
`mo`	Mobile-originated call released
`mt`	Mobile-terminated call released

Example PromQL Queries

The following queries are useful starting points for dashboards and alerting rules.

Active call monitoring -- current call load on the MSC:

omnimsc_active_calls_count

Call rate -- calls released per second, averaged over five minutes:

rate(omnimsc_call_release_count[5m])

Auth failure ratio -- authentication failures per second by reason:

rate(omnimsc_auth_failure_count[5m])

Peer availability -- identify any peers that are currently down:

omnimsc_peer_status

SMS throughput -- SMS messages per second:

rate(omnimsc_sms_sent_count[5m])

Location update rate by type -- breakdown of LU activity:

sum by (type) (rate(omnimsc_location_update_complete_count[5m]))

SS operation rate by service -- supplementary service activity:

sum by (ss_service) (rate(omnimsc_ss_operation_count[5m]))

USSD routing breakdown -- local vs HLR-relayed USSD requests:

sum by (routing) (rate(omnimsc_ussd_request_count[5m]))

Alarm System

OmniMSC raises and clears alarms for conditions that require operator attention. Each alarm has a severity level and a unique identifier.

Alarm Types

Alarm	Severity	Description
`sctp_link_down`	Critical	SCTP association to STP lost
`hlr_unreachable`	Critical	HLR not responding to MAP operations
`cdr_write_failure`	Major	CDR file write error
`overload`	Major	System overload threshold exceeded

Alarm Telemetry Events

The alarm subsystem emits telemetry events that can be consumed by external monitoring systems or attached to Prometheus metrics:

Event	Description
`[:omnimsc, :alarm, :raised]`	Emitted when an alarm condition is detected. Metadata includes alarm_id, severity, source, and descriptive text.
`[:omnimsc, :alarm, :cleared]`	Emitted when an alarm condition is resolved. Metadata includes alarm_id, severity, and source.

Alarms remain active until the underlying condition is resolved, at which point the cleared event is emitted. Multiple raises of the same alarm_id without an intervening clear are deduplicated.

Health Endpoint

OmniMSC exposes a health check endpoint for use by load balancers and orchestration systems.

GET /api/health returns the overall system health status. The response indicates whether the MSC is operational and accepting traffic. A healthy response confirms that core subsystems (VLR, CC, MAP client, SIP stack) are running. An unhealthy response indicates that one or more critical subsystems have failed.

This endpoint is suitable for Kubernetes liveness and readiness probes, or for load balancer health checks in traditional deployments.

Status Endpoint

GET /api/status returns detailed system information including active call count, registered subscriber count, peer link states, alarm summary, BEAM process count, and uptime. This endpoint provides a comprehensive snapshot for operational dashboards and diagnostic purposes.

The status response includes all the information needed to assess system capacity and identify degraded components without requiring Prometheus access.

Overload Protection

OmniMSC includes a configurable overload protection mechanism that prevents the system from exceeding safe operating limits. The overload module continuously monitors four metrics and compares them against configurable thresholds.

Overload Thresholds

Metric	Default Threshold	Description
Active calls	10,000	Maximum concurrent CS calls
Registered subscribers	50,000	Maximum subscribers in the VLR
BEAM process count	500,000	Maximum Erlang processes
Paging rate	1,000/sec	Maximum paging requests per second

When any threshold is exceeded, the overload module rejects new service requests with GSM cause 42 (switching equipment congestion). Calls already in progress are not affected. The overload state is reflected in the `[:omnimsc, :overload, :state_change]` telemetry event and the `overload` alarm.

Overload protection applies to location updates, call setup requests, and SMS transactions. Emergency calls bypass overload protection regardless of system load, per 3GPP TS 22.101.

For threshold configuration, see Configuration Reference.

MSC Pool and NRI

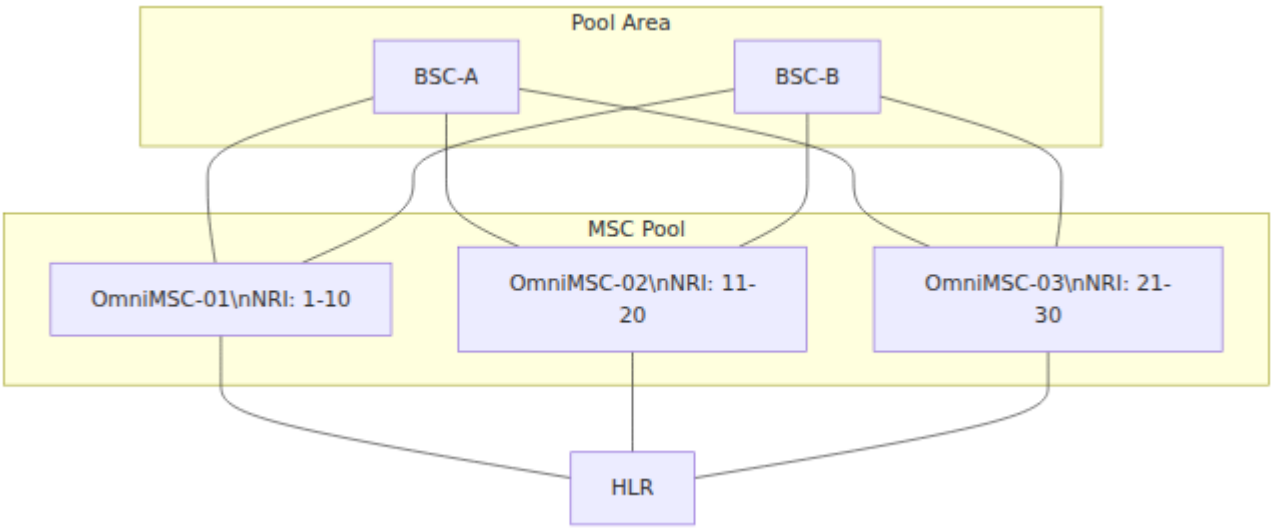
This document describes the MSC-in-Pool architecture implemented by OmniMSC by Omnitouch per 3GPP TS 23.236. Pooling allows multiple MSC servers to share a common pool area, providing load distribution across MSCs and resilience against individual MSC failures.

For pool-aware routing behavior, see Routing. For the Pool page in the web interface, see Control Panel Guide. For configuration parameters, see Configuration Reference. For TMSI allocation details including the no-rollback design and NRI embedding, see Security.

MSC-in-Pool Concept

In a traditional architecture, each BSC connects to a single MSC. If that MSC fails, all subscribers served by its BSCs lose service. MSC pooling addresses this limitation by allowing BSCs to connect to multiple MSCs simultaneously via the A-Flex interface. Any MSC in the pool can serve any subscriber arriving from any BSC in the pool area.

The key mechanism enabling pool operation is the Network Resource Identifier (NRI), a bit field embedded in the TMSI that identifies which MSC allocated that TMSI. When a subscriber presents a TMSI to a BSC, the BSC extracts the NRI and routes the signalling to the correct MSC. If that MSC is unavailable, the BSC selects an alternative MSC from the pool, and the receiving MSC handles the subscriber locally.



Each BSC maintains SCTP associations to every MSC in the pool. For returning subscribers, the BSC uses the NRI in the TMSI to select the MSC that originally registered the subscriber. For new subscribers (no TMSI or null NRI), the BSC uses round-robin or load-based selection.

Network Resource Identifier (NRI)

The NRI is encoded within the 32-bit TMSI allocated by the MSC. Per 3GPP TS 23.236, the NRI field is placed immediately after the two most significant reserved bits of the TMSI. The length of the NRI field is configurable and must be identical across all pool members.

TMSI Bit Layout

Bits 31-30	Bits 29-20	Bits 19-0
2 bits	10 bits	20 bits
Reserved	NRI	Random

The default NRI bit length is 10, supporting up to 1024 distinct NRI values. Smaller deployments may use fewer bits.

NRI Bit Length	Maximum NRI Values	Remaining TMSI Bits
5	32	25
8	256	22
10	1024	20

NRI value 0 is reserved as the "null NRI" and indicates that the TMSI was not assigned by any pool member. A subscriber presenting a null NRI is treated as a new subscriber and accepted locally with a fresh TMSI allocation.

TMSI Allocation

When pool mode is active, OmniMSC embeds one of its assigned NRI values into every TMSI it allocates. The allocation process generates a random 32-bit base TMSI and then overwrites the NRI bit field with the MSC's designated NRI value. This ensures that any BSC in the pool area can determine which MSC owns a given subscriber by inspecting the TMSI. For general TMSI allocation and confirmation behavior (including the no-rollback design choice), see Security.

The MSC uses its first configured NRI value as the default for new allocations. All NRI values assigned to the MSC are recognized as "local" when evaluating incoming TMSIs.

Configuration

Pool membership is configured under the pool key in the MSC configuration (see Configuration Reference). The essential parameters are:

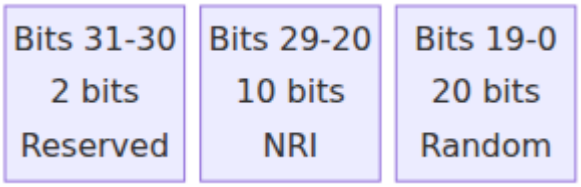
Parameter	Default	Description
pool_id	nil (pooling disabled)	Pool identifier. All MSCs in the same pool must share the same pool_id. Set to nil to operate in standalone (non-pooled) mode.
nri_bitlength	10	Number of bits used for the NRI field in the TMSI. Must be identical across all pool members. Valid range: 1 through 15 per TS 23.236 Section 5.
nri_values	(empty list)	List of NRI values owned by this MSC instance. TMSIs allocated by this MSC will contain one of these values. NRI ranges must not overlap between pool members.
members	(empty list)	List of other MSC instances in the pool. Each member entry includes a logical name, SS7 point code, and assigned NRI values. Used for NRI-based subscriber relaying and health monitoring.

Each member entry in the members list requires:

Parameter	Description
name	Logical name of the remote pool member
point_code	SS7 point code of the remote MSC, used for MAP/E-interface signalling
nri_values	NRI values assigned to the remote member, used to identify which MSC owns a given TMSI

Foreign NRI Handling

When a subscriber presents a TMSI containing an NRI that belongs to a different pool member, the receiving MSC must decide how to handle the subscriber. This situation arises when the BSC's NRI-based selection function (NNSF) routes to the wrong MSC, or when the owning MSC is unavailable.



The Pool Manager always accepts the subscriber locally and reallocates the TMSI with a local NRI value. The decision on how to resolve the subscriber's identity depends on the owning MSC's health status:

Owning MSC Status	Behavior
Reachable	Send MAP SendIdentification to the owning MSC to retrieve the IMSI and authentication vectors, then continue the location update with the received identity
Unreachable	Request the IMSI directly from the UE via an Identity Request, then fetch authentication vectors from the HLR
Unknown	Same behavior as unreachable; request the IMSI from the UE

In all cases, the subscriber is re-registered at the receiving MSC with a new TMSI containing a local NRI value.

Pool Member Health Tracking

Each MSC in the pool monitors the health of other pool members. The health state is tracked per member and directly affects foreign NRI handling behavior.

Member State	Description	Effect on Foreign NRI Handling
Up	Member is responding to health probes	Foreign NRI subscribers are identified via MAP SendIdentification to the owning MSC
Down	Member has not responded within the timeout period	Foreign NRI subscribers are identified by requesting the IMSI from the UE
Unknown	Initial state before health probes have completed	Treated the same as Down
Draining	Member has announced a drain for maintenance	No new subscribers are relayed; existing sessions are maintained until completion

When a pool member restarts, it broadcasts MAP Reset to all pool members. Receiving MSCs mark all subscriber records originally belonging to the reset MSC for re-registration on next contact. Health state changes are broadcast to the Control Panel via PubSub for real-time visibility.

Rolling Upgrade Procedure

Pool architecture enables zero-downtime upgrades by draining and upgrading one MSC at a time.

1. Set the target MSC to draining state via the Control Panel or API. BSCs stop sending new subscribers to this MSC; existing sessions continue.
2. Wait for active calls to complete naturally. Monitor the active call count on the dashboard.
3. Issue Clear Command to any remaining BSC connections.

4. Stop the MSC process. Remaining subscribers will re-register on other pool members at next contact.
5. Apply the software update or configuration change.
6. Start the upgraded MSC. It establishes SCTP associations with BSCs and announces itself via MAP Reset.
7. The MSC begins accepting new subscribers. BSCs include it in their selection algorithm.
8. Verify health on the Pool page: member state is Up, NRI allocation is correct, subscribers are registering.

Repeat for each MSC in the pool.

Pool LiveView Page

The Pool page in the Control Panel displays:

- Pool identity: pool ID, local NRI values, NRI bit length.
- Member status table: name, point code, NRI range, health state (Up, Down, or Draining), last probe timestamp, and subscriber count per member.
- NRI distribution chart: visual distribution of subscribers across NRI ranges.
- Foreign NRI events: recent foreign NRI encounters showing the TMSI presented, owning MSC, and outcome (relayed via MAP SendIdentification, resolved via IMSI request, or failed).
- Pool statistics: total pool subscribers, local versus foreign NRI ratio, relay success rate.

3GPP Specification References

Specification	Title	Relevance
TS 23.236	Intra-domain connection of RAN nodes to multiple CN nodes	MSC Pool architecture, NRI format, BSC selection
TS 23.012	Location Management Procedures	VLR interaction with pool operation
TS 29.002	MAP Specification	MAP SendIdentification, MAP Reset for pool coordination
TS 48.008	BSC-MSC Interface (A-Flex)	A-Flex extensions for pool-aware BSSAP signalling

Routing

This document describes how OmniMSC by Omnitouch analyses called numbers, selects routes, and delivers calls to their destinations. It covers the number analysis pipeline, the route table, supported destination types, the GMSC function, and route management interfaces.

For call flow diagrams showing how routing integrates with call setup, see Call Flow Diagrams. For route table configuration parameters, see Configuration Reference. For SIP peer health monitoring, see SIP Trunking. For ISUP trunk group details, see ISUP Trunking. For route management via the control panel, see Control Panel Guide.

Number Analysis

When a call is initiated, OmniMSC classifies the called number to determine its type and normalize it to E.164 format. The classification follows a priority order and uses the routing configuration for the deployment (country code, national prefix, international prefix, emergency numbers, and short codes).

Classification Order

The number analysis engine evaluates the called number against the following categories in order. The first match wins.

Priority	Number Type	Detection Rule	Normalization
1	Emergency	Exact match against the configured emergency number list (e.g., 112, 911, 000)	Unchanged; the route hint is set to :emergency
2	Short code	Exact match against the configured short code table, where each code maps to a service type (operator assistance, directory enquiry, etc.)	Unchanged; the route hint is set to the service type atom
3	International	Number begins with "+" or with the configured international prefix (default "00")	Stripped of the international prefix and prepended with "+" to produce E.164
4	National	Number begins with the configured national prefix (default "0")	National prefix is stripped, then the country code is prepended with "+" to produce E.164
5	Local	Any remaining number that does not match the above	Country code and area code are prepended with "+" to produce E.164

After classification, the normalized E.164 number is passed to the route table for longest-prefix matching.

Route Table

The route table is an ETS-backed data structure that maps number prefixes to destinations. It is the central decision point for all outbound call routing in OmniMSC.

Longest-Prefix Matching

When a number is presented for routing, the route table iterates from the full number length down to a single digit, checking for a matching prefix at each step. The first (longest) match found is used. If no prefix-specific match is found, the table falls back to the default route (prefix is the empty string). If no default route exists, the call fails with a no-route-to-destination error.

Priority Ordering

Each route entry carries a numeric priority value. Higher values take precedence. Priority is used when the route table is displayed and when managing overlapping entries. Emergency routes should be configured with the highest priority (e.g., 100) to ensure they are never shadowed by less-specific entries.

Example Route Table

The following table illustrates how the route table resolves different called numbers.

Prefix	Type	Destination	Priority	Purpose
000	:sip	SIP peer "Default-GW"	100	Emergency number — route must match `psap_address`
04	:local	VLR subscriber	50	Australian mobile numbers
0412	:sip	SIP peer "Mobile-GW"	50	Specific mobile prefix routed to a SIP gateway
001	:sip	SIP peer "International-GW"	10	International dialing prefix
07	:isup	Trunk group "Mobile-Interconnect"	10	Mobile interconnect via SS7
08	:sip_with_failover	Primary: SIP peer "Primary-SIP-GW", Failover: ISUP trunk "Backup-ISUP"	10	SIP with automatic ISUP fallback
09	:sip_i	SIP-I peer "MSC-02-SIP-I"	10	SIP with encapsulated ISUP to a peer MSC

Prefix	Type	Destination	Priority	Purpose
(empty)	:sip	SIP peer "Default-GW"	1	Catch-all default route

With this table in place:

Dialed Number	Matching Prefix	Destination	Reason
000	000	SIP: Default-GW	Exact match on emergency prefix
0412345678	0412	SIP: Mobile-GW	Longest match (4 digits beats the 2-digit "04" entry)
0498765432	04	Local VLR subscriber	Matches "04" but not "0412"
0011234567	001	SIP: International-GW	Longest match (3 digits)
0312345678	(empty)	SIP: Default-GW	No prefix match; falls to default

---

Route Types

OmniMSC supports the following destination types in the route table.

Emergency Calls

Emergency calls are not a separate route type. The MSC detects emergency calls from the CC Emergency Setup message type (3GPP TS 24.008 §9.3.8, message type 0x0E) and the CM Service Request type (`:emergency`). Authentication is attempted but the call proceeds regardless of the outcome. Ciphering is established if authentication succeeds; otherwise the call proceeds without it.

Emergency Setup messages do not carry a Called Party BCD Number — the handset sends only an optional Bearer Capability IE and Emergency Service Category IE. The MSC uses the `psap_address` from the Emergency configuration as the called number for route table lookup. This number is then routed through the normal route table like any other call. The route entry it matches can be any type (`:sip`, `:isup`, `:sip_i`, etc.).

Example: To route emergency calls to a SIP peer named "Default-GW":

```
# Route table — psap_address "000" will match this entry
%{prefix: "000", type: :sip, peer: "Default-GW", priority: 100}

# Emergency config — psap_address is used as the called number
config :omnimsc, Omnimsc.Emergency,
  psap_address: "000"
```

If authentication fails, the caller's IMEI is used as the calling party number instead of the MSISDN. Emergency numbers are also detected during number analysis and can trigger emergency handling even before the route table is consulted.

:local

Routes the call to a subscriber registered in the local VLR. The MSC looks up the called MSISDN in the VLR, pages the subscriber via the appropriate BSC or RNC, and establishes the call over the A-interface or Iu-CS interface.

:sip

Routes the call to a configured SIP peer by sending a SIP INVITE. The route entry specifies the peer by name; the peer's IP address, port, transport, and codec configuration are resolved from the SIP peer table. The Trunk Router verifies that the peer is reachable (status is not "down") and that the peer has available channels before routing.

:isup

Routes the call via an SS7 ISUP trunk group. The route entry specifies the trunk group name, destination point code, and CIC (Circuit Identification Code) range. The Trunk Router allocates an idle circuit from the trunk group, constructs an Initial Address Message (IAM), and sends it via the M3UA/SCTP transport to the remote exchange.

:sip_i

Routes the call to a SIP-I peer, where the full ISUP message is encapsulated within the SIP body. SIP-I preserves all ISUP information elements during interworking, avoiding the information loss that can occur with standard SIP-ISUP conversion. For protocol details, see SIP-I Trunking.

:sip_with_failover

Attempts the call via a primary SIP peer first. If the SIP peer is unreachable, returns a 5xx error, or times out, the Trunk Router automatically retries the call via a configured ISUP failover trunk group. This destination type requires both a SIP peer name (primary) and an ISUP trunk group with point code (failover).

:gmsc

Invokes the Gateway MSC function. The MSC sends a MAP SendRoutingInfo request to the HLR to obtain the MSRN (Mobile Station Roaming Number) for the called subscriber, then routes the call to the serving MSC using the returned MSRN. See the GMSC Function section below for details.

:transit

Routes the call as a transit call between trunk types without instantiating a full Call Control FSM. Transit routes are used for ISUP-to-SIP gateway operation, SIP-to-ISUP interworking, ISUP-to-ISUP tandem switching, and SIP-to-SIP proxy transit. Transit routes include source context (the incoming trunk type and name) to match incoming calls from specific trunks.

Route Table Management

Web UI

The Routes and Trunks page in the OmniMSC Control Panel provides a tabbed interface for managing routes, ISUP trunk groups, and SIP peers. From the Route Table tab, operators can add, edit, and delete routes through modal forms. Route changes take effect immediately without requiring a restart. Routes loaded from configuration at startup can be overridden at runtime. For details on the web interface, see Control Panel Guide.

REST API

Routes can also be managed programmatically through the REST API.

Method	Endpoint	Description
GET	/api/routes	List all routes in the table
POST	/api/routes	Add a route. Request body includes prefix, type, peer (for SIP types), and priority.
DELETE	/api/routes/:prefix	Remove a route by its prefix

Route Lookup Testing

The route lookup endpoint allows operators and integration systems to test routing decisions without placing a call.

Method	Endpoint	Description
GET	/api/routes/lookup/:number	Returns the destination that would be selected for the given called number, or a no-route indication if no match exists

---

GMSC Function

When OmniMSC receives a call for a mobile subscriber who is not registered in the local VLR, it can act as a Gateway MSC (GMSC) per 3GPP TS 23.018. The GMSC function bridges the gap between the calling network and the visited MSC where the subscriber is currently registered.

MT Call Routing Flow

1. An incoming call arrives from a PSTN trunk or SIP gateway with the subscriber's MSISDN as the called number.
2. The route table lookup returns the :gmsc destination type for this prefix.
3. OmniMSC sends a MAP SendRoutingInfo (SRI) request to the subscriber's HLR, providing the called MSISDN.
4. The HLR identifies the serving VLR and instructs it to allocate a Mobile Station Roaming Number (MSRN).
5. The HLR returns the MSRN to OmniMSC in the SRI response.
6. OmniMSC routes the call to the serving MSC using the MSRN, either via an ISUP IAM or a SIP INVITE depending on the interconnect type.
7. The serving MSC pages the subscriber and completes the MT call setup.

MSRN Pool

OmniMSC maintains a pool of MSRNs for allocation during MT call routing. When a subscriber is paged at the local MSC, an MSRN is allocated from the pool, associated with the subscriber's IMSI, and returned to the querying GMSC. The MSRN is released back to the pool once the incoming call arrives or the allocation times out.

Pool-Aware Routing

When MSC pool mode is active (see MSC Pool and NRI), the routing module considers the NRI embedded in the subscriber's TMSI during MT call routing. If a subscriber's TMSI contains a foreign NRI (belonging to another pool member), the MSC can relay the subscriber to the owning MSC via MAP SendIdentification, re-register the subscriber locally if the owning MSC is unreachable, or route MT calls to the owning MSC if the subscriber has not yet re-registered locally.

NRI-based routing is automatic when pool mode is enabled and does not require explicit route table entries.

Number Analysis Flow

The full routing decision flow, from called number to destination, proceeds through the following stages:

1. Call barring check -- if the subscriber is barred from this call type, the call is rejected with a GSM cause code.
2. CAMEL trigger check -- if a CAMEL service key matches, an InitialDP is sent to the SCP. The SCP may modify the called number, connect to a different destination, or release the call.
3. Number analysis -- the called number is classified and normalized as described above.
4. Route table lookup -- longest-prefix match against the route table.

5. Destination dispatch -- the call is handed to the appropriate handler based on the matched destination type.
6. Failover (if applicable) -- for :sip_with_failover routes, a failed SIP attempt triggers automatic retry via the ISUP fallback trunk.

ISUP Trunk Groups

Each ISUP trunk group represents a bundle of voice circuits to a remote SS7 exchange. Trunk groups are identified by name and configured with a destination point code and CIC range. When routing selects an ISUP trunk group, the Trunk Router allocates the lowest available idle circuit and sends an IAM.

Trunk groups support priority levels: primary (first choice), overflow (used when primary circuits are exhausted), and last-resort. Only active trunk groups are selected for routing; inactive or congested trunk groups are skipped.

Each trunk group tracks seizure, answer, busy, congestion, and release counters for operational monitoring.

SIP Peer Selection

SIP peers represent remote VoIP gateways, IMS nodes, or SIP trunking providers. Each peer is configured with an address, port, transport (UDP, TCP, or TLS), supported codecs, and maximum concurrent channels.

Peer health is monitored via periodic SIP OPTIONS keepalives. If a peer stops responding, its status transitions to "down" and the peer is excluded from routing. When a route specifies a SIP peer by name, the Trunk Router verifies the peer is reachable and has available capacity before routing the call.

3GPP Specification References

Specification	Title	Relevance
TS 23.018	Basic Call Handling	GMSC function, MT call routing, number analysis
TS 29.002	MAP Specification	MAP SendRoutingInfo, MSRN allocation
TS 23.078	CAMEL Phase 4	CAMEL trigger handling in the routing flow

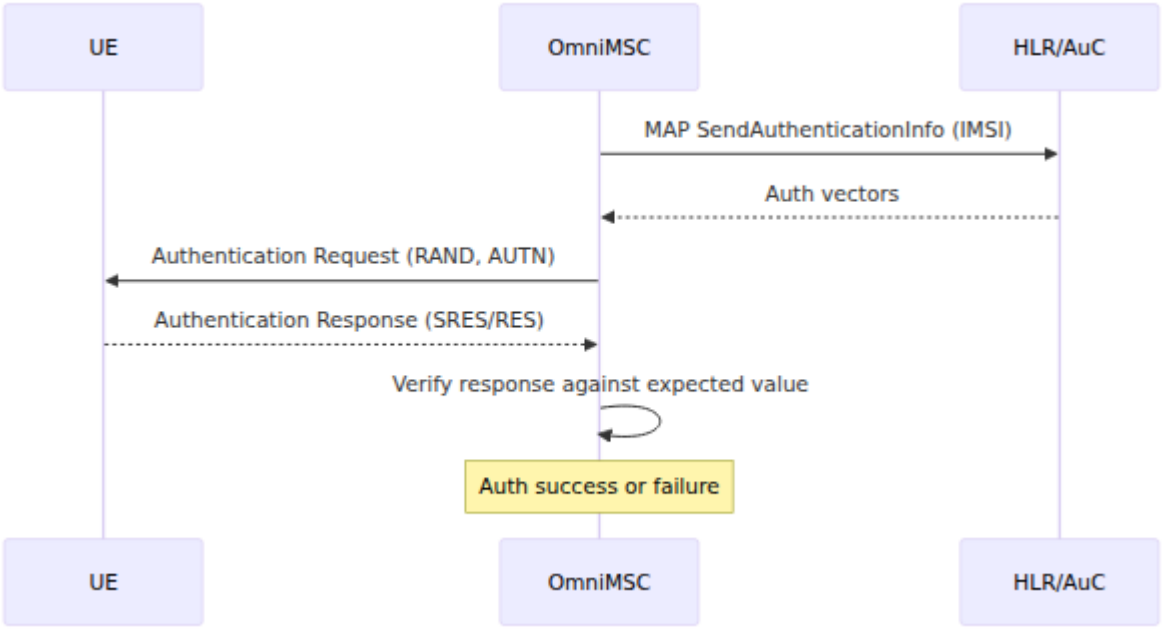
Security

This document describes the authentication, ciphering, and identity management mechanisms implemented by OmniMSC, including GSM and UMTS authentication, air-interface encryption, TMSI allocation, and identity request procedures. For configuration of cipher algorithms and security parameters, see Configuration Reference. For the MAP operations that support authentication, see MAP Operations. For TMSI allocation in MSC pool deployments with NRI encoding, see MSC Pool and NRI.

Authentication Overview

OmniMSC supports both GSM (2G) and UMTS (3G) authentication as defined in 3GPP TS 33.102 and TS 24.008 Section 4.3. Authentication is performed during location updates and, optionally, during call setup and SMS transactions.

The MSC does not store long-term subscriber credentials (Ki). Authentication vectors are obtained from the HLR/AuC via MAP SendAuthenticationInfo (see MAP Operations). The MSC sends the IMSI to the HLR, which returns a set of authentication vectors. The MSC then challenges the UE and verifies its response. The subscriber's current authentication state and remaining tuple count are visible in the control panel — see Control Panel Guide.



UMTS AKA (Authentication and Key Agreement)

UMTS Authentication and Key Agreement provides mutual authentication -- the network authenticates the subscriber and the subscriber authenticates the network. This is the preferred authentication method for USIM-equipped terminals.

The HLR returns quintuplets, each containing five values:

Field	Size	Description
RAND	128 bits	Random challenge generated by AuC
XRES	32-128 bits	Expected response, used by MSC to verify UE
CK	128 bits	Cipher key for air-interface encryption
IK	128 bits	Integrity key for air-interface integrity protection
AUTN	128 bits	Authentication token, used by UE to verify network

The MSC sends RAND and AUTN to the UE. The USIM verifies AUTN to authenticate the network, then computes RES, CK, and IK. The MSC compares the returned RES against XRES to authenticate the subscriber.

SQN Resynchronisation

AUTN contains a sequence number (SQN) that the USIM checks to prevent replay attacks. If the USIM determines that SQN is out of range (e.g., after a long period of inactivity or database restoration), it returns an Authentication Failure with cause "SQN failure" and includes an AUTS (resynchronisation token) of 112 bits. The MSC forwards AUTS to the HLR in a new MAP SendAuthenticationInfo request, allowing the AuC to resynchronise its SQN counter and return fresh vectors.

GSM AKA

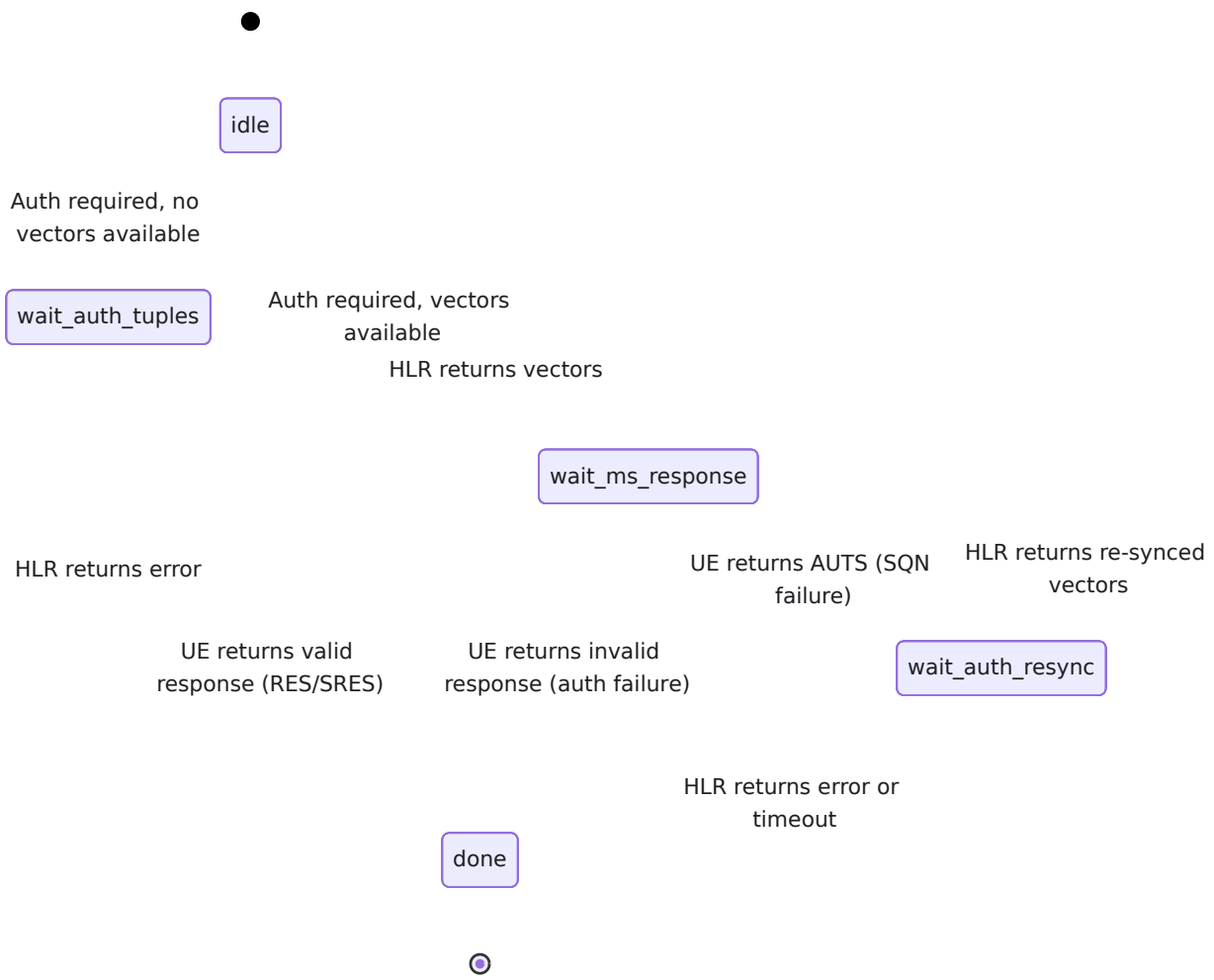
GSM authentication uses triplets for 2G-only subscribers (SIM without USIM application). Each triplet contains:

Field	Size	Description
RAND	128 bits	Random challenge
SRES	32 bits	Signed response, computed by SIM using A3(Ki, RAND)
Kc	64 bits	Cipher key, computed by SIM using A8(Ki, RAND)

GSM authentication is one-way: the network authenticates the subscriber, but the subscriber does not authenticate the network. The MSC sends RAND, the SIM computes SRES and Kc, and the MSC verifies SRES against the expected value from the triplet.

Auth FSM States

The authentication procedure is managed by a finite state machine within the VLR. The FSM tracks the progress of each authentication attempt.



In the `wait_auth_tuples` state, the MSC has sent MAP SendAuthenticationInfo and is awaiting vectors from the HLR. In `wait_ms_response`, the MSC has sent Authentication Request to the UE and is awaiting the response. The `wait_auth_resync` state handles the AUTS resynchronisation procedure when the UE reports a sequence number mismatch.

Ciphering

After successful authentication, the MSC initiates air-interface ciphering to protect signaling and user traffic on the radio path.

GERAN (2G/3G via BSC)

For the A-interface, the MSC sends a BSSMAP Cipher Mode Command to the BSC, carrying the cipher key (Kc) and the selected A5 algorithm. The BSC activates ciphering on the radio channel and returns Cipher Mode Complete.

Algorithm	Security	Description
A5/1	Moderate	Original GSM cipher, widely deployed
A5/3	Strong	KASUMI-based cipher, recommended for all deployments

UTRAN (3G via RNC)

For the Iu-CS interface, the MSC sends a RANAP Security Mode Command to the RNC, carrying CK, IK, and the selected UEA (encryption) and UIA (integrity) algorithms. The RNC activates ciphering and integrity protection and returns Security Mode Complete.

Allowed A5 Configuration

The operator configures the set of permitted A5 algorithms. The MSC selects the strongest algorithm supported by both the network configuration and the mobile station's reported capabilities.

The permitted algorithms are specified as a list: `allowed_a5: [:a5_1, :a5_3]`. The MSC intersects this list with the MS classmark capabilities and selects the highest-strength match. If no common algorithm exists and A5/0 is not permitted, the MSC rejects the connection.

TMSI Allocation

The MSC allocates a Temporary Mobile Subscriber Identity (TMSI) to each subscriber after a successful Location Update. The TMSI replaces the IMSI for

subsequent paging and identification, reducing exposure of the permanent identity on the air interface.

Allocation and Confirmation

After completing authentication, ciphering, and the HLR UpdateLocation exchange, the MSC generates a new TMSI and sends it to the mobile station in the Location Update Accept message. The MS stores the TMSI and responds with TMSI Reallocation Complete.

If the TMSI Reallocation Complete is not received within the reallocation timer, the MSC confirms the new TMSI rather than rolling back. This design choice prevents a scenario where the MS has successfully stored the new TMSI but the confirmation was lost on the air interface -- rolling back would leave the MSC and MS with mismatched TMSIs, breaking subsequent paging.

TMSI in MSC Pool

When operating in an MSC pool (3GPP TS 23.236), the TMSI carries NRI (Network Resource Identifier) bits that allow the BSC to route returning subscribers to the correct MSC in the pool. The NRI is embedded in a configurable bit range within the TMSI. When a BSC receives a service request or paging response containing a TMSI, it extracts the NRI and routes the signaling to the MSC that owns that NRI range.

For MSC pool configuration and NRI assignment, see MSC Pool & NRI.

Identity Request

When the MSC cannot resolve a subscriber's identity -- for example, when a TMSI is presented that is not found in the local VLR (after MSC restart, pool re-routing, or VLR overflow) -- the MSC sends an Identity Request to the mobile station requesting the IMSI.

The MS responds with an Identity Response containing its IMSI. The MSC then proceeds with authentication using the IMSI. This procedure is defined in 3GPP

TS 24.008 Section 4.3.3.

Identity Request is also used to obtain the IMEI (International Mobile Equipment Identity) when equipment checking is required.

3GPP Specification References

Specification	Title	Relevance
TS 33.102	3G Security; Security Architecture	UMTS AKA, quintuplets, SQN resynchronisation, key hierarchy
TS 24.008	Mobile Radio Interface Layer 3	Authentication Request/Response (Sec 4.3), Identity Request (Sec 4.3.3), TMSI Reallocation (Sec 4.3.1)
TS 43.020	Security Related Network Functions	GSM A3/A8, A5 cipher algorithms
TS 48.008	MSC-BSS Interface (BSSMAP)	Cipher Mode Command/Complete
TS 25.413	UTRAN Iu Interface (RANAP)	Security Mode Command/Complete
TS 23.236	Intra-Domain Connection of RAN Nodes to Multiple CN Nodes	NRI allocation, TMSI structure for MSC pool
TS 29.002	MAP Specification	MAP SendAuthenticationInfo

SGs Interface and CSFB

This document describes the SGs interface and Circuit-Switched Fallback (CSFB) implementation in OmniMSC by Omnitouch per 3GPP TS 29.118. The SGs interface connects the MSC/VLR to the MME, enabling combined EPS/IMSI attach, CS paging via the LTE network, and SMS delivery without CS fallback.

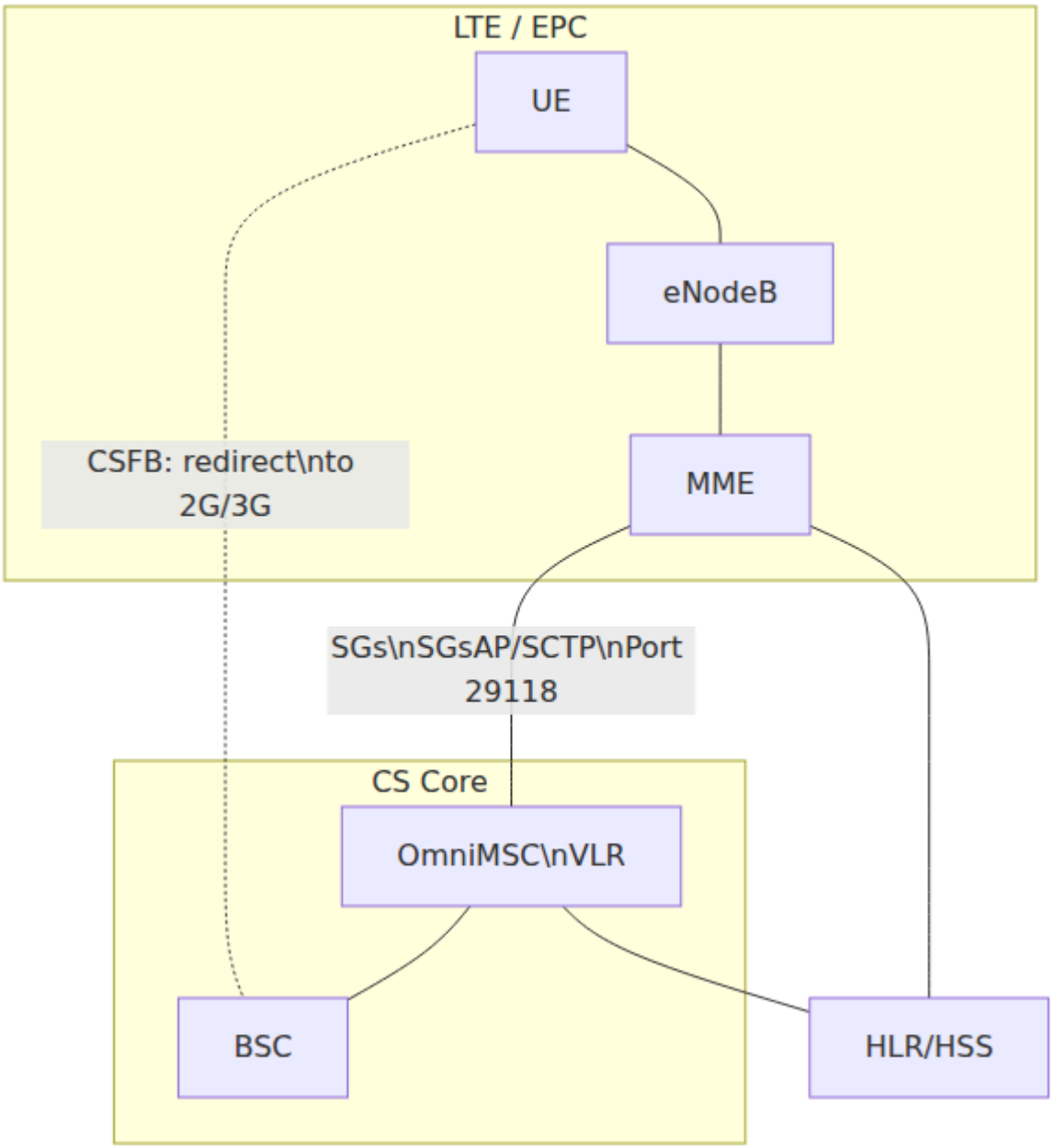
For the CSFB MT call sequence diagram, see Call Flow Diagrams. For configuration parameters, see Configuration Reference. For authentication during combined attach, see Authentication & Security. For SMS over SGs, see SMS. For MSC Pool considerations with CSFB, see MSC Pool.

Overview

In LTE networks, the MME handles mobility management for packet-switched services. However, LTE does not natively support circuit-switched voice prior to VoLTE deployment. CSFB allows LTE-attached subscribers to receive and make CS voice calls and SMS by falling back to the 2G/3G CS domain.

The SGs interface is the signalling link between the MSC/VLR and the MME, carrying SGsAP messages over SCTP (default port 29118). Through SGs, the MSC can:

- Perform combined EPS/IMSI attach, registering a subscriber in both the EPC and CS domains simultaneously.
- Page LTE-attached subscribers for incoming CS calls, triggering a fallback to GERAN or UTRAN.
- Deliver SMS to LTE-attached subscribers without CSFB, tunnelling the NAS PDU through the SGs interface.



SGsAP Message Types

The SGs interface carries the following SGsAP message types per 3GPP TS 29.118.

Location Update

Message	Direction	Purpose
SGsAP-LOCATION-UPDATE-REQUEST	MME to MSC	Combined EPS/IMSI attach or periodic location area update
SGsAP-LOCATION-UPDATE-ACCEPT	MSC to MME	Location update accepted, includes new TMSI
SGsAP-LOCATION-UPDATE-REJECT	MSC to MME	Location update rejected with cause code

Paging and Service

Message	Direction	Purpose
SGsAP-PAGING-REQUEST	MSC to MME	Page subscriber for MT call or MT SMS
SGsAP-SERVICE-REQUEST	MME to MSC	Subscriber responding to CS paging (CSFB in progress)
SGsAP-SERVICE-ABORT-REQUEST	MSC to MME	Abort a CS fallback service request

SMS Tunnelling

Message	Direction	Purpose
SGsAP-DOWNLINK-UNITDATA	MSC to MME	MT SMS delivery: NAS PDU carried to UE via SGs
SGsAP-UPLINK-UNITDATA	MME to MSC	MO SMS submission: NAS PDU carried from UE via SGs

Detach

Message	Direction	Purpose
SGsAP-EPS-DETACH-INDICATION	MME to MSC	Subscriber detached from EPS
SGsAP-EPS-DETACH-ACK	MSC to MME	Acknowledge EPS detach
SGsAP-IMSI-DETACH-INDICATION	MME to MSC	Subscriber IMSI detach
SGsAP-IMSI-DETACH-ACK	MSC to MME	Acknowledge IMSI detach

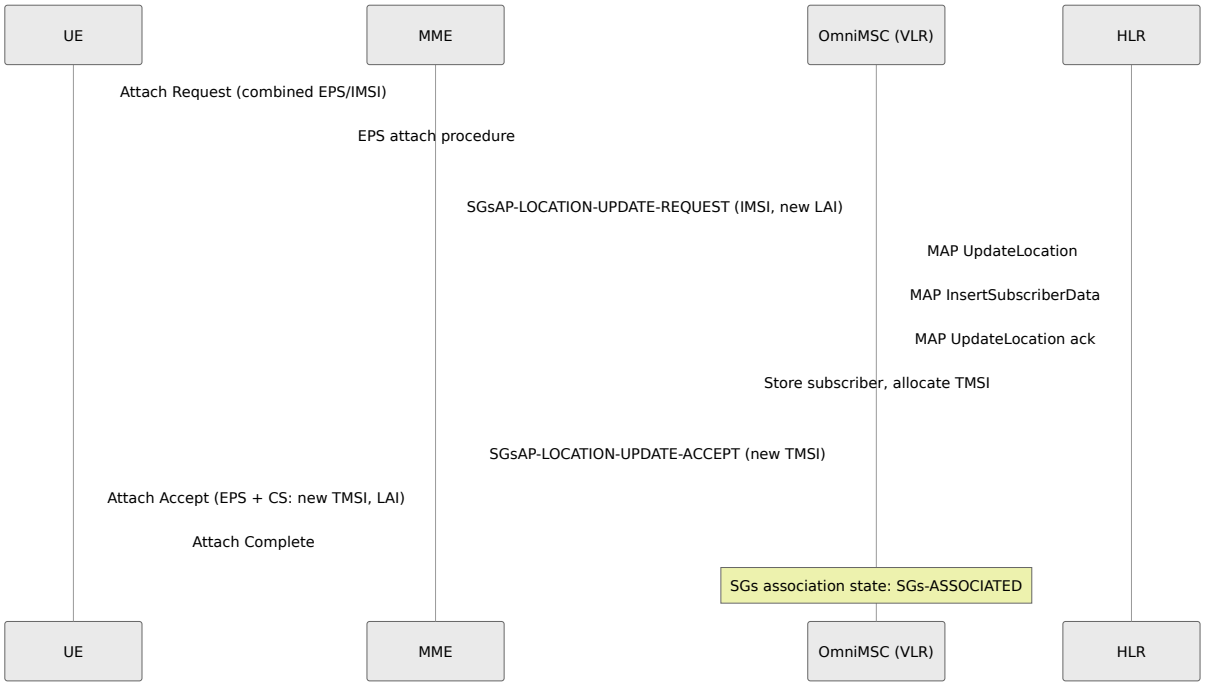
Reset and Status

Message	Direction	Purpose
SGsAP-RESET-INDICATION	Either direction	Peer has restarted; receiver must re-register affected subscribers
SGsAP-RESET-ACK	Either direction	Acknowledge reset indication
SGsAP-STATUS	Either direction	Error indication with cause and erroneous message
SGsAP-MM-INFORMATION-REQUEST	MSC to MME	Network name and time zone information
SGsAP-ALERT-REQUEST	MSC to MME	Alert request after paging failure
SGsAP-ALERT-ACK	MME to MSC	Acknowledge alert
SGsAP-UE-ACTIVITY-INDICATION	MME to MSC	UE has become active
SGsAP-RELEASE-REQUEST	MSC to MME	Release SGs connection for this subscriber

---

Combined EPS/IMSI Attach

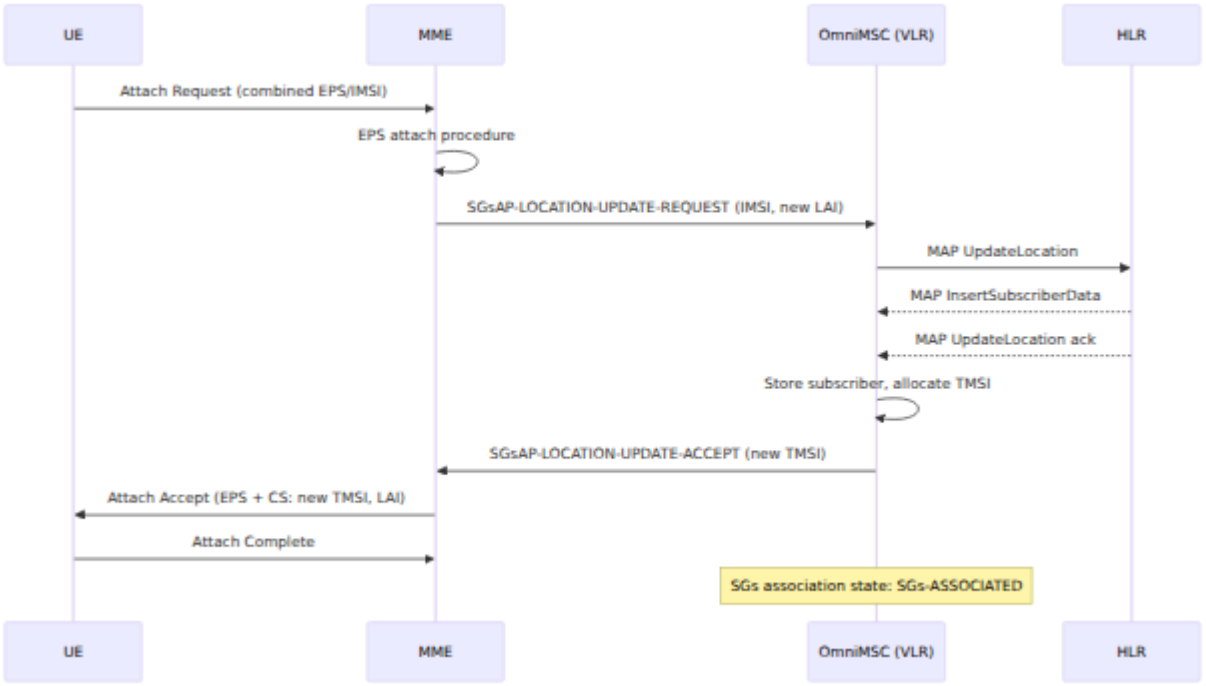
When a UE performs a combined EPS/IMSI attach in LTE, the MME sends an SGsAP-LOCATION-UPDATE-REQUEST to the MSC. The MSC performs a VLR location update, which may include HLR interrogation, and responds with acceptance or rejection. Upon acceptance, the subscriber is registered in both the EPC (via the MME) and the CS domain (via the MSC/VLR) simultaneously.



After a successful combined attach, the subscriber's SGs association state transitions to SGs-ASSOCIATED. The MSC can now page the subscriber via SGs and deliver SMS without CSFB.

MT Call Paging via CSFB

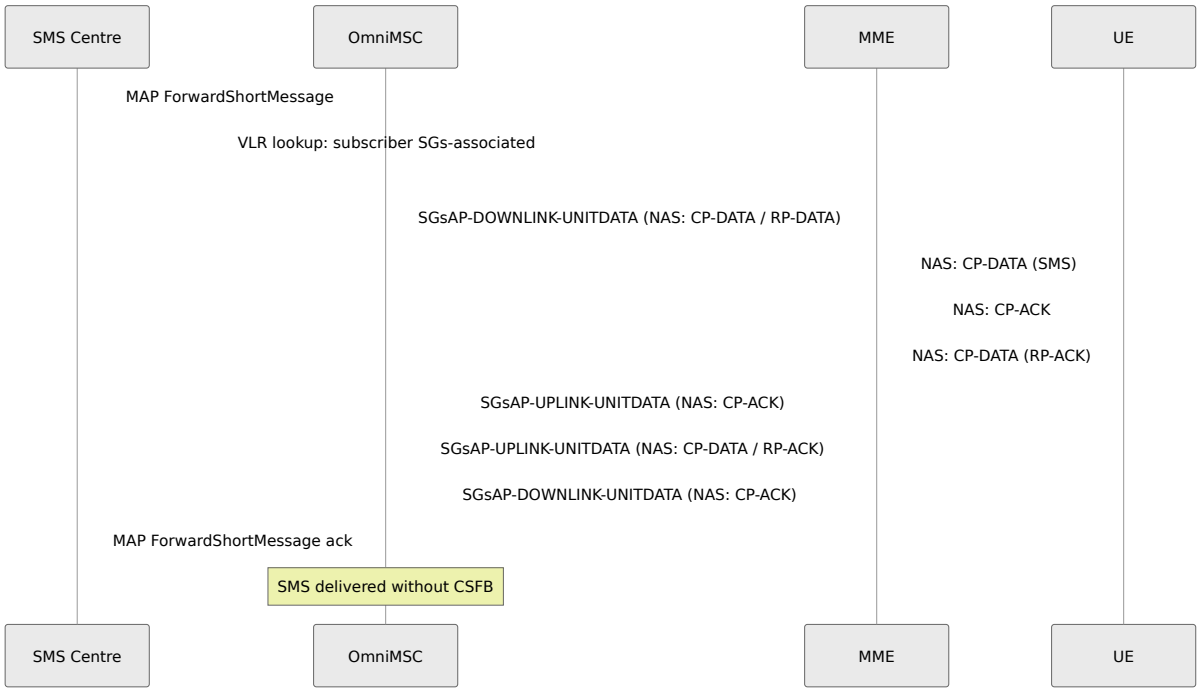
When an MT call arrives for a subscriber that is attached via LTE (SGs-associated), the MSC pages the subscriber through the MME rather than through BSCs. The MME instructs the UE to fall back to 2G or 3G, where the call proceeds over the A-interface or Iu-CS interface.



Once the UE has fallen back to the CS domain and sent a Paging Response via the BSC, the call proceeds as a normal MT call. The MSC-A state machine handles the E-UTRAN/SGs RAN type by omitting the Clear Complete step that would normally be expected from a BSC, since the SGs association does not use BSSMAP connection management.

MT SMS via SGs

SMS can be delivered to LTE-attached subscribers without requiring CSFB. The MSC tunnels the SMS NAS PDU through the SGs interface to the MME, which delivers it to the UE over the LTE air interface. This avoids the latency and radio resource cost of a CS fallback for a short message.

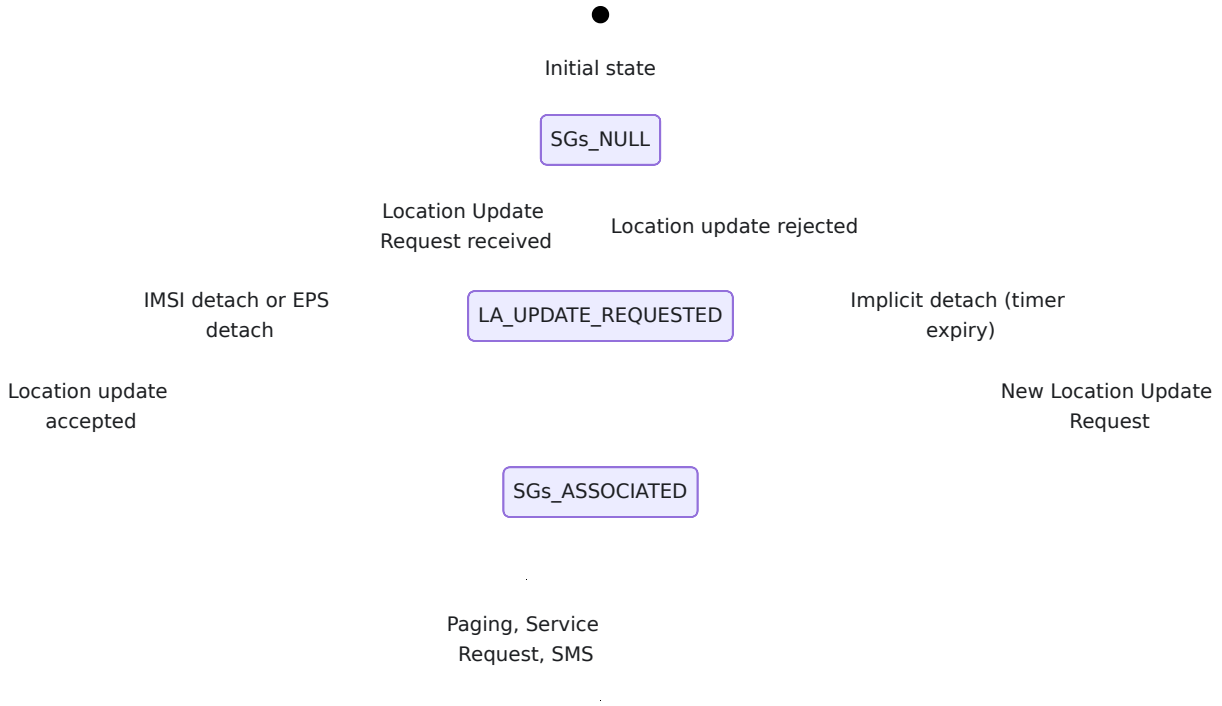


MO SMS follows the reverse path: the UE sends the SMS NAS PDU via the MME as SGsAP-UPLINK-UNITDATA, and the MSC forwards it to the SMS Centre.

SGs Association States

Each subscriber's SGs association is tracked as a state machine per 3GPP TS 29.118 Section 4.

State	Description
SGs-NULL	No SGs association exists. CS paging via SGs is not possible. This is the initial state.
LA-UPDATE-REQUESTED	A location update is in progress. The MSC has received a request from the MME but has not yet completed the HLR update.
SGs-ASSOCIATED	The subscriber has a valid SGs association. The MSC can page via SGs and deliver SMS without CSFB.



MME Tracking

The SGs handler maintains a registry of known MMEs. Each MME is identified by its FQDN (the MME Name information element in SGsAP messages). For each MME, the handler tracks:

- The SCTP connection reference used for sending outbound messages.
- The set of IMSIs (subscribers) currently associated with that MME.

This per-MME subscriber registry enables targeted reset handling: when an MME fails, only the subscribers associated with that specific MME are affected.

MME Reset Handling

Reset procedures ensure state consistency when either the MSC or an MME restarts.

Scenario	Initiator	Action by Receiver
MSC restart	MSC sends SGsAP-RESET-INDICATION to all known MMEs	Each MME re-sends SGsAP-LOCATION-UPDATE-REQUEST for its associated subscribers, allowing the MSC to rebuild VLR state
MME restart	MME sends SGsAP-RESET-INDICATION to the MSC	MSC marks all subscribers associated with that MME as SGs-detached (SGs-NULL state) and clears the subscriber list for that MME
SGs link failure	Detected by either side	Both sides treat the failure as an implicit reset

When the MSC receives a Reset Indication from an MME, it iterates over all subscribers registered against that MME, transitions each to the SGs-NULL state, and clears the MME's subscriber set. On the next contact from any affected subscriber (location update or paging response), the MSC performs a full re-registration.

SGsAP Codec

OmniMSC includes a codec module that handles encoding and decoding of SGsAP messages per the formats defined in 3GPP TS 29.118. The codec processes the binary SGsAP message type octet and information elements (IEs), supporting all mandatory and optional IEs for each message type. Encoded messages are transmitted over SCTP; decoded messages are dispatched to the SGs handler for processing.

Configuration

The SGs interface is configured under the sgs key in the MSC configuration.

Parameter	Default	Description
listen_port	29118	SCTP listen port for SGsAP connections from MMEs. Port 29118 is the 3GPP-defined port for SGs per TS 29.118.
vlr_name	(required)	VLR name in FQDN format, used in SGsAP messages. The MME uses this to identify the VLR. Must match the VLR name configured on the MME side.

RAN Type: E-UTRAN via SGs

The MSC-A state machine supports E-UTRAN via SGs as a distinct RAN type (:eutran_sgs). When a subscriber is SGs-associated, the MSC-A FSM adjusts its behavior for the SGs interface:

- No BSSMAP connection management is used; there is no Clear Command / Clear Complete exchange.
- Paging is performed via SGsAP-PAGING-REQUEST to the MME rather than via BSSMAP Paging to BSCs.
- SMS delivery uses SGsAP Downlink/Uplink Unitdata rather than DTAP over the A-interface.
- Once the subscriber falls back to GERAN or UTRAN (after CSFB), the connection transitions to the corresponding RAN type for the remainder of the call.

3GPP Specification References

Specification	Title	Relevance
TS 29.118	MME-VLR SGs Interface Specification	SGsAP protocol, message formats, procedures
TS 23.272	Circuit Switched Fallback in EPS	CSFB architecture, call flows, SMS over SGs
TS 23.012	Location Management Procedures	VLR location update procedures used in SGs
TS 24.008	Mobile Radio Interface Layer 3	NAS messages tunnelled via SGs Unitdata

SIP-I Trunking

This document describes the SIP-I (SIP with encapsulated ISUP) trunking interface implemented by OmniMSC. SIP-I enables transparent transport of ISUP messages within SIP signaling, preserving full ISUP information across IP-based trunk segments.

For pure SIP trunking, see SIP Trunking. For routing configuration, see Routing Configuration. For configuration parameters, see Configuration Reference. For general operations, see Operations Guide.

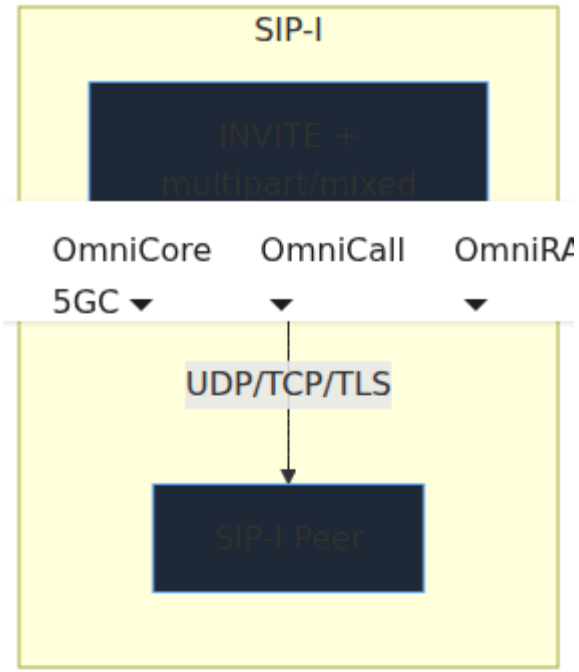
What Is SIP-I?

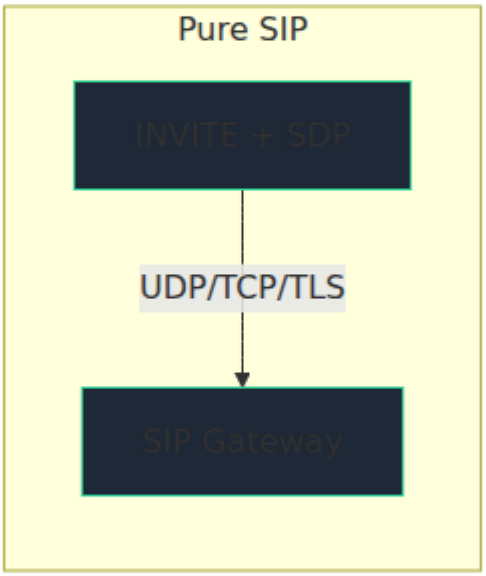
SIP-I (Session Initiation Protocol with encapsulated ISUP) is defined by ITU-T Q.1912.5 and uses the SIP protocol as a transport mechanism for ISUP messages. Unlike pure SIP trunking, which maps ISUP parameters to SIP headers (potentially losing information), SIP-I includes the complete ISUP message as a MIME body alongside the SDP, ensuring lossless interworking.

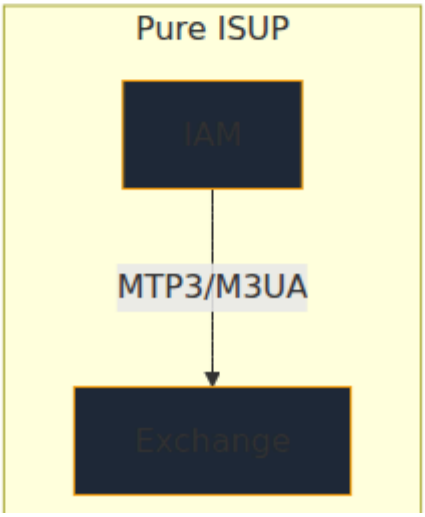
SIP-I is the standard trunking protocol used between MSC servers and media gateways in 3GPP IMS-based core networks and is widely deployed for PSTN interconnection.

The ISUP body is carried per RFC 3204 (MIME media type for ISUP) and RFC 3261 (SIP).

Protocol Comparison


SIP-I
INVITE +
multipart/mixed
OmniCore
5GC
OmniCall
OmniRA
UDP/TCP/TLS
SIP-I Peer


Pure SIP
INVITE + SDP
UDP/TCP/TLS
SIP Gateway


Pure ISUP
IAM
MTP3/M3UA
Exchange

Aspect	Pure ISUP	Pure SIP	SIP-I
Transport	MTP3/M3UA/SCTP	UDP/TCP/TLS	UDP/TCP/TLS
Signaling info	Full ISUP	Mapped to SIP headers	Full ISUP preserved
Media description	Bearer capability in IAM	SDP	SDP + ISUP bearer cap
Information loss	None	Possible (parameter mapping)	None
Codec negotiation	TMR in IAM	SDP Offer/Answer	SDP Offer/Answer
Use case	Legacy PSTN	VoIP interconnect	MSC-MSC, PSTN gateway

Multipart Body Format

SIP-I messages use a `multipart/mixed` MIME body containing two parts: the SDP offer/answer and the ISUP message encoded per RFC 3204.

```
Content-Type: multipart/mixed;boundary=boundary42

--boundary42
Content-Type: application/sdp

v=0
o=OmniMSC 12345 12345 IN IP4 203.0.113.10
s=OmniMSC
c=IN IP4 203.0.113.10
t=0 0
m=audio 10042 RTP/AVP 0 8
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000

--boundary42
Content-Type: application/ISUP;version=itu-t92+

<binary ISUP IAM>
--boundary42--
```

The `application/ISUP` content type is defined in RFC 3204. The `version` parameter identifies the ISUP variant (e.g., `itu-t92+` for ITU-T Q.767).

SIP-I Peer Configuration

SIP-I peers are configured under the `:sip_i` key, separately from pure SIP peers.

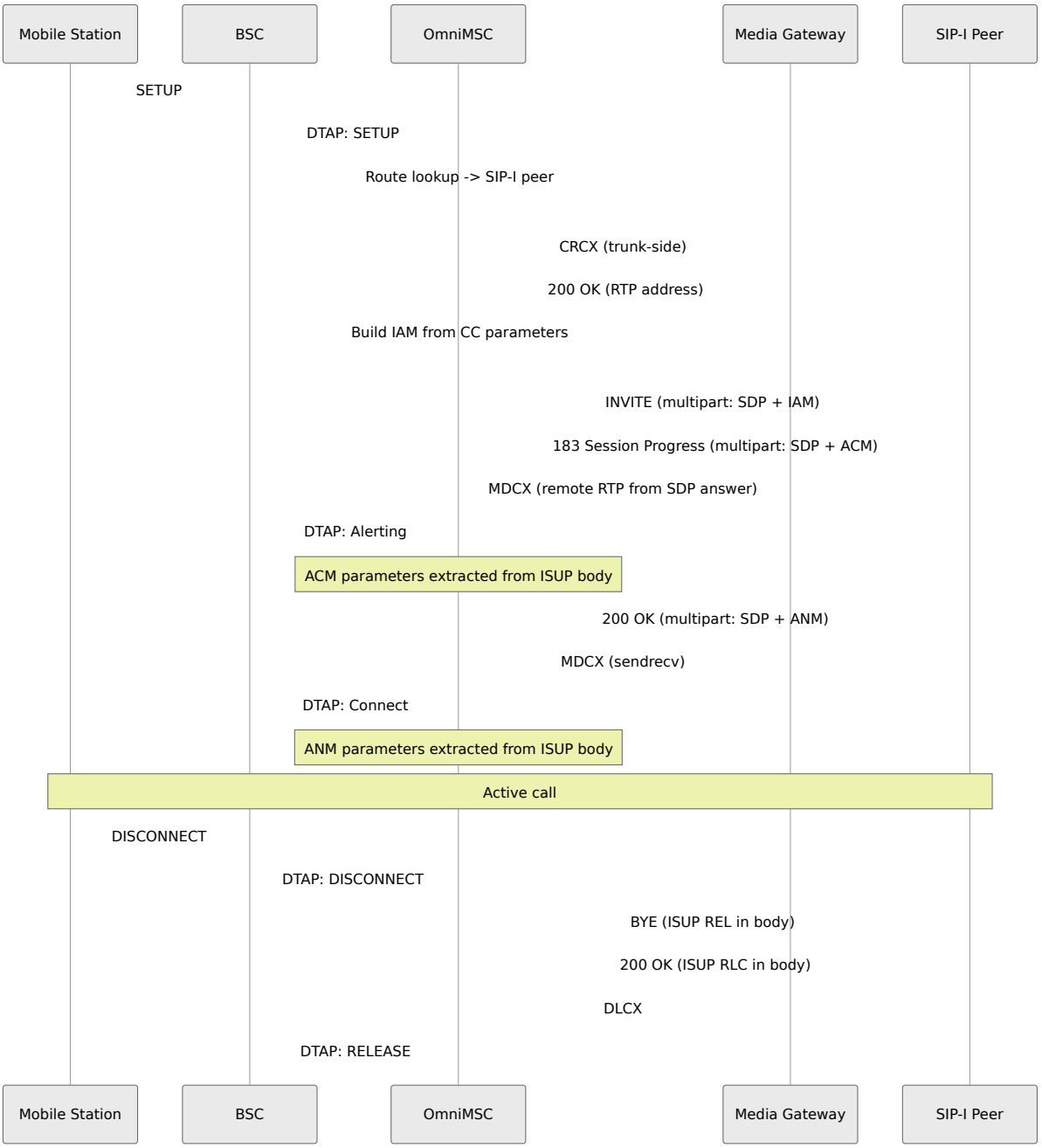
```
config :omnimsc, :sip_i,
  peers: [
    [name: "MSC-02-SIP-I",
     address: "10.2.1.100",
     port: 5060,
     transport: :tcp,
     isup_variant: :itu_t92,
     codecs: [:pcmu, :pcma, :amr],
     max_channels: 500,
     options_interval: 15]
  ]
```

SIP-I Peer Parameters

Parameter	Type	Default	Description
`name`	`string`	-- (required)	Logical peer name. Referenced in route table entries with type `:sip_i`.
`address`	`string`	-- (required)	Peer IP address or hostname.
`port`	`integer`	`5060`	Peer SIP port.
`transport`	`atom`	`:tcp`	Transport protocol: `:udp`, `:tcp`, or `:tls`. TCP is recommended for SIP-I due to larger message sizes.
`isup_variant`	`atom`	`:itu_t92`	ISUP encoding variant: `:itu_t92` (ITU-T Q.767), `:ansi` (ANSI T1.113), `:etsi` (ETSI EN 300 356).
`codecs`	`list(atom)`	`[:pcmu, :pcma]`	Supported audio codecs for the SDP portion.
`max_channels`	`integer`	`500`	Maximum concurrent calls to this peer.
`options_interval`	`integer` or `nil`	`nil`	Interval in seconds for SIP OPTIONS keepalive probes.

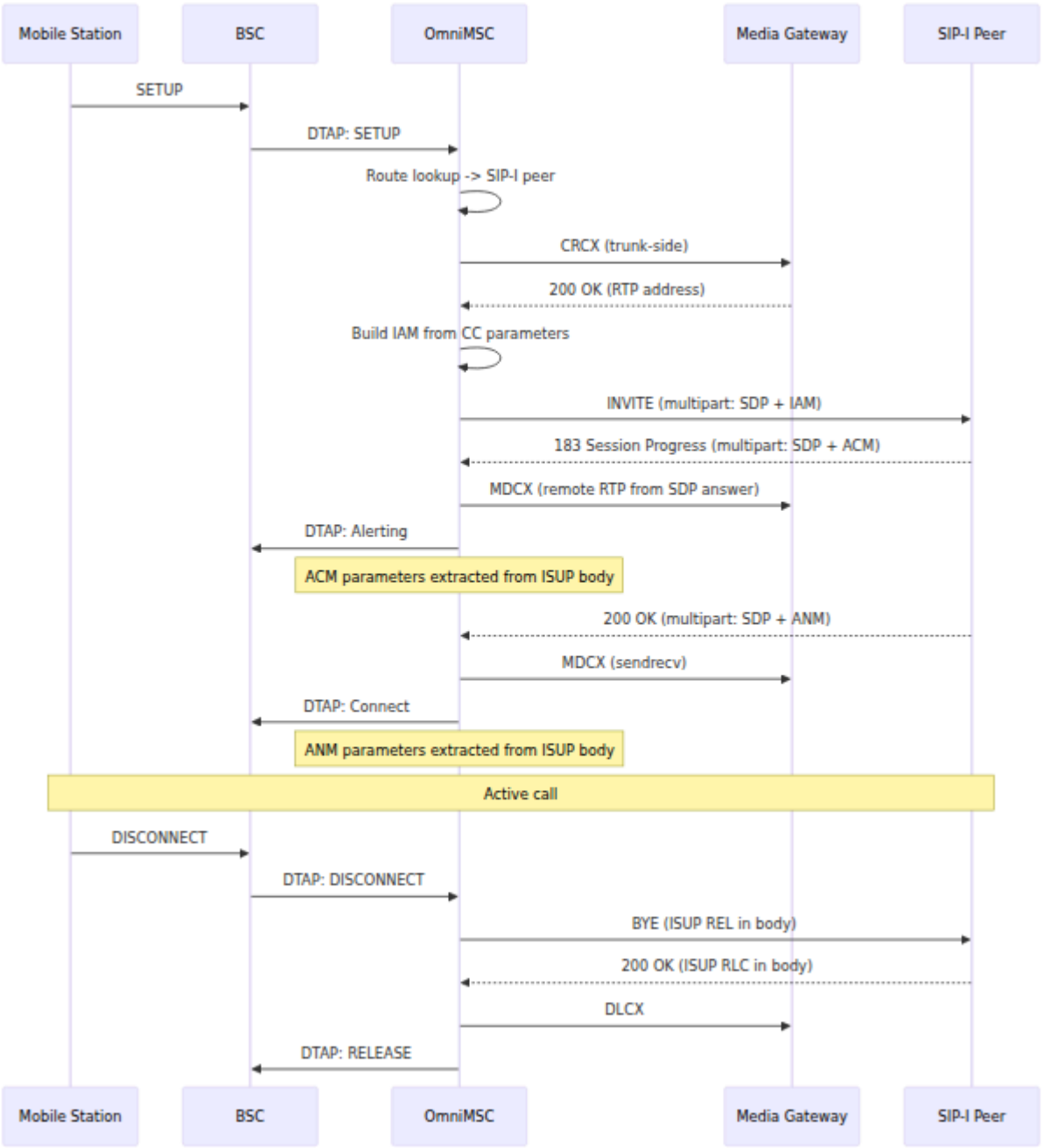
Outgoing Call (SIP-I)

When OmniMSC routes a call to a SIP-I peer, it constructs the SIP INVITE with a multipart body containing both SDP and the ISUP IAM.



Incoming Call (SIP-I)

When an INVITE arrives from a SIP-I peer with a multipart body, OmniMSC extracts the ISUP message and uses it to populate the CC FSM parameters.



ISUP-SIP Header Mapping

When interworking between the ISUP body and SIP headers, OmniMSC applies the following mapping. The ISUP body is authoritative; SIP headers are populated for the benefit of SIP-only intermediaries.

ISUP Parameter (IAM)	SIP Header	Notes
Called Party Number	Request-URI, To	E.164 format in `tel:` URI
Calling Party Number	From, P-Asserted-Identity	Presentation indicator controls `Privacy` header
Nature of Connection Indicators	Via	Satellite hop indicator
Forward Call Indicators	--	Encoded in ISUP body only
Calling Party Category	P-Asserted-Identity	Operator/priority category
Transmission Medium Requirement	SDP `m=` line	Speech, 3.1kHz audio, 64k unrestricted
User Service Information	SDP codec lines	Codec and rate mapping
Optional Forward Call Indicators	Supported	ISDN access indicator

ISUP Parameter (ACM/ANM)	SIP Response	Notes
Backward Call Indicators	183/200	Charge indicator, interworking flag
Cause Indicators (REL)	Reason header	Q.850 cause per RFC 3326
Optional Backward Call Indicators	--	Encoded in ISUP body only

---

Cause Code Mapping Reference

SIP-I preserves the full ISUP cause code in the ISUP body. Additionally, the SIP `Reason` header carries the Q.850 cause for intermediary nodes.

For call clearing, the ISUP REL message in the BYE body takes precedence over the SIP Reason header if both are present.

3GPP and ITU-T References

Reference	Title	Relevance
ITU-T Q.1912.5	Interworking between SIP and Bearer Independent Call Control (BICC) or ISUP	SIP-I protocol definition
RFC 3204	MIME Media Type for ISUP and QSIG Objects	`application/ISUP` content type
RFC 3261	SIP: Session Initiation Protocol	SIP transport
RFC 3264	Offer/Answer Model with SDP	SDP negotiation within SIP-I
RFC 3326	Reason Header Field	Cause code in SIP responses
ITU-T Q.767	Application of the ISUP	ISUP message encoding
ITU-T Q.850	Usage of Cause in ISDN	Cause code definitions
3GPP TS 29.163	Interworking between SIP-I based circuit-switched core and other networks	3GPP SIP-I profile

SIP Trunking

This document covers SIP peer configuration, OPTIONS keepalive monitoring, SDP codec negotiation, in-dialog re-INVITE handling, session timers, DTMF relay, and SIP trunk call states in OmniMSC.

For SIP-related routing, see Routing Configuration. For SIP with encapsulated ISUP, see SIP-I Trunking. For SIP trunk troubleshooting, see Troubleshooting Guide. For call flow sequences showing SIP signaling in context, see Call Flow Diagrams. For media gateway codec negotiation, see Media Control. For SIP peer configuration parameters, see Configuration Reference.

SIP Peer Configuration

Each SIP peer represents a remote endpoint such as a VoIP gateway, SBC, IMS node, or SIP trunking provider. Peers are defined in the `:sip` configuration block and referenced by name in the route table.

Parameter	Type	Default	Description
`name`	`string`	-- (required)	Logical peer name. Referenced in route table entries.
`address`	`string`	-- (required)	Peer IP address or hostname.
`port`	`integer`	`5060`	Peer SIP port.
`transport`	`atom`	`:udp`	Transport protocol: `:udp`, `:tcp`, or `:tls`.
`codecs`	`list(atom)`	`[:pcmu, :pcma]`	Supported audio codecs for SDP negotiation.
`max_channels`	`integer`	`100`	Maximum concurrent calls to this peer.
`options_interval`	`integer` or `nil`	`nil`	Interval in seconds for SIP OPTIONS keepalive probes.

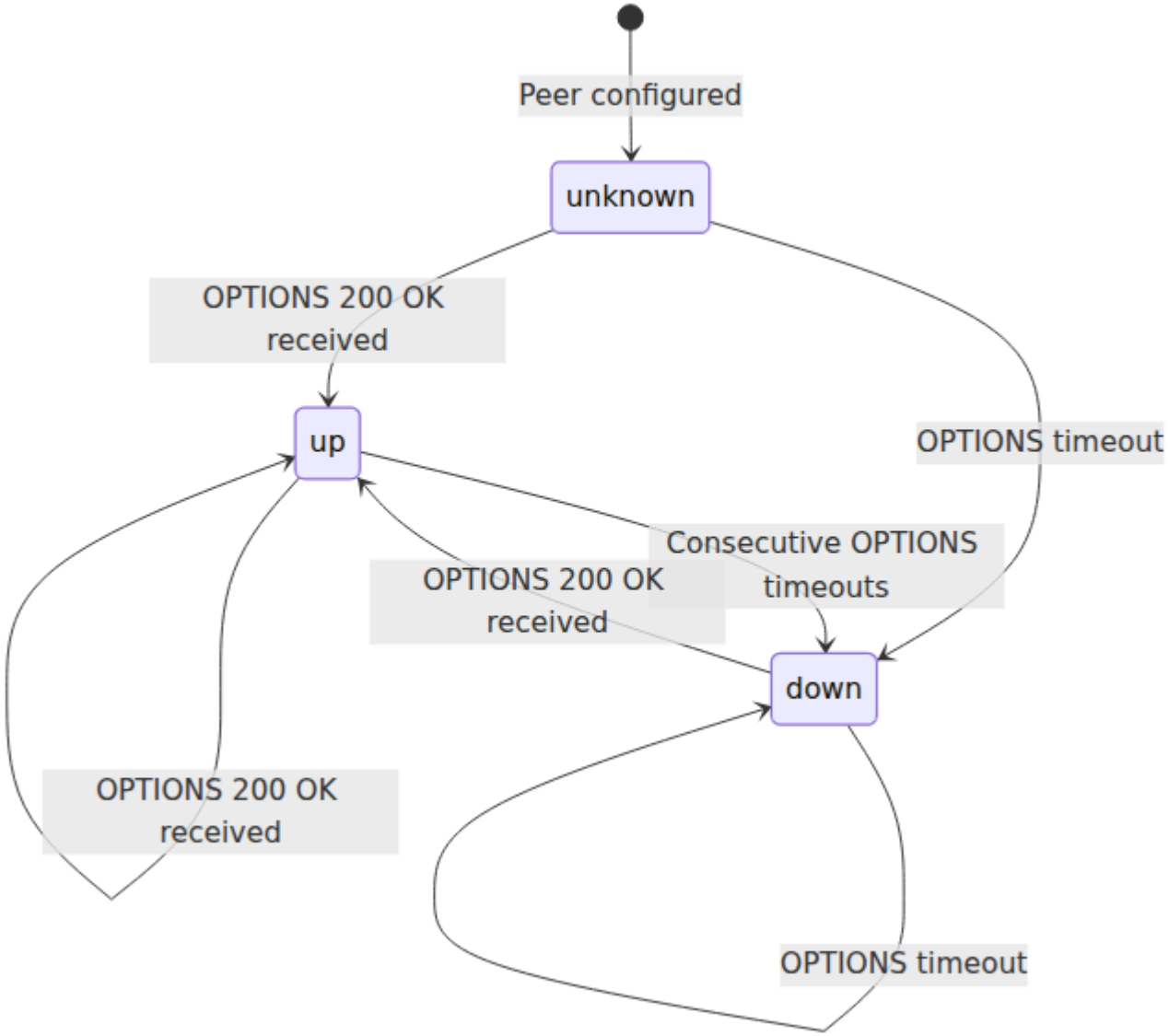
OmniMSC identifies itself with the User-Agent header `OmniMSC/0.1` in all outgoing SIP requests and responses.

SIP OPTIONS Keepalive

When `options_interval` is configured for a peer, the SIP Peer Manager sends periodic SIP OPTIONS requests to monitor peer health. The peer status determines whether the peer is eligible for call routing.

Peer Health States

Each peer tracks a status of `:up`, `:down`, or `:unknown`. On startup, all peers begin in the `:unknown` state.

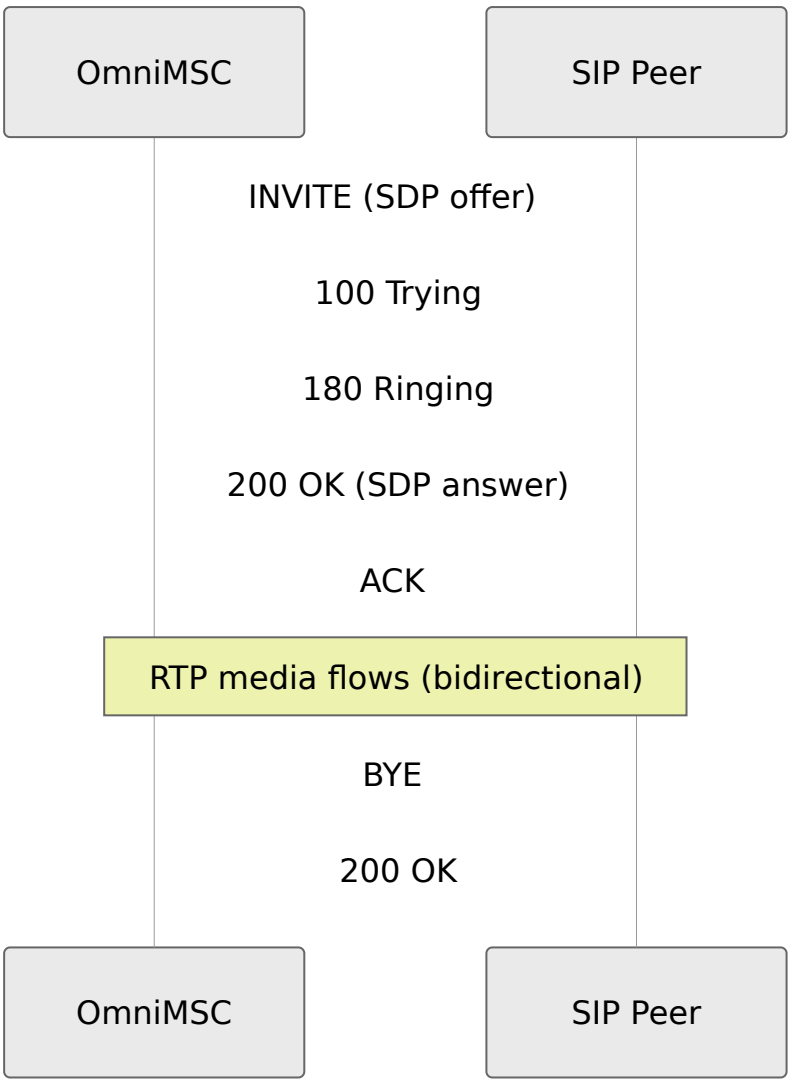


Event	Transition	Effect
OPTIONS 200 OK received	Any -> up	Peer eligible for routing
Consecutive OPTIONS timeouts	up/unknown -> down	Peer excluded from routing, alarm raised
OPTIONS 200 OK after down	down -> up	Peer re-eligible, alarm cleared
`max_channels` reached	up -> up (soft limit)	New calls rejected for this peer, existing calls unaffected

For SIP peer monitoring in the control panel, see Control Panel Guide.

MO Call SIP Flow

When OmniMSC routes a mobile-originating call to a SIP peer, the following SIP signaling exchange takes place between OmniMSC and the remote peer.



The INVITE carries an SDP offer with codecs based on the peer configuration and BSC capabilities. The 200 OK contains the SDP answer with the selected codec and remote RTP address. After ACK, the RTP media path is established through the media gateway.

In-Dialog Re-INVITE Handling

A SIP peer may send a re-INVITE within an established dialog for several purposes: call hold, codec change, or session refresh. OmniMSC processes re-INVITEs and responds with 200 OK using the current session SDP.

Re-INVITE Purpose	SDP Indicator	OmniMSC Behavior
Call hold	`a=sendonly`	Acknowledge hold, update MGW mode to `recvonly`
Call resume	`a=sendrecv`	Resume media, update MGW mode to `sendrecv`
Codec change	Modified `m=` line	Renegotiate codec if supported, reject with 488 if not
Session refresh	No SDP change	Respond with 200 OK, reset session timer

When OmniMSC receives a re-INVITE it cannot accept (unsupported codec, missing SDP), it responds with 488 Not Acceptable Here. The existing dialog and media session remain unaffected.

Session Timer (RFC 4028)

OmniMSC supports SIP session timers per RFC 4028 to detect and clean up orphaned SIP sessions. Session timers ensure that both endpoints periodically refresh the session, preventing stale call state after network failures.

Parameter	Value	Description
Session-Expires	1800s (default)	Maximum time between session refreshes
Min-SE	90s	Minimum acceptable Session-Expires value
Refresher	UAC or UAS	Determined during negotiation

Session Timer Negotiation

OmniMSC includes the `Session-Expires` and `Min-SE` headers in outgoing INVITE requests and 200 OK responses. When a peer proposes a Session-Expires value below the configured Min-SE, OmniMSC responds with 422 Session Interval Too Small and includes the Min-SE header indicating the minimum acceptable value.

Session refresh is performed via re-INVITE. If no refresh arrives before the session expires, OmniMSC sends BYE to tear down the call and releases all associated resources.

DTMF Relay

OmniMSC relays DTMF tones using SIP INFO messages per the `application/dtmf-relay` content type. This method is used when the peer does not support RFC 2833 telephone-event RTP payload or when out-of-band DTMF is preferred.

Field	Description	Example
Content-Type	MIME type for DTMF relay	`application/dtmf-relay`
Signal	DTMF digit (0-9, *, #, A-D)	`Signal=5`
Duration	Tone duration in milliseconds	`Duration=160`

When DTMF events are detected from the radio side (via the media gateway), OmniMSC generates a SIP INFO message toward the SIP peer with the corresponding signal and duration. In the reverse direction, incoming SIP INFO DTMF events are forwarded to the media gateway for playout toward the mobile station.

SDP Codec Negotiation

OmniMSC generates SDP offers based on the intersection of the peer's configured codec list and the BSC-reported speech codec capabilities. Codecs are offered in preference order.

Supported Codecs

Codec	RTP Payload Type	Bandwidth	fmtp Parameters
AMR	dynamic (96)	4.75-12.2 kbps	`octet-align=1`
GSM-EFR	dynamic (97)	12.2 kbps	--
GSM-FR	3	13 kbps	--

AMR is offered with `octet-align=1` (RFC 4867) for interoperability with 3GPP access networks. GSM-EFR and GSM-FR are offered when the BSC indicates support for these codecs in the speech version list during assignment.

Codec Selection

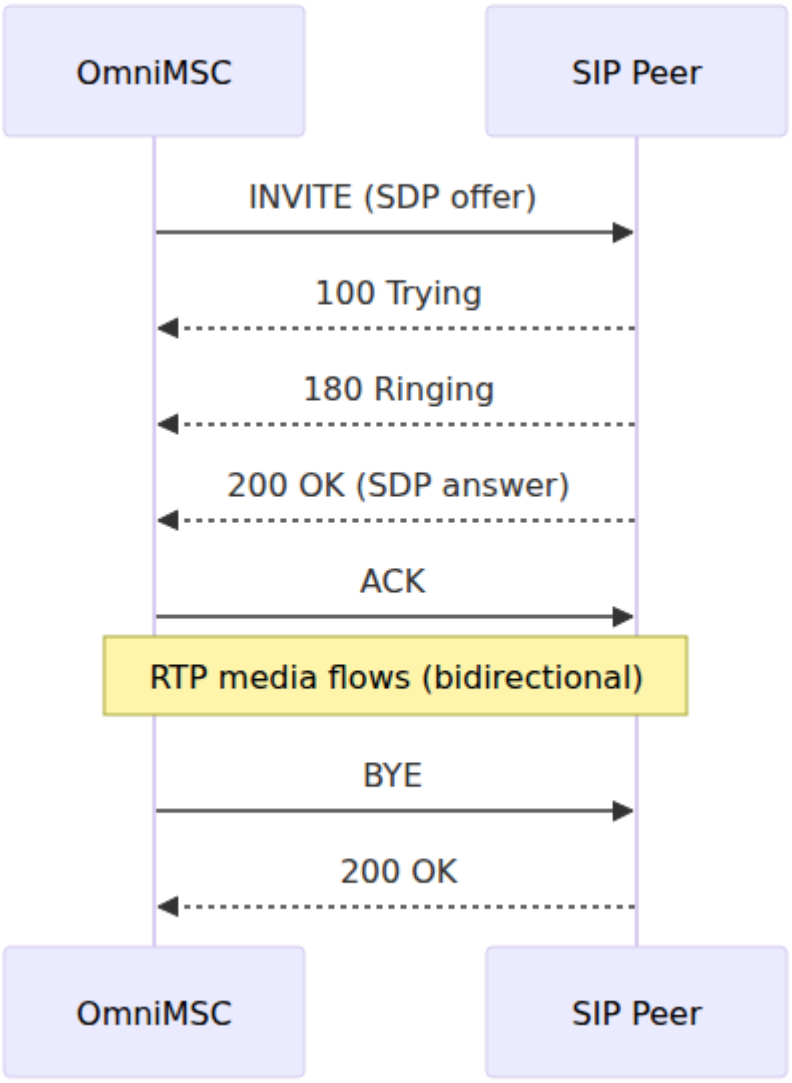
The codec selection follows the SDP Offer/Answer model (RFC 3264):

1. OmniMSC builds the SDP offer from the peer codec list, filtered by BSC speech capabilities.
2. The remote peer responds with an SDP answer containing one or more accepted codecs.
3. OmniMSC selects the first common codec from the original offer order.
4. The media gateway is instructed (via MDCX) with the selected codec and RTP parameters.

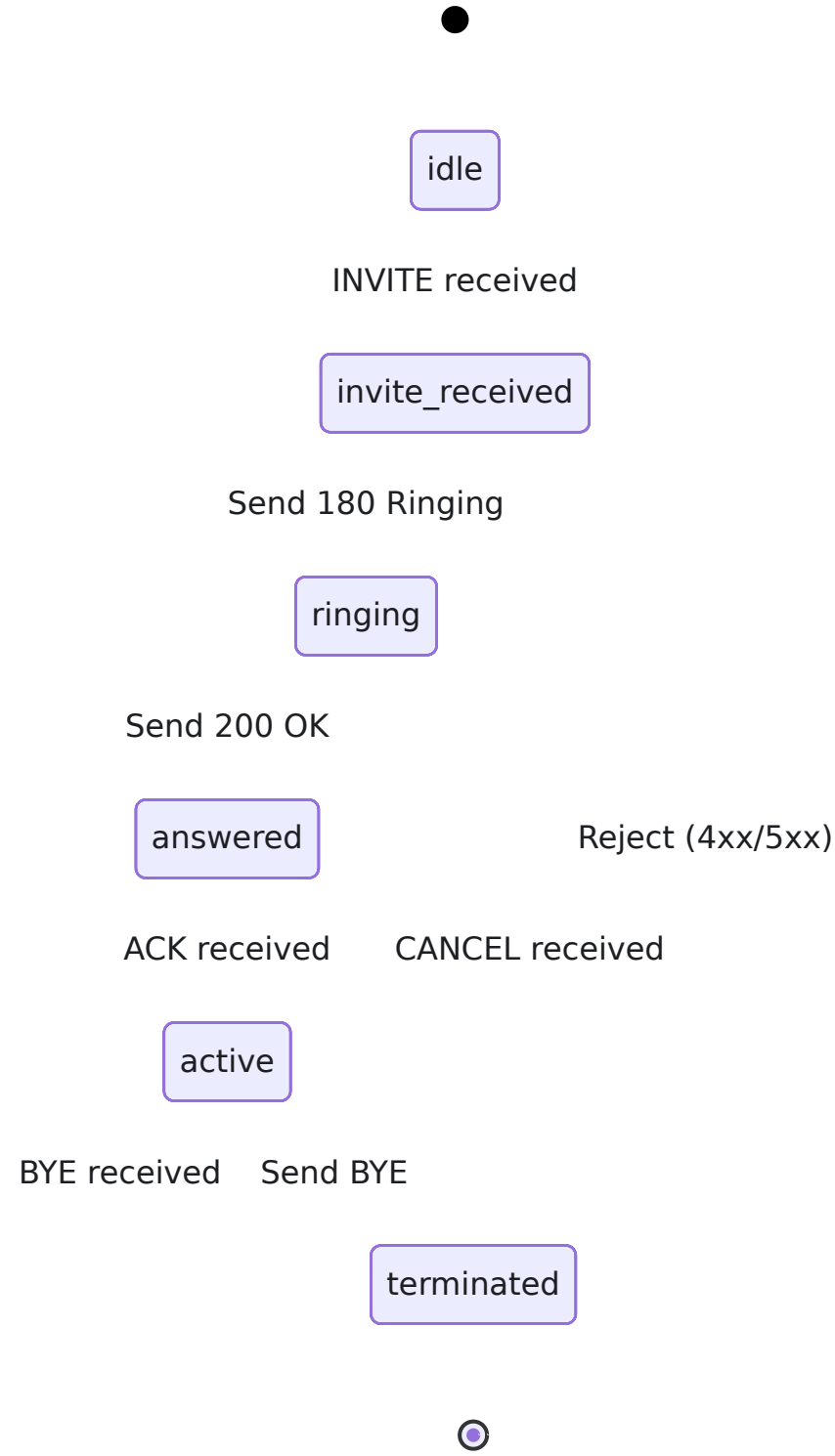
If no common codec exists, OmniMSC responds with or receives 488 Not Acceptable Here.

SIP Trunk Call States

Outgoing Call States



Incoming Call States



References

Reference	Title	Relevance
RFC 3261	SIP: Session Initiation Protocol	Core SIP signaling
RFC 4028	Session Timers in SIP	Session-Expires, Min-SE, refresh mechanism
RFC 2833	RTP Payload for DTMF Digits	Telephone-event RTP payload type
RFC 3264	Offer/Answer Model with SDP	SDP codec negotiation
RFC 4867	RTP Payload Format for AMR and AMR-WB	AMR octet-align parameter
RFC 3326	Reason Header Field	Cause code in BYE/CANCEL

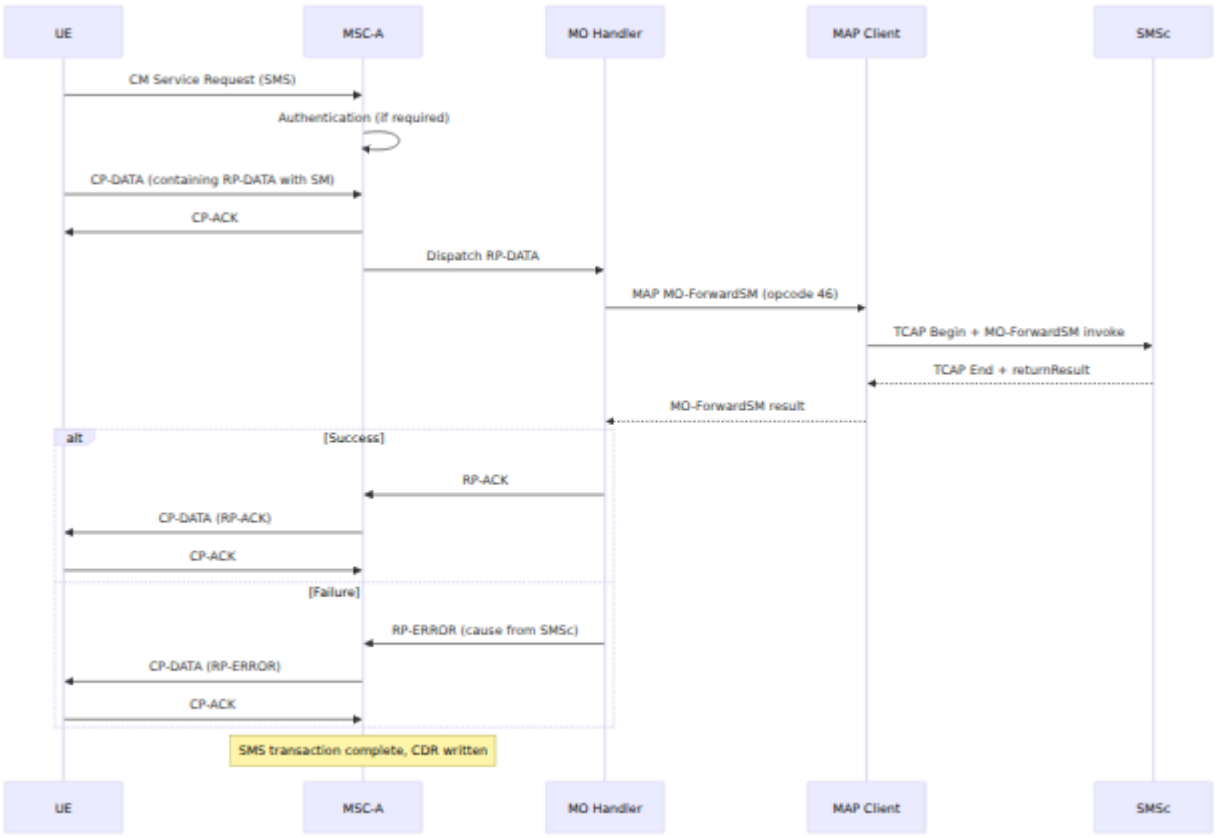
SMS

This document describes the Short Message Service implementation in OmniMSC, covering mobile-originating and mobile-terminating SMS flows, DTAP transaction identifier allocation, A-interface SAPI handling, MAP response routing, and the SMS codec layers.

For call flow diagrams that include SMS alongside voice, see Call Flow Diagrams. For MAP interface details covering MO-ForwardSM and MT-ForwardSM operations, see MAP Operations. For SMSc address and point code configuration, see Configuration Reference. For common SMS delivery issues, see Troubleshooting — SMS Issues.

MO-SMS (Mobile Originating SMS)

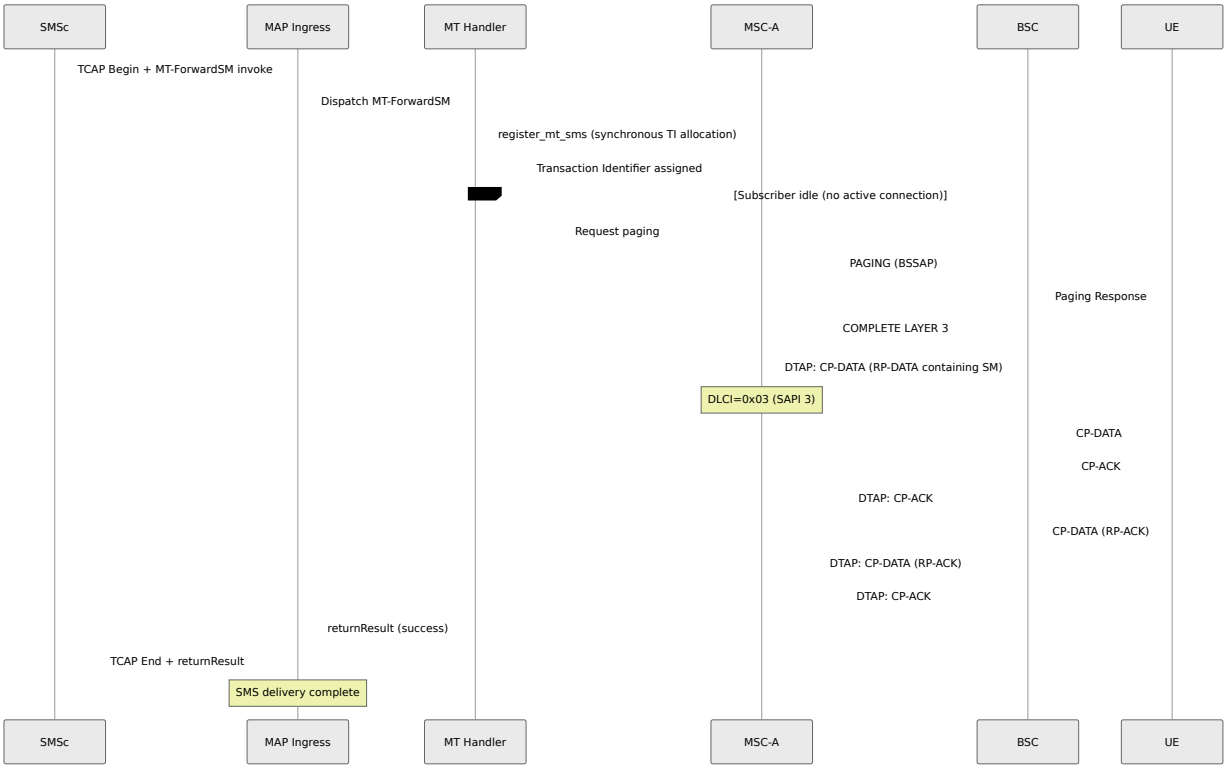
A subscriber sends a short message via the MSC to the SMS Centre (SMSc). The MSC acts as a relay, receiving the SM from the air interface and forwarding it to the SMSc using MAP MO-ForwardSM.



The MO Handler extracts the SM-RP-DA (destination address, typically the SMSc address) and SM-RP-OA (originating address, the subscriber MSISDN) from the RP-DATA, then constructs the MAP MO-ForwardSM request. The MAP response determines whether the MSC sends RP-ACK or RP-ERROR back to the UE.

MT-SMS (Mobile Terminating SMS)

The SMSc delivers a short message to a subscriber via the MSC. The SMSc sends a MAP MT-ForwardSM (opcode 44) to the MSC, which pages the subscriber if necessary and delivers the SM over the air interface.



Synchronous TI Allocation

The MT Handler calls `register_mt_sms` on the MSC-A process as a synchronous operation. This allocates a DTAP Transaction Identifier for the MT-SMS delivery and prevents a race condition where two concurrent MT-SMS deliveries to the same subscriber could be assigned the same TI. The synchronous call ensures that TI values are unique across all active SMS transactions for a given subscriber.

DTAP Transaction Identifier

For MT-SMS, the network allocates the Transaction Identifier (TI). Per 3GPP TS 24.007, the TI flag distinguishes the originator:

Direction	TI Flag	Meaning
Network → UE (CP-DATA)	0	Network originated this transaction
UE → Network (CP-ACK, RP-ACK)	1	UE responding to network-originated transaction

The MSC sets TI flag=0 in the CP-DATA sent to the UE. The UE mirrors the TI value but sets flag=1 in all responses (CP-ACK, CP-DATA containing RP-ACK). This convention allows both sides to distinguish between multiple concurrent SMS transactions.

SAPI 3

SMS NAS PDUs (CP-DATA, CP-ACK, CP-ERROR) are carried on SAPI 3 of the A-interface per 3GPP TS 48.006. The DLCI (Data Link Connection Identifier) byte in the BSSAP DTAP header is set to 0x03, indicating SAPI=3.

SAPI 3 provides a separate logical channel from the main signaling channel (SAPI 0), which carries CC and MM messages. This separation allows SMS delivery to occur concurrently with an active voice call without interfering with call control signaling.

MAP Response Routing

When an MT-ForwardSM arrives from the SMSc, the MSC must route the TCAP End response back to the correct originating point code. The ingress module captures the OPC (Originating Point Code) from the incoming M3UA transfer message and stores it as `routing_info[:opc]`.

When constructing the TCAP End response, the MSC uses this stored OPC as the DPC (Destination Point Code) for the outgoing M3UA message. This ensures the response reaches the correct SMSc, which is important in networks where

multiple SMSc instances use different point codes, or where an STP routes based on point code rather than SCCP global title.

The OPC/DPC swap follows the standard M3UA convention: the response OPC is the MSC's own point code (the incoming DPC), and the response DPC is the SMSc's point code (the incoming OPC).

SMS Codec

The SMS codec handles two protocol layers per 3GPP TS 24.011:

CP Layer (Connection Management Sublayer)

Message	Direction	Description
CP-DATA	Both	Carries an RP message as payload
CP-ACK	Both	Acknowledges receipt of CP-DATA
CP-ERROR	Both	Reports a CP-layer error (cause value included)

CP-DATA contains a single RP-layer PDU. Each CP-DATA must be acknowledged with a CP-ACK before the next CP-DATA can be sent on the same transaction.

RP Layer (Relay Protocol)

Message	Direction	Description
RP-DATA	Both	Carries the SM-TP-DU (the actual short message) along with RP-DA and RP-OA addresses
RP-ACK	Both	Confirms successful RP-DATA delivery
RP-ERROR	Both	Reports an RP-layer error (cause value from TS 24.011 table 8.4)

For MO-SMS, the RP-DATA from the UE contains the SM-RP-DA (SMSc address) and SM-RP-OA (subscriber address). For MT-SMS, the MSC constructs RP-DATA with SM-RP-DA (subscriber IMSI) and SM-RP-OA (SMSc address).

References

Specification	Title	Relevance
TS 24.011	Point-to-Point Short Message Service support on mobile radio interface	CP and RP protocol layers, message formats, cause codes
TS 29.002 Section 12	MAP Specification - Short Message Service procedures	MAP MO-ForwardSM (opcode 46), MT-ForwardSM (opcode 44), routing info for SM
TS 23.040	Technical realization of SMS	SM-TP layer encoding, validity period, status reports
TS 48.006	Signalling transport mechanism specification for the BSC-MSC interface	DLCI/SAPI assignment for A-interface DTAP
TS 24.007	Mobile radio interface signalling layer 3 - General aspects	Transaction Identifier allocation and TI flag conventions

Supplementary Services

This document describes the supplementary services implemented in OmniMSC, covering the SS dispatch mechanism, call forwarding, call barring, call waiting, line identification, call hold, multi-party conferencing, and HLR interaction. For call flow sequences, see Call Flow Diagrams. For configuration parameters, see Configuration Reference. For advanced call features such as ECT and CCBS, see Advanced Call Features. For the MAP operations used to relay SS changes to the HLR (RegisterSS, ActivateSS), see MAP Operations. For MPTY conference bridge details at the media layer, see Media Control.

SS Dispatch Overview

A mobile station interacts with supplementary services by sending DTAP messages from the SS message group: SS REGISTER, SS FACILITY, and SS RELEASE COMPLETE, as defined in 3GPP TS 24.010. Each message carries a facility information element containing ASN.1-encoded component(s) with an operation code and SS code that identify the target service. The provisioned supplementary services for each subscriber, along with their status badges, are visible in the Subscribers page of the control panel — see Control Panel Guide.

When OmniMSC receives one of these messages, the CC/SS dispatcher extracts the SS code from the facility IE and routes the request to the appropriate service handler. The SS code is a single-byte value defined in 3GPP TS 29.002 that uniquely identifies the supplementary service being invoked.

SS Operations

Every supplementary service supports a subset of the five standard operations:

Operation	Description
Register	Provision the service with parameters (e.g., forwarded-to number, no-reply timer)
Erase	Remove a previously registered service
Activate	Enable a registered service so that it takes effect
Deactivate	Disable an active service without removing the registration
Interrogate	Query the current status and parameters of a service

The MS encodes these operations using invoke components within the facility IE. OmniMSC processes the request, determines whether HLR involvement is required, and returns a return result or return error component in the response.

Call Forwarding

OmniMSC implements the four 3GPP call forwarding variants defined in TS 24.082. Each variant can be provisioned per subscriber via MAP INSERT SUBSCRIBER DATA from the HLR, or managed by the subscriber using SS Register, Activate, Deactivate, Erase, and Interrogate operations.

Variant	SS Code	Trigger	3GPP Reference
Call Forwarding Unconditional (CFU)	0x21	All incoming calls	TS 24.082 Sec 4.1
Call Forwarding on Busy (CFB)	0x29	Called subscriber busy	TS 24.082 Sec 4.2
Call Forwarding on No Reply (CFNRy)	0x2A	No answer within timer	TS 24.082 Sec 4.3
Call Forwarding on Not Reachable (CFNRc)	0x2B	Subscriber not reachable	TS 24.082 Sec 4.4

CFU (Call Forwarding Unconditional)

All calls to the subscriber are unconditionally forwarded to the configured forwarded-to number. When CFU is active, the MSC does not page the subscriber. CFU takes priority over all other forwarding variants.

CFB (Call Forwarding on Busy)

Calls are forwarded when the subscriber is busy (all traffic channels occupied or user-determined user busy). The MSC detects the busy condition from either the radio interface (UDUB) or the CC FSM state.

CFNRy (Call Forwarding on No Reply)

Calls are forwarded when the subscriber does not answer within a configurable timer (default 20 seconds, range 5-30 seconds per 3GPP TS 24.082). The no-reply timer starts when the paging response or alerting indication is received.

CFNRc (Call Forwarding on Not Reachable)

Calls are forwarded when the subscriber cannot be reached: IMSI detached, no paging response, or radio link failure. The MSC determines reachability from the VLR subscriber state and paging outcome.

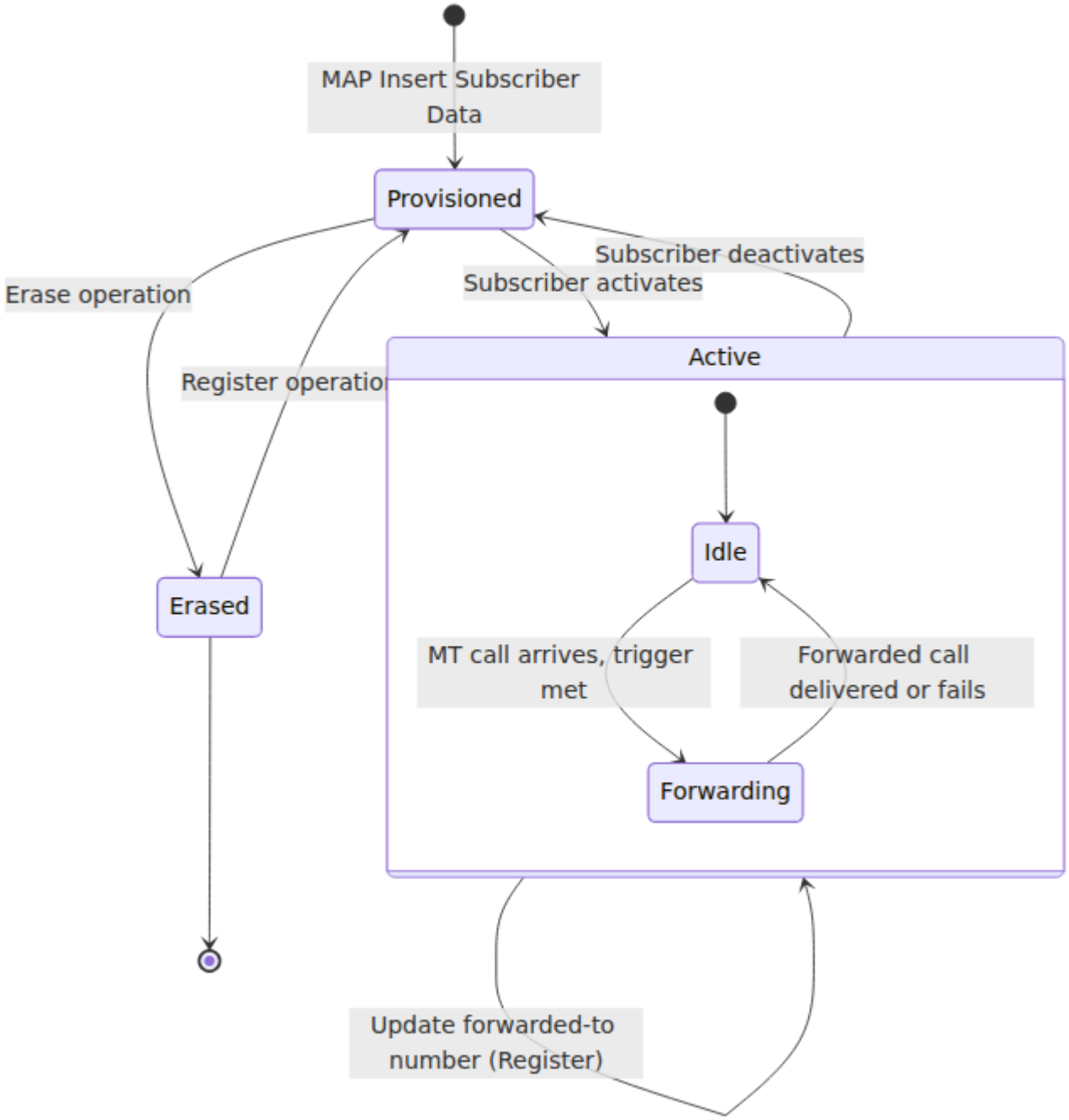
Call Forwarding Operations

A subscriber manages call forwarding through the standard SS operations:

Operation	Effect
Register	Store forwarded-to number and optional no-reply timer; implicitly activates the service
Erase	Remove forwarded-to number and deactivate the service
Activate	Enable a previously registered forwarding variant
Deactivate	Disable forwarding without removing the registered number
Interrogate	Return current status (active/inactive) and forwarded-to number

Registration and erasure operations are relayed to the HLR via MAP, since the forwarding data is part of the subscriber profile. Activation, deactivation, and interrogation can be handled locally from VLR data when the information is already available, or relayed to the HLR otherwise.

Call Forwarding State Diagram



Call Barring

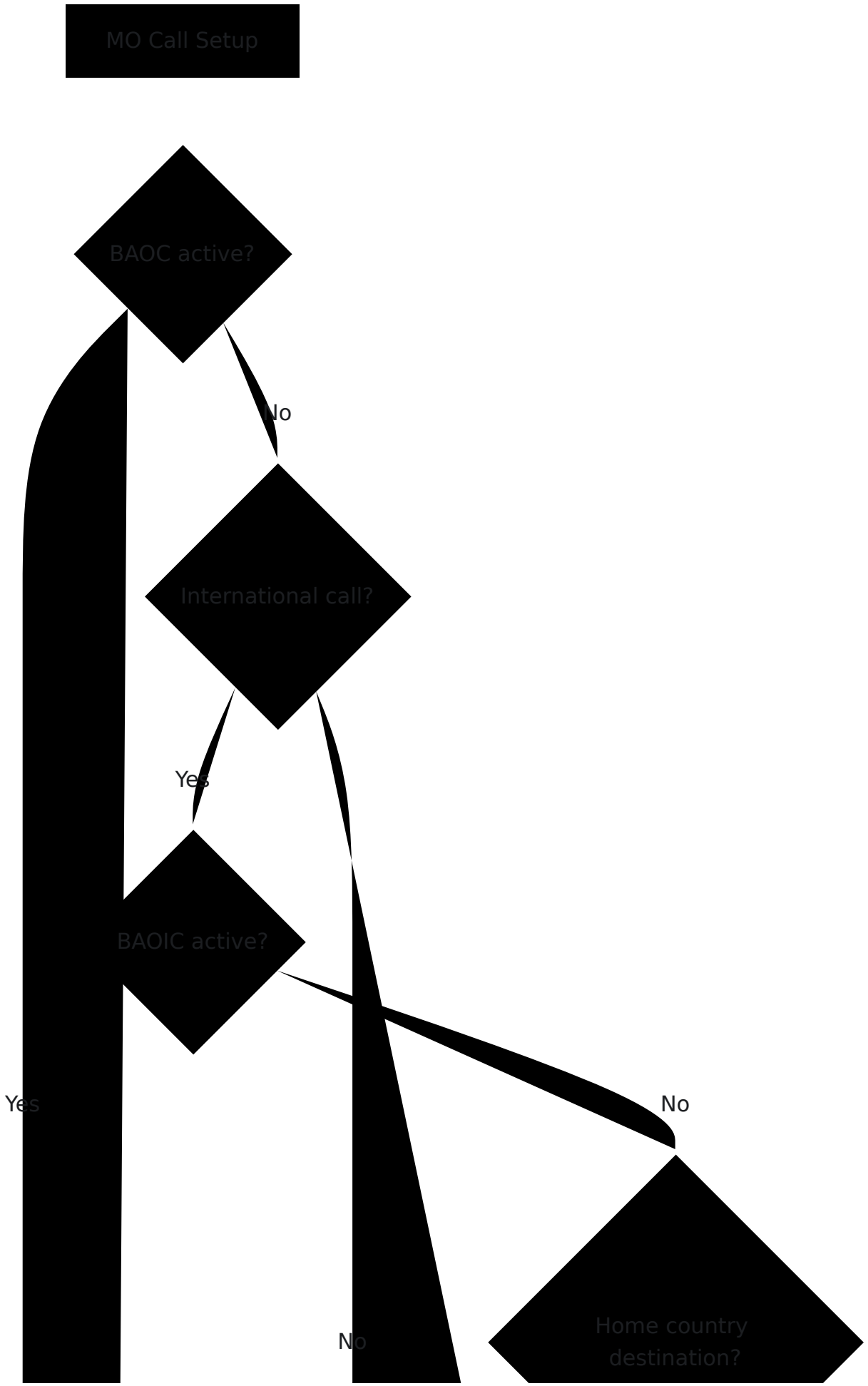
OmniMSC implements call barring services per 3GPP TS 24.088. Barring categories are provisioned by the HLR via MAP INSERT SUBSCRIBER DATA and are enforced by the MSC during call setup. The subscriber cannot override operator-provisioned barring.

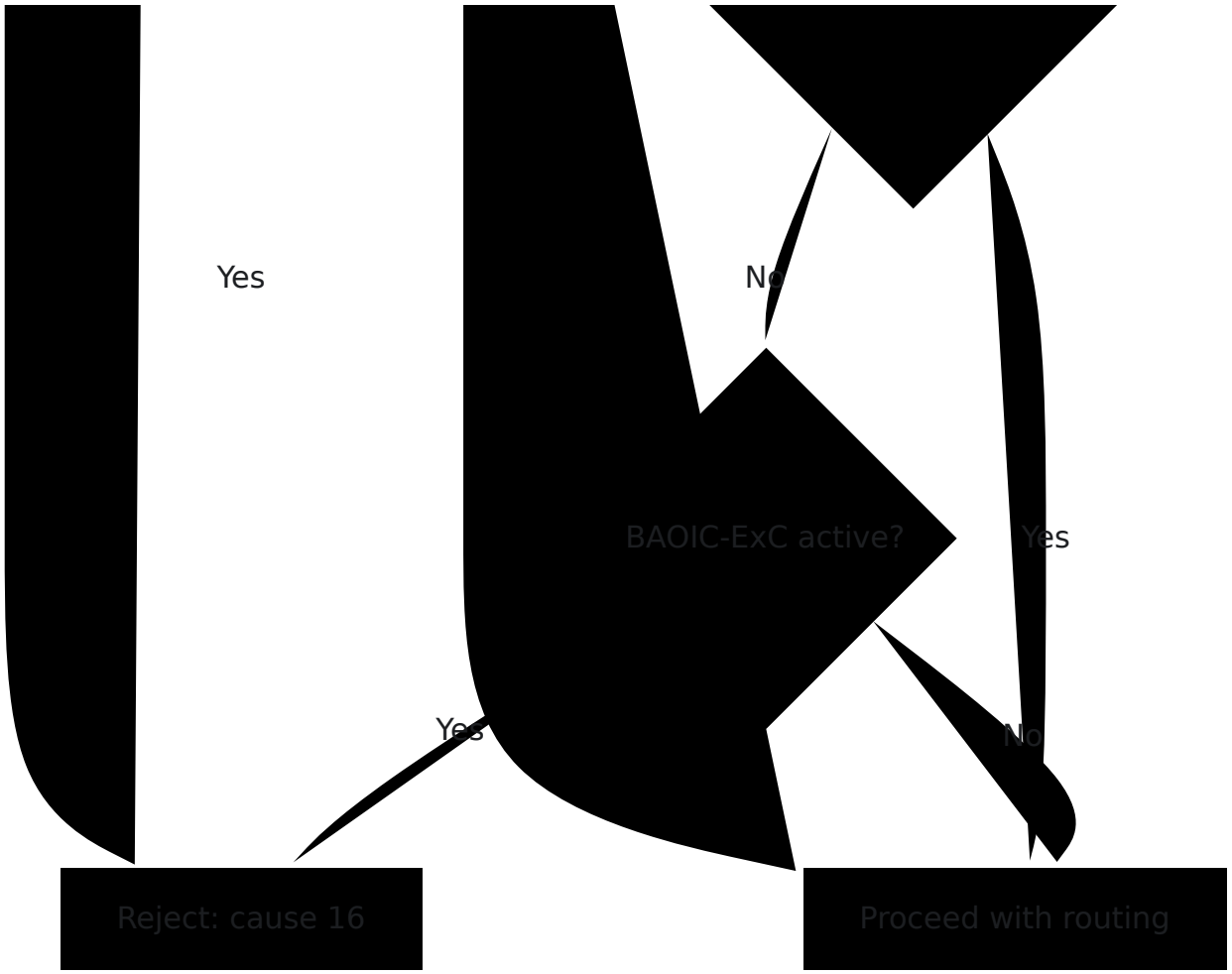
Category	SS Code (within barring group)	Direction	Effect
Barring of All Outgoing Calls (BAOC)	0x21	MO	Block all outgoing calls
Barring of Outgoing International Calls (BAOIC)	0x22	MO	Block international outgoing calls
Barring of Outgoing International Calls except to Home Country (BAOIC-ExC)	0x23	MO	Block international outgoing calls except to home PLMN country
Barring of All Incoming Calls (BAIC)	0x24	MT	Block all incoming calls
Barring of Incoming Calls when Roaming (BAIC-Roam)	0x25	MT	Block incoming calls when subscriber is roaming outside HPLMN

Barring Provisioning

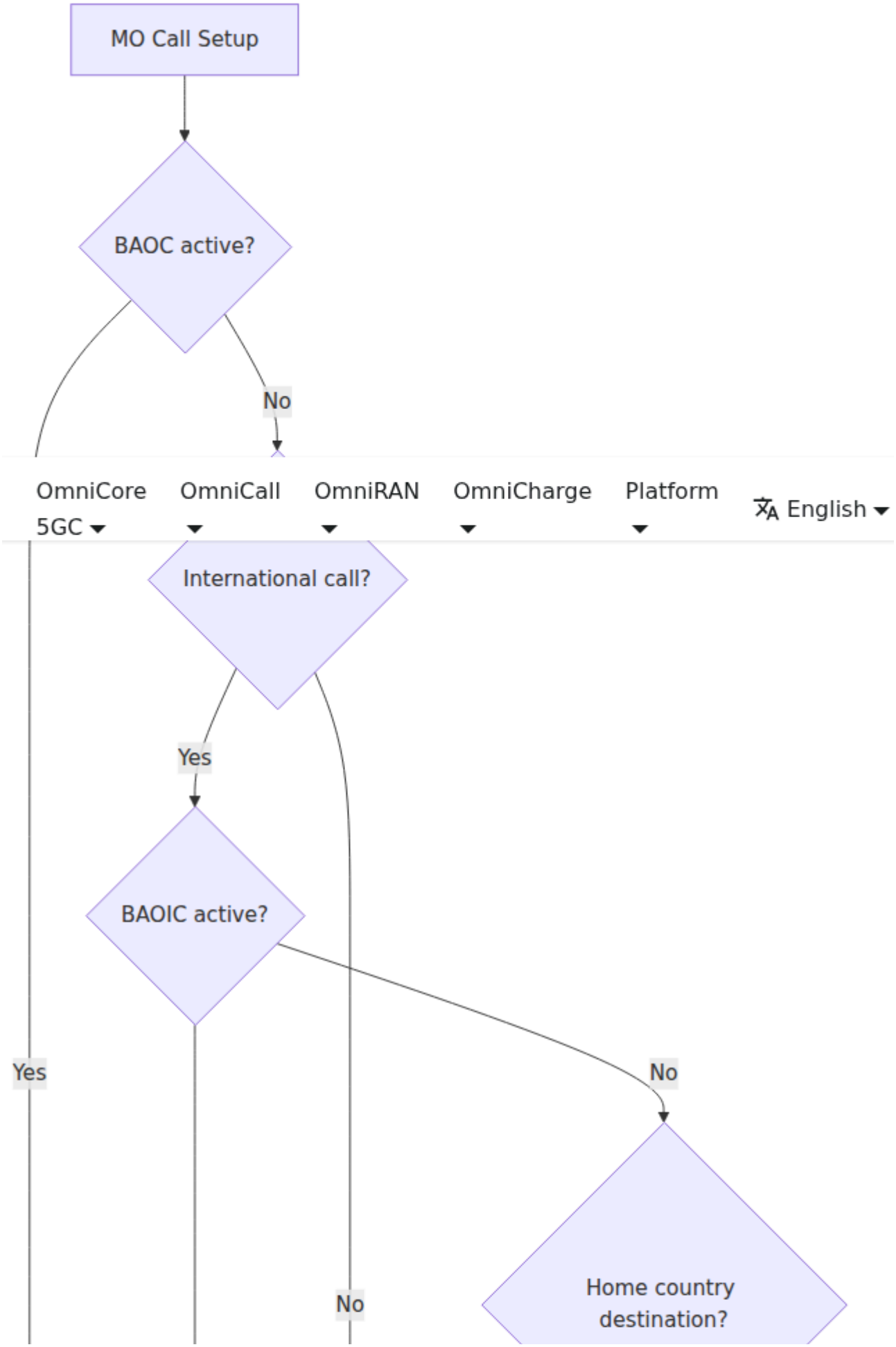
Call barring data arrives from the HLR as part of the subscriber profile in the MAP INSERT SUBSCRIBER DATA operation during Location Update. The VLR stores the active barring categories and the MSC evaluates them at call setup time. Since barring is operator-controlled, activation and deactivation by the subscriber require HLR authorization via a barring password.

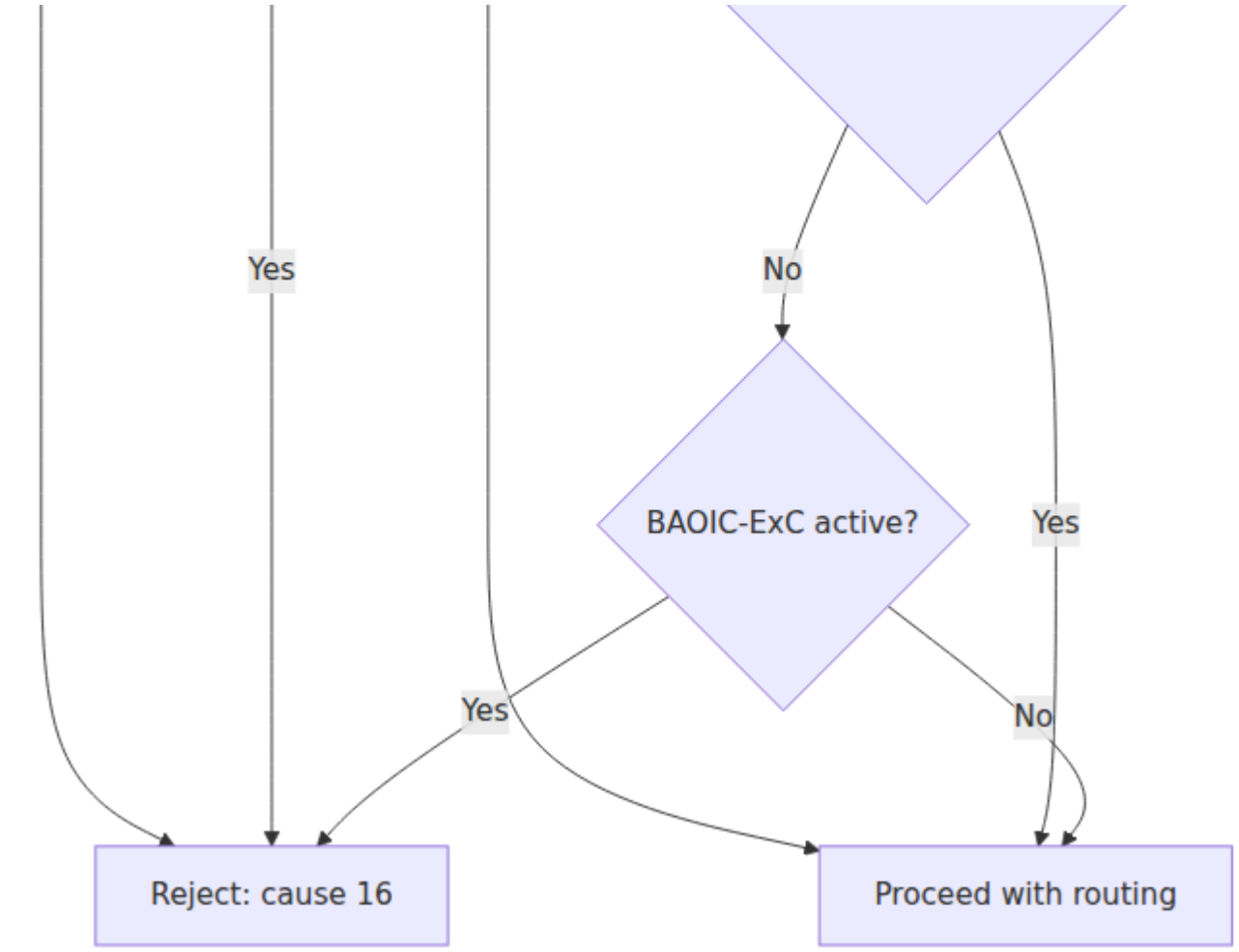
Barring Evaluation -- Outgoing Calls


MO Call Setup
BAOC active?
No
International call?
Yes
BAOIC active?
Yes
No
No
Home country destination?



Barring Evaluation -- Incoming Calls


MO Call Setup
BAOC active?
No
OmniCore 5GC
OmniCall
OmniRAN
OmniCharge
Platform
English
International call?
Yes
BAOIC active?
Yes
No
No
Home country destination?

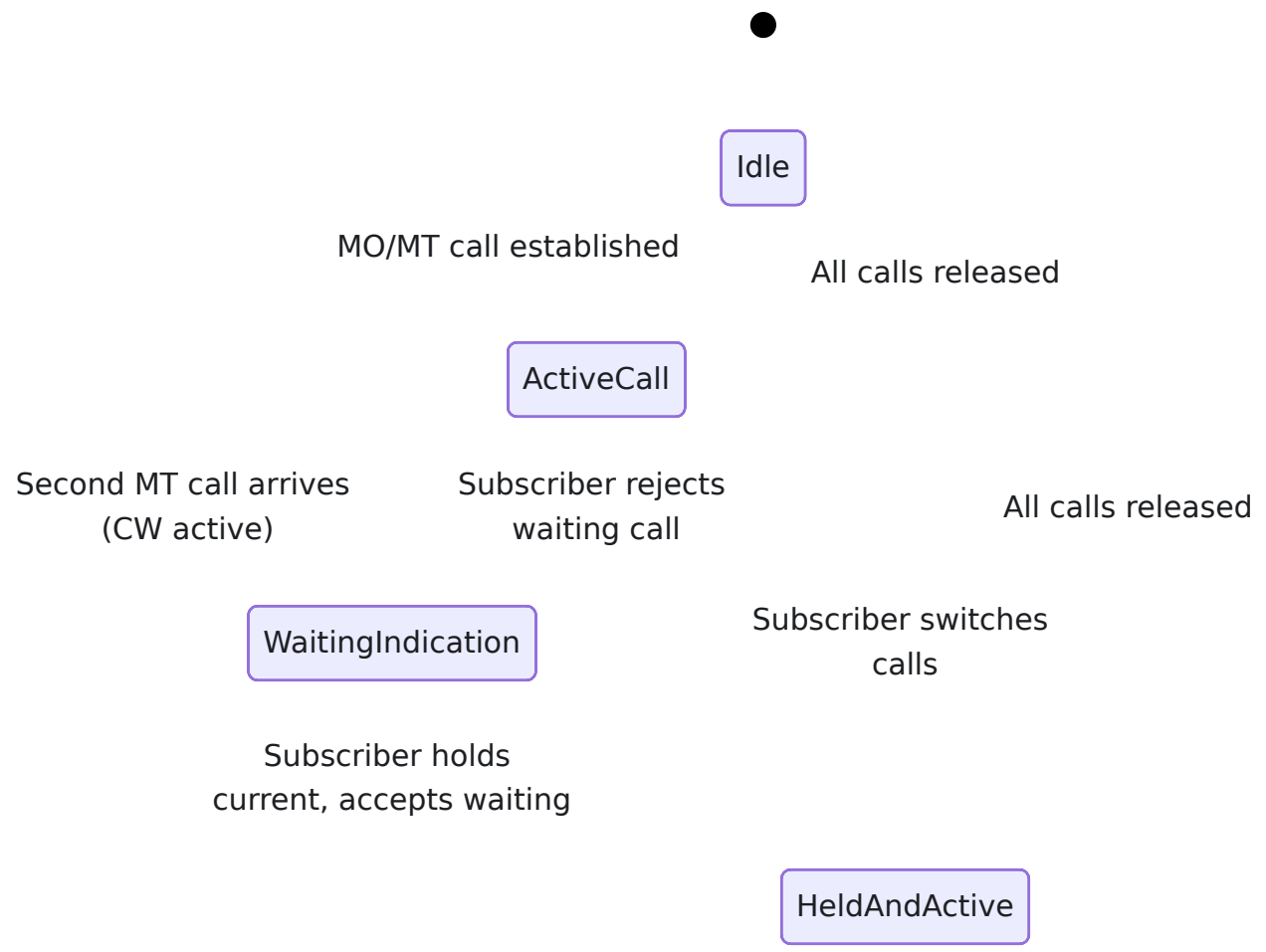


Call Waiting

OmniMSC implements Call Waiting (CW) per 3GPP TS 24.083. Call Waiting has SS code 0x41. The subscriber can activate or deactivate Call Waiting using the standard SS Activate and Deactivate operations.

When a subscriber with CW active is engaged in a call and receives a second incoming call, the MSC includes a Signal information element in the SETUP message sent to the mobile station. The Signal IE instructs the MS to play a call-waiting tone, alerting the subscriber to the incoming call. Without CW active, the MSC returns a busy indication to the second calling party, which may trigger CFB if configured.

The subscriber may then accept the waiting call by placing the current call on hold (HOLD message) and answering the new call, or reject the waiting call, which proceeds to normal busy handling.



Line Identification

OmniMSC implements Calling Line Identification Presentation (CLIP) and Calling Line Identification Restriction (CLIR) per 3GPP TS 24.081 and TS 24.083.

CLIP (Calling Line Identification Presentation) -- SS Code 0x11

CLIP provides the called party with the calling party number. The MSC includes the calling number in the SETUP message sent to the mobile station on the MT leg, and in the IAM or SIP INVITE on the outgoing trunk for MO calls. CLIP is a network-provisioned service: when CLIP is provisioned for the called subscriber, the network presents the calling number subject to CLIR restrictions of the calling party.

CLIR (Calling Line Identification Restriction) -- SS Code 0x12

CLIR allows the calling party to restrict presentation of their number to the called party. The HLR provisions the default CLIR mode, and the subscriber can invoke per-call overrides where permitted.

Mode	Behavior
Permanent	Number is always restricted; per-call override not available
Temporary (default restricted)	Number restricted unless subscriber invokes per-call presentation
Temporary (default allowed)	Number presented unless subscriber invokes per-call restriction

The MSC evaluates the CLIR setting of the calling subscriber when populating the Calling Party Number in outgoing signaling. A network operator override can force presentation regardless of CLIR setting, for scenarios such as lawful intercept and emergency callback.

Call Hold

OmniMSC handles Call Hold per 3GPP TS 24.083. The mobile station places an active call on hold by sending a HOLD message and retrieves it by sending a RETRIEVE message. Both are CC messages handled directly by the CC FSM within the MSC.

When the MSC receives a HOLD message, the CC FSM transitions the call leg to the held state, instructs the media gateway to set the held leg to receive-only, and sends a HOLD ACKNOWLEDGE back to the MS. On RETRIEVE, the CC FSM reverses the process: the media gateway leg returns to send-receive mode, and the MSC responds with RETRIEVE ACKNOWLEDGE.

Call Hold is a prerequisite for several other supplementary services including Call Waiting (accepting a second call), Multi-Party conference (bridging held and active calls), and Explicit Call Transfer (connecting held and active remote parties).

Multi-Party (MPTY) Conference

OmniMSC implements Multi-Party calling per 3GPP TS 24.084. MPTY allows a subscriber to bridge multiple calls into a conference. The conference bridge is hosted on the media gateway, with the MSC controlling participation through CC FACILITY invoke and return result exchanges.

MPTY Operations

Operation	SS Code	Description
BuildMPTY	0x51	Create or extend the conference by merging the held and active calls
HoldMPTY	0x52	Place the entire conference on hold
RetrieveMPTY	0x53	Retrieve a held conference
SplitMPTY	0x54	Extract one party from the conference into a private call

BuildMPTY Flow

The subscriber first establishes two calls (one active, one held) using normal call setup and the HOLD procedure, then sends a BuildMPTY invoke in a CC FACILITY message. OmniMSC creates a conference context on the media gateway, redirects all call legs into the conference bridge, and returns a BuildMPTY return result to the MS. Subsequent calls can be added to the

conference by repeating the hold-and-dial pattern followed by another BuildMPTY invoke.

HoldMPTY and RetrieveMPTY

HoldMPTY places all conference legs into receive-only mode on the media gateway, allowing the subscriber to make a private call outside the conference. RetrieveMPTY reverses this, restoring all legs to send-receive mode and reconnecting the subscriber to the conference.

SplitMPTY

SplitMPTY extracts a single party from the conference into a private two-party call with the subscriber. The remaining conference participants are placed on hold. The MS identifies the party to be split by call reference in the SplitMPTY invoke.

Conference Bridge on MGW

The media gateway maintains the conference context with one termination per participant. OmniMSC uses CRCX to create terminations and MDCX to add or remove them from the conference context. The gateway performs audio mixing for all participants in the context.

HLR Relay

Several SS operations require interaction with the HLR, since the subscriber profile is the authoritative source for supplementary service state. When the MSC receives an SS operation that modifies persistent state (Register, Erase, Activate, Deactivate for forwarding, barring password verification), the request is relayed to the HLR via the appropriate MAP operation. The HLR processes the request, updates the subscriber profile, and returns the result, which the MSC forwards to the mobile station.

Operations relayed via MAP include:

- RegisterSS / EraseSS for call forwarding numbers
- ActivateSS / DeactivateSS for service state changes that must persist beyond the current VLR registration
- RegisterPassword for call barring password management
- InterrogateSS when the VLR does not hold the requested data

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Local Interrogation

When the VLR already holds current supplementary service data (received during Location Update via INSERT SUBSCRIBER DATA), interrogation requests can be answered locally without contacting the HLR. This reduces signaling load and response time. The MSC queries the VLR subscriber record for the requested SS code and returns the current status directly to the mobile station.

Local interrogation is used for status queries where the VLR data is known to be fresh, such as immediately after a Location Update or after a successful Register/Activate/Deactivate operation that updated the local VLR copy.

3GPP Specification References

Specification	Title	Relevance
TS 24.010	Mobile Radio Interface Layer 3 -- Supplementary Services	SS message structure, facility IE, component types
TS 24.080	Mobile Radio Interface Layer 3 SS Specification -- Formats and Coding	ASN.1 encoding of SS operations and components
TS 24.081	Line Identification Supplementary Services	CLIP, CLIR
TS 24.082	Call Forwarding Supplementary Services	CFU, CFB, CFNRy, CFNRc
TS 24.083	Call Waiting and Call Hold Supplementary Services	CW, HOLD, RETRIEVE
TS 24.084	Multi-Party Supplementary Services	BuildMPTY, HoldMPTY, RetrieveMPTY, SplitMPTY
TS 24.088	Call Barring Supplementary Services	BAOC, BAOIC, BAOIC-ExC, BAIC, BAIC-Roam
TS 29.002	MAP Specification	SS code assignments, MAP operations for SS provisioning

Troubleshooting Guide

This document covers common operational issues encountered with OmniMSC deployments and their resolution steps. For configuration reference, see Configuration Reference. For system metrics and alerting, see Metrics Reference. For real-time status and subscriber inspection, see Control Panel Guide.

SS7 Connectivity

STP Link Down

Symptoms: No new calls are established, location updates do not complete, the `sctp_link_down` alarm is raised, and `omnimsc_peer_status` shows 0 for the affected STP peer.

Investigation: Verify that the SCTP association to the STP is established. Check that the remote IP address and port match the STP configuration. Confirm that SCTP protocol 132 is permitted through any firewalls between the MSC and STP. Verify the STP routing context matches the value configured in OmniMSC. Check that the M3UA ASP state machine has progressed through ASPUP and ASPAC to the ACTIVE state. If the association establishes but M3UA does not reach ACTIVE, check the M3UA logs for ERR chunks indicating a routing context or traffic mode mismatch.

Resolution: Correct the SCTP endpoint configuration (IP, port), firewall rules, or M3UA routing context as indicated by the investigation. Once the SCTP association re-establishes and M3UA reaches ACTIVE, the alarm clears automatically and signaling resumes.

MAP Dialogue Timeout

Symptoms: Location updates hang without completing, SMS delivery fails, the `hlr_unreachable` alarm may be raised. MAP operations time out without

receiving a TC-CONTINUE or TC-END from the HLR.

Investigation: Confirm that the HLR is reachable at the network level. Verify that point codes are correctly configured on both ends. Check that Global Title translation is correctly configured if GT-based SCCP routing is in use. Verify that M3UA has an active route to the HLR point code. Inspect logs for TC-BEGIN messages being sent without corresponding TC-CONTINUE responses.

Resolution: Correct point code or Global Title configuration as needed. If the HLR is reachable but not responding, the issue is on the HLR side. Verify that the HLR has a return route to the MSC point code.

Wrong DPC for MAP Responses

Symptoms: The SMSc never receives the MT-SMS acknowledgement. MAP ForwardSM responses are sent but do not arrive at the originating SMSc. The MT-SMS delivery appears to complete from the MSC perspective but the SMSc retransmits the request.

Investigation: Check the Destination Point Code (DPC) in the M3UA DATA message carrying the MAP response. The response DPC must match the Originating Point Code (OPC) from the incoming MAP request, which is the SMSc point code. If the response is being sent to the STP or HLR point code instead of the SMSc, the routing_info OPC is not being used for the response DPC.

Resolution: Ensure that the MAP response uses the OPC from the incoming routing_info as the DPC for the return message. This guarantees the response reaches the correct SMSc point code rather than a default route.

Call Issues

One-Way Audio

Symptoms: Call is established (both parties hear ringing and answer) but audio flows in only one direction, or not at all.

Investigation: Verify that the MGCP CRCX (Create Connection) and MDCX (Modify Connection) messages are being sent to the media gateway and that the gateway responds with 200 OK. Check that the MGW is reachable and healthy. Inspect the SDP connection address (c= line) in the SIP 200 OK to confirm it contains a routable IP address, not a private or loopback address. If behind NAT, verify that the external_ip parameter is configured. Confirm that the RTP port range is open in both directions through any firewalls.

Resolution: Correct the MGW connectivity, SDP address configuration, or firewall rules as identified. For NAT deployments, configure the external_ip parameter in the SIP configuration.

SIP BYE Not Sent on Call Clear

Symptoms: When the A-interface connection is released (e.g., radio link failure, BSC clear), the SIP peer is not notified. The remote SIP endpoint keeps the session open until its own timer expires.

Investigation: Verify that the connection_lost event is being sent to the CC FSMs when MSC-A releases the connection. Each active call transaction must receive the release indication so that the CC FSM can trigger SIP BYE on the outgoing leg.

Resolution: Ensure that the MSC-A release handler iterates all active transactions and delivers the connection_lost event. The CC FSM should transition to the release state and send SIP BYE upon receiving this event.

Assignment Complete to Wrong CC FSM

Symptoms: A new call in an existing connection (e.g., a second MO call while the first is held) receives the Assignment Complete for the wrong transaction, causing call state corruption.

Investigation: Verify that the active_trans reference is updated when a new call starts while the connection is already in the communicating state. The Assignment Complete is delivered to whatever transaction is marked as active_trans at the time of receipt.

Resolution: Ensure that the MSC-A process updates active_trans to the new transaction when processing the second CM Service Request or call setup in the communicating state.

DTMF Not Working

Symptoms: In-call DTMF tones are not recognized by the remote party or IVR system. The subscriber presses digits but no action occurs.

Investigation: Verify that SIP INFO messages are being sent with content type application/dtmf-relay when DTMF events are received from the radio side. Confirm that the SIP peer supports the INFO method for DTMF relay. Check the SIP peer configuration for DTMF handling preferences.

Resolution: Ensure SIP INFO-based DTMF relay is enabled and that the peer is configured to accept it. If the peer requires RFC 2833 telephone-event RTP packets instead of SIP INFO, adjust the DTMF relay method accordingly.

SMS Issues

MT-SMS Delivery Timeout (TC1)

Symptoms: MT-SMS delivery initiated by the SMSc times out. The MAP ForwardSM operation does not complete within the TC1 timer. The SMSc retransmits the delivery attempt.

Investigation: Check that the register_mt_sms operation uses a synchronous call to ensure the DTAP CP-DATA is sent before the MAP response. Verify that the DTAP Transaction Identifier (TI) is correctly allocated for the MT direction. Confirm that SMS PDUs are being sent on SAPI 3 (the dedicated signaling channel for SMS) rather than SAPI 0.

Resolution: Ensure synchronous MT-SMS registration, correct TI allocation with the TI flag set for network-originated transactions, and SAPI 3 usage for SMS DTAP messages.

MT-SMS Re-Delivery Loop

Symptoms: The MT-SMS handler process crashes and restarts repeatedly, causing the same SMS to be re-delivered in a loop. The subscriber may receive duplicate messages.

Investigation: Check the child_spec of the MT-SMS handler process in the supervision tree. If the restart strategy is set to permanent, the supervisor will restart the handler unconditionally after every termination, including normal completion. This causes re-delivery because the restarted handler re-initiates the delivery.

Resolution: Set the MT-SMS handler child_spec to restart: :temporary so that normal termination does not trigger a restart. Only abnormal crashes should cause a restart, and the SMSc will handle re-delivery via its own retry mechanism.

MAP Response Not Reaching SMSc

Symptoms: MT-SMS delivery completes on the radio side (subscriber receives the message) but the SMSc does not receive the MAP ForwardSM response. The SMSc treats the delivery as failed and may re-deliver.

Investigation: Check the DPC in the M3UA DATA message carrying the MAP ForwardSM response. The DPC must match the SMSc point code, which is the OPC from the incoming MAP request (routing_info[:opc]). If the response is routed to the wrong point code, the SMSc never sees it.

Resolution: Ensure the MAP response DPC is derived from routing_info[:opc] of the incoming request, not from a default or HLR-associated point code.

Authentication

Auth Failure -- MAC Mismatch

Symptoms: Subscriber authentication fails with reason mac_failure. The UE sends Authentication Failure with cause "MAC failure" indicating the network authentication check failed on the USIM.

Investigation: This indicates a mismatch between the subscriber key (Ki/K) stored in the USIM and the key stored in the HLR/AuC. The authentication vector generated by the AuC does not match what the USIM computes. This can occur after SIM replacement, HLR migration, or provisioning errors.

Resolution: Verify that the subscriber key in the HLR/AuC matches the key programmed into the USIM. Re-provision the subscriber profile in the HLR if necessary. This is not an MSC-side issue -- the MSC faithfully relays vectors from the HLR.

Auth Resync Loop

Symptoms: The subscriber enters an authentication resynchronization cycle that repeats without resolution. The UE sends Authentication Failure with cause "SQN failure" and an AUTS parameter. The MSC relays the resync to the HLR, but subsequent authentication attempts also fail with SQN failure.

Investigation: OmniMSC permits a maximum of 2 resynchronization attempts per authentication cycle. Check whether the resync count has been exceeded. Verify that the HLR correctly processes the AUTS parameter and advances the SQN. If the HLR SQN is significantly out of range from the USIM SQN, a single resync may not be sufficient.

Resolution: If the resync loop persists after 2 attempts, the subscriber must be re-provisioned in the HLR. The HLR SQN must be reset to align with the USIM SQN. Contact HLR operations to resolve the SQN discrepancy.

Subscriber Issues

TMSI Unknown on Paging Response

Symptoms: A subscriber responds to paging using a TMSI that is not recognized by the MSC. This can occur after MSC restart or TMSI table corruption.

Investigation: When the MSC receives a Paging Response with an unknown TMSI, it does not reject the subscriber. Instead, the MSC sends an Identity Request to obtain the IMSI directly from the mobile station. This is normal recovery behavior per 3GPP TS 24.008.

Resolution: No action required. The MSC automatically resolves the unknown TMSI by requesting the subscriber's IMSI. If this occurs frequently, investigate whether TMSI reallocation is functioning correctly during location updates.

CM Service Request for Unknown IMSI

Symptoms: A subscriber sends a CM Service Request but is not registered in the VLR. The MSC does not have subscriber data for this IMSI.

Investigation: OmniMSC triggers an implicit location update when it receives a CM Service Request from an unregistered subscriber. The MSC initiates the full location update procedure (authentication, ciphering, MAP UPDATE LOCATION to HLR) before proceeding with the requested service.

Resolution: No action required. The implicit location update is automatic. If the location update itself fails (HLR unreachable, authentication failure), the CM Service Request is rejected and the subscriber must retry.

Subscriber Stuck in VLR

Symptoms: A subscriber record remains in the VLR despite the subscriber being unreachable or having moved to another MSC. Paging attempts for this subscriber consistently fail.

Investigation: Check whether the subscriber has completed a location update at another MSC (the HLR would have sent a MAP CANCEL LOCATION, which removes the VLR record). If the cancel was missed or the VLR record is otherwise stale, manual removal may be necessary.

Resolution: Use the DELETE /api/subscribers/{id} endpoint to purge the subscriber record from the VLR. The subscriber will re-register via location update when they next access the network.

Performance

Overload Rejection

Symptoms: New calls and location updates are rejected with GSM cause 42 (switching equipment congestion). The `overload` alarm is active.

Investigation: Check the current overload thresholds and the system metrics that are exceeding them. The four monitored metrics are: active calls (default max 10,000), registered subscribers (default max 50,000), BEAM process count (default max 500,000), and paging rate (default max 1,000/sec). Determine which metric is triggering the overload condition.

Resolution: If the thresholds are appropriate for the hardware, investigate the source of the overload: abnormal call volume, paging storms, or process leaks. If the thresholds are too conservative for the deployment, increase them in the overload configuration. See Configuration Reference.

High Memory Usage

Symptoms: BEAM memory consumption grows over time or spikes unexpectedly. The system may become unresponsive if memory is exhausted.

Investigation: Check ETS table sizes for the major in-memory stores: VLR subscriber table, MPTY conference contexts, and TCAP transaction table. Large numbers of stale entries in any of these tables can cause memory growth. Also

check for process leaks -- a rising BEAM process count without corresponding subscriber or call growth indicates processes that are not terminating correctly.

Resolution: Identify and resolve the source of the growth. Stale VLR entries can be purged via the subscriber API. TCAP transaction timeouts should clean up abandoned dialogues automatically. If process leaks are identified, investigate the supervision tree for processes with incorrect restart strategies or missing timeout handling.

SIP Issues

For SIP peer configuration and OPTIONS keepalive details, see SIP Trunking.

SIP Peer Showing UNKNOWN Status

Symptoms: A SIP peer appears with UNKNOWN status in the control panel or metrics, rather than UP or DOWN.

Investigation: OmniMSC determines SIP peer status via periodic OPTIONS keepalive messages. If OPTIONS keepalive is not configured for the peer, or if the peer has never responded to an OPTIONS request, the status remains UNKNOWN. Check whether the peer configuration includes keepalive settings. Verify that the peer is reachable and responds to SIP OPTIONS with 200 OK.

Resolution: Enable OPTIONS keepalive for the peer in the SIP configuration. If keepalive is enabled but the peer does not respond, investigate network connectivity and confirm the peer supports the OPTIONS method.

re-INVITE Retransmissions from Peer

Symptoms: A SIP peer sends re-INVITE requests during an established call (for codec renegotiation, session refresh, or hold/unhold) and retransmits them because the MSC does not respond. The peer may eventually terminate the call due to the unanswered re-INVITE.

Investigation: Verify that the MSC SIP stack handles in-dialogue re-INVITE messages when the call is in the active state. If the re-INVITE arrives during a state transition or is not matched to the correct dialogue, it may be silently dropped.

Resolution: Ensure the SIP transaction layer correctly matches in-dialogue re-INVITE to the existing call leg and that the CC FSM processes the re-INVITE in the active state. The MSC should respond with 200 OK containing an updated SDP answer.

Session Timer Expiry

Symptoms: Established calls are unexpectedly disconnected after a fixed interval (typically 1800 seconds). The SIP peer sends BYE with reason "Session Timer Expired."

Investigation: SIP session timers (RFC 4028) require periodic re-INVITE or UPDATE to keep the session alive. Check the Session-Expires header negotiation during call setup. Verify that the MSC respects the Min-SE (Minimum Session Expires) value proposed by the peer. If the MSC is the refresher, confirm that it sends re-INVITE or UPDATE before the session timer expires.

Resolution: Ensure the MSC participates in session timer negotiation and performs session refresh as required by the negotiated role (refresher or refreshee). If session timers are not needed in the deployment, the peer may need to be configured to not require them.

3GPP Specification References

Specification	Title	Relevance
TS 24.008	Mobile Radio Interface Layer 3	Authentication, identity, DTAP procedures
TS 29.002	MAP Specification	HLR operations, point code routing
TS 48.008	MSC-BSS Interface (BSSMAP)	Assignment, cipher mode, paging
TS 23.018	Handover Procedures	Inter-MSC handover
TS 22.101	Service Principles	Emergency call, overload

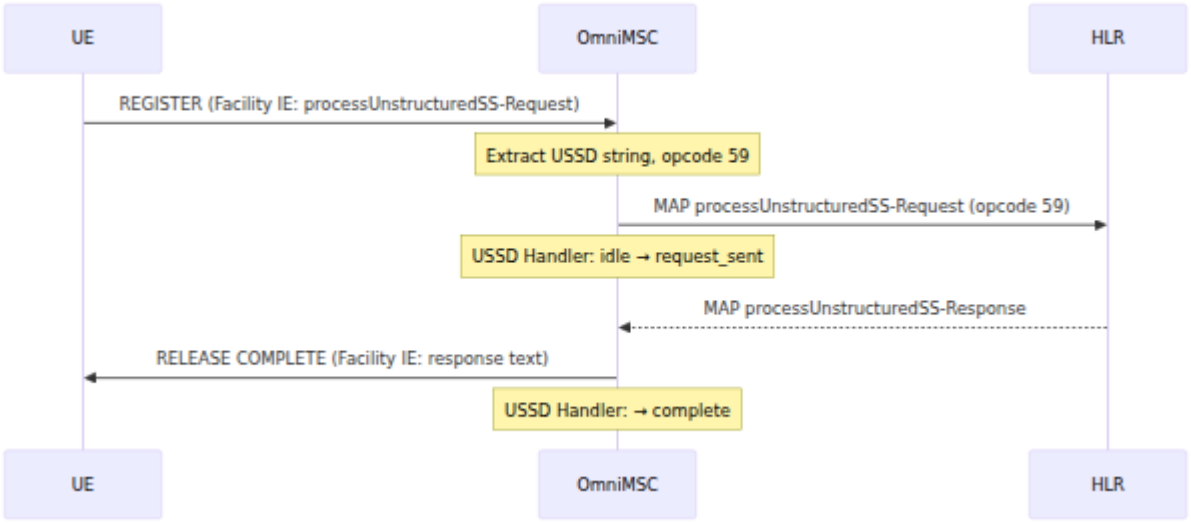
USSD

This document describes the Unstructured Supplementary Service Data (USSD) implementation in OmniMSC, covering HLR relay, local handling, the diagnostic easter egg menu, the USSD codec, and the SS message flow.

For supplementary service codes handled via USSD (call forwarding MMI codes, call barring, call waiting), see Supplementary Services. For the MAP ProcessUnstructuredSS-Request operation and dialogue management, see MAP Operations. For USSD-related Prometheus metrics, see Metrics and Monitoring. For USSD gateway configuration, see Configuration Reference. For troubleshooting USSD issues, see Troubleshooting.

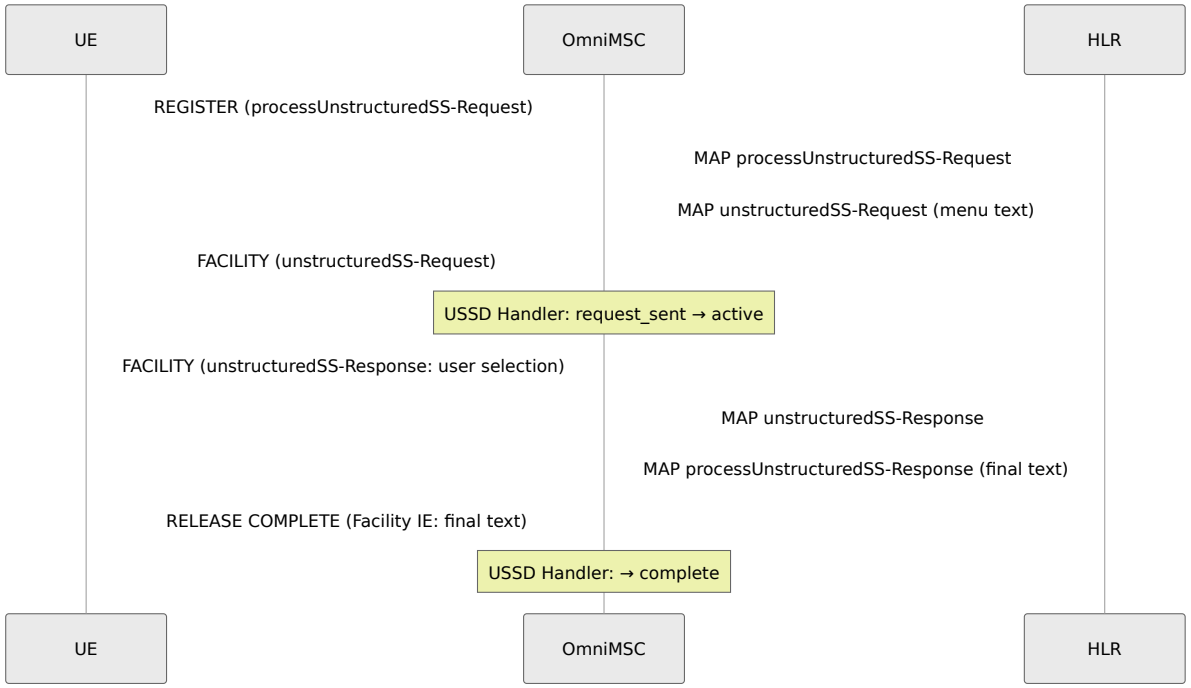
USSD Relay to HLR

When a subscriber sends a USSD string that is not handled locally by the MSC, the request is relayed to the HLR via MAP. The MS sends a REGISTER message containing a Facility IE with a processUnstructuredSS-Request (opcode 59). The MSC extracts the USSD string and forwards the request to the HLR using the same MAP operation.



For multi-step interactive sessions, the HLR responds with an unstructuredSS-Request (opcode 60) instead of a final response. The MSC relays this to the UE as a FACILITY message, and the UE's reply is forwarded back to the HLR. This

dialogue continues until the HLR sends a final processUnstructuredSS-Response.



Local USSD Handling

The MSC can handle certain USSD codes locally without forwarding to the HLR or an external gateway. Local handling is used for:

- Supplementary service activation/deactivation codes (CFU, CFB, CFNR, CFNRC, CLIP, CLIR, Call Barring, Call Waiting) using the standard MMI format defined in 3GPP TS 22.030
- Operator-defined diagnostic codes handled internally by the MSC

When a USSD string matches a locally handled service code, the USSD Handler routes it to the SS Handler or the appropriate internal module. The response is sent directly to the UE without MAP signaling.

For codes in the operator-defined range (100-199), the USSD Gateway module routes the request to an external USSD gateway if one is configured, or rejects the request if no gateway is available.

Easter Egg: System Diagnostic Menu

OmniMSC includes a hidden diagnostic menu accessible by dialing `*#6664#` (which spells `*#OMNI#` on a phone keypad). The MSC recognizes several keypad variants of this code and responds with an interactive USSD menu.

The main menu provides the following options:

Key	Menu Item	Information Displayed
1	System Status	Uptime, registered subscribers, active connections, active calls, memory usage, scheduler count
2	VLR	Registered subscriber count, active LU FSMs, active Auth FSMs
3	Active Calls	Count and summary of active calls (direction, calling/called numbers)
4	SS7 Stack	SCCP connection count, TCAP transaction count, pending paging requests
5	BEAM VM	Process count/limit, run queue depth, reduction count, memory breakdown (total, ETS, binary)
6	About	OmniMSC version info, OTP release, Omnitouch Network Services branding
9	Sales Enquiry	Contact information
0	Exit	Ends the session

Each submenu displays a "0. Back" option that returns to the main menu. The diagnostic data is gathered from live system metrics at the time of the request.

The easter egg is handled entirely within the MSC without any MAP signaling. The USSD Handler detects the magic code, routes it to the EasterEgg module, and manages the multi-step menu dialogue using network-initiated unstructuredSS-Request messages to the UE.

USSD Codec

The USSD codec handles encoding and decoding of USSD protocol elements per 3GPP TS 24.080 and TS 24.090.

Facility IE Structure

The Facility IE (tag 0x1C) wraps an ASN.1 BER-encoded component:

Component Type	ASN.1 Tag	Usage
Invoke	0xA1	Carries a request (processUnstructuredSS-Request, unstructuredSS-Request, unstructuredSS-Notify)
ReturnResultLast	0xA2	Carries a response (processUnstructuredSS-Response, unstructuredSS-Response)

Each component contains an invoke ID, an operation code, and an optional parameter SEQUENCE with the USSD data coding scheme and USSD string.

MAP Operation Codes

Opcode	Operation	Direction
59	processUnstructuredSS-Request	MO: UE → MSC → HLR
60	unstructuredSS-Request	MT: HLR → MSC → UE (interactive menu step)
61	unstructuredSS-Notify	MT: HLR → MSC → UE (notification, no response expected)

GSM 7-bit Encoding

USSD strings are encoded using the GSM 7-bit default alphabet defined in 3GPP TS 23.038. Each character is represented as a 7-bit septet, and septets are packed into octets (8 septets fit into 7 bytes). The data coding scheme byte (DCS=0x0F) indicates GSM 7-bit encoding.

For strings requiring characters outside the GSM 7-bit alphabet, UCS-2 encoding (DCS=0x48) is used, where each character occupies two bytes in big-endian UTF-16.

The maximum USSD string length is 182 characters for GSM 7-bit encoding (160 bytes packed) or 80 characters for UCS-2 (160 bytes).

SS REGISTER and RELEASE COMPLETE Message Flow

USSD sessions use the Supplementary Service (SS) message types defined in 3GPP TS 24.010:

Message	Direction	Purpose
REGISTER	UE → MSC	Initiates a new SS transaction. Contains a Facility IE with the USSD request.
FACILITY	Both	Carries mid-session data (interactive menu steps). Used for multi-step USSD dialogues.
RELEASE COMPLETE	Both	Ends the SS transaction. May contain a Facility IE with the final response.

A simple USSD request-response uses REGISTER (from UE) followed by RELEASE COMPLETE (from MSC). An interactive multi-step session uses REGISTER, then one or more FACILITY exchanges, then RELEASE COMPLETE.

The session has a configurable guard timeout (default 30 seconds). If neither the UE nor the gateway responds within this window, the MSC terminates the session with RELEASE COMPLETE.

References

Specification	Title	Relevance
TS 24.090	Unstructured Supplementary Service Data (USSD)	USSD procedures, session management
TS 29.002 Section 14	MAP Specification - Supplementary service operations	MAP processUnstructuredSS-Request (opcode 59), unstructuredSS-Request (opcode 60), unstructuredSS-Notify (opcode 61)
TS 24.080	Mobile radio interface layer 3 SS specification - Formats and coding	Facility IE structure, component encoding
TS 23.038	Alphabets and language-specific information	GSM 7-bit default alphabet, septet packing
TS 22.030	Man-Machine Interface (MMI)	Service code format, USSD string syntax
TS 24.010	Mobile radio interface layer 3 SS specification - General aspects	REGISTER, FACILITY, RELEASE COMPLETE message types