# Signaling & Media

Implementing Cisco Advanced Call Control and Mobility Services

KNOWLEDGE CHEATSHEET

**SEPARATE SIGNALING FROM MEDIA**

- **SIP / H.323** — Call control
- **SDP / H.245** — Negotiate media
- **RTP** — Carry media
- **RTCP** — Report quality

## 01 Signaling and Media Protocols 20%

### SIP dialogs and call changes

- **INVITE** starts session setup; **200 OK** accepts it and **ACK** completes the INVITE exchange. **BYE** ends an established session; **CANCEL** cancels a pending INVITE.
- **180 Ringing** indicates alerting. **183 Session Progress** conveys call progress and can carry SDP for early media. A ringing response alone does not prove that an RTP path exists.
- **REFER** requests a transfer; **Refer-To** identifies the transfer target. Track the original dialog and any replacement dialog when troubleshooting transferred calls.
- **Re-INVITE** and **UPDATE** can change session parameters. Hold/resume can alter SDP direction or media addresses; verify the new offer, answer and RTP destination after each change.
- **Session-Expires** and **Min-SE** negotiate session refresh timing. The refresher uses re-INVITE or UPDATE. A **422 Session Interval Too Small** response requires an acceptable interval; CUCM selects its refresh method in the SIP profile.

### Offers, early media and PRACK

- **Early offer** includes SDP in the initial INVITE. **Delayed offer** omits it; the offer and answer appear in later messages. Offer timing and pre-answer media are separate concepts.
- **Early media** carries announcements, tones or carrier prompts before answer. Check the provisional SDP, negotiated codec, reachable media address and bidirectional firewall policy.
- **100rel** makes supported provisional responses reliable. **PRACK** acknowledges them; **RSeq** and **RAck** correlate the exchange. ACK for a final INVITE response does not replace PRACK.
- Align CUCM **SIP Rel1XX Options** and CUBE **rel1xx** behavior with the peer. Requiring 100rel from an unsupported carrier can prevent call setup; sending PRACK for SDP-bearing provisional responses can be necessary for early media.
- A **180 with SDP** may request remote ringback instead of locally generated tones. CUBE **disable-early-media 180** changes this behavior; choose it only when the intended ringback source is understood.

# Signaling & Media / CME & SRST

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## SDP, RTP and audio diagnosis

- In **SDP**, **c=** supplies the connection address and **m=audio** supplies the media port and payload types. **a=rtpmap** identifies dynamic payload mappings; **a=fmtp** adds format parameters. **o=** identifies session ownership and revision.
- **sendrecv**, **sendonly**, **recvonly** and **inactive** describe media direction. During hold or resume, compare both legs; an unexpected inactive stream can explain silence even when the call remains connected.
- **RTP sequence numbers** reveal missing or reordered packets; **timestamps** represent sampling time for playout. **RTCP** reports loss, jitter and timing information that helps locate quality problems.
- **Signaling success** does not prove media reachability. Check routing, NAT, SDP rewrites, media-source validation and UDP ranges on each endpoint. Use the configured ranges; 16384-32767 is a common Cisco phone range, not a universal requirement.

## H.323, MGCP and trace tools

- **H.225** handles H.323 call signaling: SETUP, CALL PROCEEDING, ALERTING and CONNECT. **ALERTING** means the destination is being alerted; CONNECT indicates acceptance.
- **H.245** exchanges capabilities and logical channels. For slow-start calls, inspect **OpenLogicalChannel** and its acknowledgment for negotiated media transport information. Fast start carries channel proposals earlier in H.225 signaling.
- **MGCP** uses CUCM as call agent and a controlled gateway for bearer connections. **CRCX**, **MDCX** and **DLCX** create, modify and delete connections; inspect SDP and connection responses when signaling works but audio fails.
- Use **RTMT** Session Trace for SIP call flows and Trace and Log Central for CUCM logs. **CallManager SDI/SDL traces** expose routing and digit analysis. On IOS, **debug ccsip messages**, **debug h225 asn1**, **debug h245 asn1** and **debug mgcp media** target different protocols; limit debug scope and disable it afterward.

# 02 Gateway Technologies 10%

## CME phone registration

- **Cisco Unified CME** provides router-based call control. SIP phones use **voice register global**, **voice register dn** and **voice register pool**; SCCP phones use **telephony-service**, **ephone-dn** and **ephone**.
- Under **voice register global**, **mode cme** selects CME operation. **max-pool** limits SIP phone pools and **max-dn** limits directory-number objects; they count different resources.

# CME & SRST

Implementing Cisco Advanced Call Control and Mobility Services

KNOWLEDGE CHEATSHEET

**BRANCH CALL CONTROL**

**Phone** Register → **CME** Local control → **SRST** CUCM fallback → **Dial peers** PSTN access

- A **voice register dn** defines a directory number. A **voice register pool** represents a SIP phone and associates its identity, type and number assignments. Configure source addressing, SIP registrar support and the phone provisioning path.
- After changing phone configuration, regenerate the relevant **profiles** and apply the required phone reset or restart. Check IP connectivity, TFTP/provisioning, credentials and registrar settings before blaming the dial plan.

## CME dial plans and restrictions

- **Dial peers** route calls beyond local registered DNs. Match the destination, configure the session protocol and target, and align codec and DTMF settings on the remote system.
- A **SIP phone dial plan** controls local digit recognition and collection. Associate it with the relevant voice register pool; router call routing and handset digit collection are separate stages.
- **Class of Restriction (COR)** controls allowed calling paths using named capabilities. The incoming COR list must satisfy the outgoing dial peer's requirements. Assign phone-side restrictions at the appropriate SIP pool or SCCP DN.
- **Voicemail access** through the Messages button and **busy/no-answer forwarding** are separate settings. **Call park** uses a park slot with timeout and recall behavior; verify the configured interval, retry limit and recall destination.

## SRST survivability and toll fraud

- **SRST** supplies local call control when phones lose CUCM connectivity. Configure the CUCM SRST reference and device-pool assignment, reachable gateway addressing, local registration capacity and PSTN routing.
- **call-manager-fallback** configures SCCP SRST. SIP fallback uses **voice register global** with **mode srst** and SIP registrar settings. **mode esrst** enables supported enhanced SIP SRST functions such as shared lines and B-ACD.
- Fallback features depend on **platform, release and phone support**, especially for IPv6, secure registration, conferencing and fax. Test pickup, inbound DID routing, emergency dialing and restoration to CUCM during an outage.
- **Toll-fraud prevention** includes trusted VoIP sources, restricted dial plans and COR. Keep **ip address trusted authenticate** enabled and explicitly define trusted peers; disabling authentication removes a protection rather than fixing authorization.

# Cisco Unified Border Element

Implementing Cisco Advanced Call Control and Mobility Services

KNOWLEDGE CHEATSHEET

**CUBE CALL PROCESSING**

**Inbound peer** Classify → **Translation** Normalize → **Outbound peer** Route → **Media** Interwork

## 03 Cisco Unified Border Element 15%

### CUBE call legs and dial peers

- **CUBE** terminates and originates separate call legs as a back-to-back user agent. For SIP-to-SIP calls, enable the allowed connection type and configure **dial-peer voice <tag> voip** with **session protocol sipv2**.
- SIP inbound matching checks **URI Via, Request, To, From**, then number-based criteria such as **incoming called-number** and **answer-address**. Via matching identifies the neighboring SIP entity; ANI matching uses the calling number.
- **Outbound dial peers** select a route using the called destination or configured URI/policy mechanisms. An inbound match does not guarantee an outbound match. Check the post-translation number and selected session target.
- **OPTIONS keepalives** assess peer availability and can remove failed destinations from call routing. Verify responses, transport and network reachability before changing route patterns. Use **show dial-peer voice summary**, **show call active voice brief** and targeted call-control traces.

### Number translation and SIP profiles

- **voice translation-rule** defines ordered regular-expression rules. **voice translation-profile** assigns them to calling, called or redirecting numbers; apply the profile to the correct call leg and incoming/outgoing direction.
- **^** anchors the start, **$** anchors the end, **.** matches one character and **.*** matches a variable string. IOS capture groups use escaped parentheses; replacement **\1** reuses the first group and **\0** the full match.
- For example, **rule 1 /^9\(.*\)$/ /\1/** removes one leading access digit 9. Use **test voice translation-rule** with valid and nonmatching inputs; constrain length and prefixes when normalizing carrier numbers.
- **voice class sip-profiles** can add, remove or modify selected SIP/SDP fields. Distinguish **From** identity, **Diversion** forwarding identity and SDP **Connection-Info** media address. Inspect the actual outbound INVITE; changing one field does not necessarily update the others.

### Codecs and DTMF interworking

- **voice class codec** defines a preference list; apply it using **voice-class codec** on the dial peer. A supported offer-all option advertises multiple configured codecs. If call legs have no common codec, provide compatible **transcoding** resources.
- **rtp-nte** transports named telephone events in RTP, commonly called RFC 2833 or RFC 4733 DTMF. It differs from audible tones encoded in the voice codec. **sip-kpml** and **sip-notify** signal digits outside the RTP media path.

# Cisco Unified Border Element / Call Routing & Dial Plans

Implementing Cisco Advanced Call Control and Mobility Services

KNOWLEDGE CHEATSHEET

**CUBE CALL PROCESSING**

**Inbound peer** Classify → **Translation** Normalize → **Outbound peer** Route → **Media** Interwork

- Configure **dtmf-relay** on the VoIP dial peer and inspect negotiation on both legs. A **telephone-event** payload mapping is evidence of RTP event support; ordinary codec negotiation alone does not establish it.
- DTMF interworking may require **DSP resources**, especially conversion to audible in-band tones. Flow-around has additional limitations because media bypasses CUBE. For NTE-to-OOB duplicate digits, **dtmf-relay rtp-nte digit-drop** on the inbound leg suppresses the redundant NTE events.

## Bindings, media changes and secure trunks

- **bind control source-interface** selects SIP signaling addressing; **bind media source-interface** selects RTP addressing. Dial-peer **voice-class sip bind** settings can select a call leg's interface. Ensure every bound address, including a loopback, has a reachable return path.
- **Flow-through** anchors media on CUBE; **flow-around** lets endpoints exchange it directly. NAT and firewall rules must match the selected mode. SDP address rewriting cannot repair missing IP routes or blocked UDP traffic.
- **midcall-signaling passthru** forwards supported mid-call changes; **passthru media-change** limits passage to media-changing events. Align UPDATE/re-INVITE support with the carrier. **supplementary-service media-renegotiate** is relevant to media changes such as RTP/SRTP interworking after transfer.
- **TLS** protects SIP signaling and **SRTP** protects media. Configure certificate trust, peer identity, transport and media policy independently. **Tenants** separate supported trunk settings; registration-based trunks also need registrar, credentials and authentication-realm alignment.

## SIP recording and VoIP Trace

- **SIPREC** sends replicated media and session metadata to a recording server. Metadata describes participants, streams and their association, including changes during transfer; metadata itself is not the recorded RTP audio.
- **Network-based recording** can fork media at a supported gateway/CUBE. Verify recording signaling, media reachability and recorder capability separately from the primary call.
- **VoIP Trace** keeps an in-memory record of call activity for troubleshooting. Correlate its signaling and call identifiers with dial peers and media state; it complements targeted debugs and packet captures.

## 04 Call Control and Dial Planning 25%

## Patterns, partitions and search spaces

- A **partition** groups reachable numbers and patterns. A **calling search space (CSS)** is an ordered list of partitions available to a call. Inspect line, device and forwarding CSS contexts when a pattern exists but is unreachable.
- **Route patterns** send calls to gateways or route lists. **Translation patterns** change digits and trigger another routing analysis using the applicable CSS. **Transformation patterns** format calling or called numbers at a selected boundary.

# Call Routing & Dial Plans

Implementing Cisco Advanced Call Control and Mobility Services

KNOWLEDGE CHEATSHEET

**GLOBALIZED CALL ROUTING**

**Ingress** Normalize → **CSS** Permit → **Route list** Select exit → **Egress** Localize

- CUCM favors the **closest, most specific match** among reachable patterns. **X** matches one digit, **!** matches one or more digits, and **[0-5]** restricts a digit range. A pattern such as **813510[0-5]** targets six exact numbers.
- **Urgent Priority** routes a complete match without waiting for more digits. Use specific emergency patterns to avoid delay from overlapping longer patterns; **T302** is the interdigit timer, not the preferred per-pattern remedy.
- After translation, a broad **\+!** pattern can cause another wait. **Do Not Wait For Interdigit Timeout On Subsequent Hops** on a suitable translation pattern controls that subsequent analysis. Avoid loops between translation patterns.

## Globalization and digit presentation

- **Globalization** normalizes numbers to a consistent **+E.164** form for internal routing. **Localization** converts them at egress to the carrier's national or international format. Preserve a dialable external calling identity.
- Escape a literal plus as **\+** in a CUCM route pattern. In a pattern such as **9011.!**, the dot marks a discard boundary; **PreDot** removes the access and international prefix, after which a plus prefix can normalize the remaining number.
- **Calling party** transformations change ANI/caller ID; **called party** transformations change DNIS/destination. Keep masks, prefix digits, discard instructions, number type and numbering plan aligned with the trunk contract.
- **External Phone Number Mask** can supply a public calling number when the route configuration uses it. **Significant Digits** retains trailing inbound digits for a local DN plan; explicit DID translation handles nonuniform mappings.
- Do not assume transformations at route pattern, route-list detail and trunk/device boundaries form a simple cumulative chain. Check inheritance/override settings and compare original, transformed and transmitted digits in traces.

## Local route groups and TEHO

- The usual egress hierarchy is **route pattern → route list → route group → gateway/trunk**. Route lists and group distribution determine preferred and backup paths.
- **Standard Local Route Group** is a logical reference in a route list. Its physical route group is resolved from the relevant device-pool context, allowing shared route patterns with site-specific PSTN exits.
- Create **named local route groups** for distinct policies such as emergency and normal PSTN access. Map each name to a route group in device pools and reference it from the appropriate route list.
- **Tail End Hop Off (TEHO)** carries calls over the enterprise IP WAN to an exit near the destination. Validate WAN availability, media routing, permitted PSTN breakout and backup routing. For forwarded or roaming calls, verify the actual device-pool context used for local route-group selection.

# Call Routing & Dial Plans / CUCM Call Control

Implementing Cisco Advanced Call Control and Mobility Services

KNOWLEDGE CHEATSHEET

**GLOBALIZED CALL ROUTING**

**Ingress** Normalize → **CSS** Permit → **Route list** Select exit → **Egress** Localize

## SIP trunks, URI routing and analysis

- **SIP route patterns** route URI domains or supported address patterns to SIP trunks. Configure the destination trunk and partition, then grant access with CSS. A bare domain and its subdomains may require separate matches; do not assume email-style wildcard syntax.
- **Trunk security profiles** define transport/security behavior and incoming port. The **trunk destination port** controls the remote target. TLS often uses 5061, but both ends must agree on any explicitly configured port.
- For trunk failure, verify **DNS resolution**, selected CUCM node, TCP/TLS establishment, certificates, OPTIONS status and destination address. If no INVITE leaves CUCM, check routing and trunk availability before investigating RTP.
- **Dialed Number Analyzer (DNA)** simulates routing without a live call. Activate its required analyzer services, select the correct calling device/context and inspect partition access, translations and final route. Use actual CallManager traces to confirm runtime behavior.
- **Forced Authorization Codes (FAC)** authorize calls using an authorization level. **Client Matter Codes (CMC)** associate calls with billing/accounting codes. They serve different purposes; code collection is incompatible with CUCM overlap sending.

## 05 Cisco Unified CM Call Control Features 20%

## Admission control and media resources

- **Regions** constrain media codec/bit-rate relationships; **locations** model available call bandwidth. A low codec bit rate does not limit the number of simultaneous calls by itself.
- **Location Bandwidth Manager** performs admission control using the configured topology and bandwidth pools. Audio and video limits are separate. A Not Enough Bandwidth message can result from configured accounting even when a link looks lightly utilized.
- **Call Admission Control (CAC)** prevents oversubscription; **QoS** prioritizes admitted traffic. Size bandwidth for codec, packetization and overhead. **Automated Alternate Routing (AAR)** can send eligible CAC-denied calls via the PSTN using reachable external numbers and an appropriate CSS.
- **MTP** assists compatible signaling/media interworking; a **transcoder** changes codecs; a **conference bridge** mixes media. **MRGs/MRGLs** control resource selection. Check registration, capacity, codec support and DSP availability rather than adding MTPs to every trunk.

# CUCM Call Control

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KNOWLEDGE CHEATSHEET

**DISTRIBUTE INCOMING CALLS**

**Hunt pilot** — Called number → **Hunt list** — Select groups → **Line group** — Distribute → **Queue** — Wait or overflow

## ILS and global dial-plan replication

- **Intercluster Lookup Service (ILS)** exchanges intercluster routing information. **Global Dial Plan Replication (GDPR)** distributes advertised URIs and number patterns; it does not itself carry a voice call.
- Build an **ILS hub/spoke topology** with unique cluster IDs and reachable publishers. Configure shared-password authentication or **TLS certificate** authentication consistently; certificate-based ILS relies on publisher **Tomcat** trust.
- **CA-signed certificates** can simplify trust at scale. Validate CA chains, names and reachability on the participating clusters. Securing the SIP call trunk is separate from securing ILS exchanges.
- Review **advertised patterns**, locally assigned URIs and learned objects. The **ILS Max Number of Learned Objects in Database** parameter limits stored learned data. Learned routes still need a usable route string, SIP route pattern/trunk and appropriate CSS permissions.

## Hunt groups and native call queuing

- Configure **directory numbers → line group → hunt list → hunt pilot**. A line group contains DNs and defines distribution; the hunt pilot is the callable entry point.
- **Top Down** starts at the first eligible member; **Circular** advances the starting point; **Longest Idle Time** selects the longest-idle eligible member; **Broadcast** rings eligible members together. **RNA Reversion Timeout** controls how long an unanswered offer lasts.
- **Native Call Queuing** is configured on the hunt pilot. Set maximum queued callers, wait limits and destinations for overflow or unavailable members. Queue-enabled pilots use their queue handling instead of the ordinary Forward Hunt Busy/No Answer controls.
- Provide **Cisco IP Voice Media Streaming** resources and a supported unicast **MOH** source. Upload music through **MOH Audio File Management**; associate **Network Hold MOH Source & Announcements** on the hunt pilot and configure announcement files.
- **HLog** controls member login/logout. Queuing options can log out members who do not answer; logoff notification can alert logged-out members. Test agent eligibility, busy status, timeout, full-queue and no-member cases.

## Time schedules, park and pickup

- **Time-of-day routing** associates one time schedule with a partition; a schedule contains one or more time periods. Combine active/inactive partitions and suitable CSS access for business-hours routing; verify time zone and overlapping matches.
- **Call park** puts a call at a retrievable park number; reversion returns an unclaimed call after configured timers. **Call pickup** answers a ringing call: standard pickup serves the same group, group pickup selects a group, and other-group pickup uses associated groups.
- **BLF pickup** combines monitored speed-dial state with pickup permission; enable the pickup option on the BLF entry. **Meet-me** needs a reachable conference number, an available bridge through the initiator's MRGL and the Meet-me softkey.

# CUCM Call Control / Mobility

Implementing Cisco Advanced Call Control and Mobility Services

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**MOBILITY MECHANISMS**

- **Extension Mobility** — User profile
- **Device Mobility** — Site settings
- **Unified Mobility** — PSTN destination
- **MRA** — Remote UC client

## Hold music and recording sources

- **User Hold MOH** applies to user-initiated hold; **Network Hold MOH** applies to network-controlled scenarios such as transfer or queuing. Source selection follows the controlling device/feature; resource and codec selection must also succeed.
- **Built-in Bridge (BiB)** recording forks audio at a supported phone. The phone's **Default** BiB setting inherits the cluster service parameter. Gateway recording forks at a supported gateway; **SPAN-based recording** observes mirrored traffic.
- Recording needs both **session control** and a valid **media path**. Verify the recording profile, recording CSS/trunk, recorder reachability and selected recording source. A successful original call does not prove that the recorder receives both streams.

## 06 Mobility 10%

## Extension Mobility

- **Extension Mobility (EM)** loads a user's DNs and phone settings onto a supported desk phone. Activate the EM service, configure its phone-service URL and create a device profile associated with the end user.
- Enable EM on the **physical phone** and subscribe both the phone and the **device profile** to the EM service. A service visible before login can disappear after login if the profile lacks the subscription.
- Check **service URL**, reachability and returned XML when the service opens without a login prompt or shows a parse error. Check user ID/PIN, profile association and phone compatibility for authentication or profile-loading failures.
- **Logged-out CSS** should permit only the intended base privileges, including required emergency access. Grant additional permissions through the logged-in profile. Configure both maximum login duration and its enforcement; concurrency limits control a different behavior.

## Unified Mobility and call anchoring

- **Single Number Reach / Mobile Connect** rings supported remote destinations with the enterprise DN. Associate a **Remote Destination Profile (RDP)** with the user and desk DN, then add the mobile number as a remote destination.
- Enable the user's mobility permissions and give the RDP/rerouting CSS access to the mobile route. Configure schedules and **caller access lists** for permitted or blocked callers; compare the normalized incoming caller ID with the filter.

# Mobility

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**MOBILITY MECHANISMS**

- **Extension Mobility** — User profile
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- **Mobility** softkeys support movement between the desk phone and an anchored mobile call. **Answer Too Soon**, **Answer Too Late** and wait-before-ring settings distinguish acceptable mobile answers and help keep mobile voicemail from capturing calls.
- For desk pickup after mobile disconnect, examine **Maximum Wait Time for Desk Pickup**, gateway disconnect/progress signaling and **Retain Media on Disconnect with PI for Active Call** behavior. The enterprise must retain the anchored call long enough to present Resume.

## Device Mobility, MVA and remote access

- **Device Mobility** changes location-dependent settings when a phone roams between IP subnets. **Device Mobility Info** maps subnets to device pools; physical locations and mobility groups help determine roaming behavior.
- Verify subnet mapping, home/roaming **device pools**, physical locations and Device Mobility CSS. Correct region and location selection matters for codec negotiation and CAC; local route groups determine the relevant PSTN exit.
- **Mobile Voice Access (MVA)** lets a user dial into the enterprise and place an anchored outbound call with enterprise identity. Verify the access number, authentication, IVR digit handling and outgoing permissions; its gateway DTMF requirements differ from merely ringing a mobile destination.
- **Mobile and Remote Access (MRA)** uses Expressway-C/E for supported remote UC clients. Cloud collaboration services provide other mobility options. Keep remote client access, desk-phone EM, subnet-based Device Mobility and PSTN-based Unified Mobility distinct.