

MTAS SIP Trunking Management Guide

MTAS

USER GUIDE

Copyright

© Ericsson AB 2016. All rights reserved. No part of this document may be reproduced in any form without the written permission of the copyright owner.

Disclaimer

The contents of this document are subject to revision without notice due to continued progress in methodology, design and manufacturing. Ericsson shall have no liability for any error or damage of any kind resulting from the use of this document.

Trademark List

All trademarks mentioned herein are the property of their respective owners. These are shown in the document Trademark Information.



Contents

1	Introduction	1
1.1	Prerequisites	1
2	Overview	3
2.1	Subfunctions	3
3	Connection and Control Services in ST AS Configuration	11
3.1	ST AS Administrative State Configuration	11
3.2	Transit Function Address Configuration	12
3.3	SIP Trunking Charging Profile Reference	12
3.4	Feature Tag Handling Configuration	12
3.5	No Reply Timer Configuration	12
3.6	Access Control Profile Configuration	12
4	Fault Management	15





1 Introduction

This document describes how to configure the basic SIP Trunking Functionality provided by the SIP Trunking Application Server (ST AS) in MTAS. The basic SIP Trunking functionality in ST AS consists of managing of access and routing between the operators IMS Network and enterprises IP-Private Branch Exchange (IP-PBX), and to provide charging support.

1.1 Prerequisites

It is assumed that the user of this document is familiar with the Operation and Maintenance (O&M) area, in general.

1.1.1 Licenses

To enable the functionality in ST AS, the Capacity license must be installed.

For more information about the ST AS license, refer to *MTAS Licenses*.

1.1.2 Documents

Before starting any of the procedures in this document, the following documents must be available:

- *Ericsson Command-Line Interface User Guide*
- *Managed Object Model (MOM)*

1.1.3 Conditions

The following condition must apply:

- An Ericsson Command-Line Interface (ECLI) session in Exec mode is in progress.





2 Overview

This document describes SIP Trunking Connection and Control services offered by ST AS.

ST AS provides interconnection between the IMS network and enterprises PBXs in static or in dynamic mode.

Static mode implies that the PBX does not need to register with IMS. Instead, the PBX has a peering connection with the IMS network and the routing paths are defined in provisioning data.

Dynamic mode connectivity implies that the IP-PBX registers each of its access routes with the IMS network using IMS registration procedure.

2.1 Subfunctions

This section describes the subfunctions provided by the basic SIP Trunking functions, that is, the connection and control services in the ST AS.

2.1.1 PBX Originating Call Handling

This section describes the handling of PBX originating calls.

2.1.1.1 Static Mode Connect

The PBX user can originate calls when the PBX is configured in the IMS network using the static mode PBX connect.

The HSS provisioning is used to select the two following alternatives for routing to the ST AS:

- **Ma mode**

The user is provisioned with the capabilities of the ST AS. The list of capabilities is then used by Interrogating Call Session Control Function (I-CSCF) to perform a distinct Public Service Identity (PSI) routing to the ST AS.

- **ISC mode**

The user is provisioned with the CSCF address serving the user. The address is used by the I-CSCF to route the request to the Serving CSCF (S-CSCF), where Initial Filter Criteria (iFC) is used to trigger the ST AS. This alternative allows the multiple ASs to be used, a so called AS chaining.



For PBX originating calls, the Interconnection Border Control Function (IBCF) authenticates the PBX, inserts the PBX identity and `orig` parameter into the added P-Served-User header field, and signals this to the I-CSCF. The I-CSCF performs a location request to HSS. Depending on the HSS provisioning, the HSS returns the S-CSCF address (ICS mode) or the ST AS capabilities (Ma mode).

If the Ma mode is returned, the I-CSCF uses the received list of capabilities to match to a ST AS name to be used for PSI direct routing.

If the ISC mode is returned, the I-CSCF sends the request to S-CSCF. The S-CSCF executes originating iFCs, and sends the request to the ST AS over ISC.

ST AS can (depending on configuration) validate the calling user identity. ST AS executes the originating services, and thereafter, the ST AS forwards the request to the Transit Function (Ma mode), see Section 3.2 on page 11, or routes the call-back to the S-CSCF (ISC mode) that checks if additional ASs are to be started, and then routes the request to the terminating network.

ST AS performs feature tag processing as described in Section 2.1.6 Feature Tag Processing on page 7, and media policing of concurrent media lines as described in Section 2.1.5 Media Policing of Concurrent Media Lines on page 7.

ST AS uses preconfigured Transit Function Address to forward the call request, see Section 3.2 on page 11.

2.1.1.2 Dynamic Mode Connect

The PBX user can originate calls when the PBX is configured in the IMS network using the dynamic mode PBX connect.

The call is originating from the PBX user using one of the previously registered PBX routes. The Proxy CSCF (P-CSCF) searches for a matching Wildcarded Public User Identity (wIMPU) for the served user, and adds the matched wIMPU into a P-Profile-Key header. The P-CSCF follows the service route stored at registration to Serving CSCF (S-CSCF). The S-CSCF executes originating Initial Filter Criteria (IFCs) sending the request to the ST AS over the IMS Service Control (ISC) interface.

ST AS performs the feature tag processing as described in Section 2.1.6 Feature Tag Processing on page 7, and media policing of concurrent media lines as described in Section 2.1.5 Media Policing of Concurrent Media Lines on page 7.

The ST AS follows the route inserted by S-CSCF back to the S-CSCF that routes the call towards the terminating network.



2.1.2 PBX Terminating Call

The PBX user can receive calls when the PBX is configured in the IMS network using the Static mode PBX connect.

For static mode, the call request is received in the terminating I-CSCF that makes a location query to HSS. All number series served by the PBX must be provisioned in the IMS Common Name (CN) as wildcarded PSIs.

After successful wildcard match, the HSS, depending on provisioning, returns the S-CSCF address (ISC mode), or a list of capabilities that I-CSCF match to a direct route (Ma mode) to the ST AS. I-CSCF populates received wildcarded identity in the PPK header and forwards the call request to S-CSCF or ST AS depending on ISC or Ma mode. In case of ISC mode, the S-CSCF executes terminating iFCs, and sends the request to the ST AS over ISC.

For dynamic mode connect, the PBX user can receive calls when the PBX has active PBX routes registered. The PBX terminating calls reaches ST AS via S-CSCF over the ISC interface that uses the wIMPU to trigger an ICF for the terminating registered case.

ST AS performs the feature tag processing as described in Section 2.1.6 Feature Tag Processing on page 7, and media policing of concurrent media lines as described in Section 2.1.5 Media Policing of Concurrent Media Lines on page 7.

The ST AS performs the route selection as described in Section 2.1.3 ST AS Route Handling Static Mode on page 5.

If the route selection is successful, the ST AS uses the selected route to forward the call request.

For static mode (Ma mode), the ST AS inserts the selected route in the Route header of the call request, and the call is routed to the PBX through IBCF (Ma mode), or routes the call-back to the S-CSCF (ISC mode) that routes to the PBX through IBCF.

For dynamic mode, an Accept-Contact header is added to the `INVITE` with required and explicit user preferences saved from the route registration. ST AS follows the route inserted by the S-CSCF, and sends the `INVITE` back to the S-CSCF that terminates the call through P-CSCF.

2.1.3 ST AS Route Handling Static Mode

A PBX can be connected to IMS with multiple PBX routes simultaneously for increased trunking capacity and robustness. For PBX terminating calls, the ST AS uses random selection among all active routes. If no active route is available, a standby route is used, if defined.

For handling the connection errors, ST AS supports configurable profiles. The profile defines the fault codes to be classified as connection errors.

A connection error implies that ST AS puts the route into `error_guard` state and cancels the outstanding `INVITE` request, if necessary. Reselection of another route is not attempted.

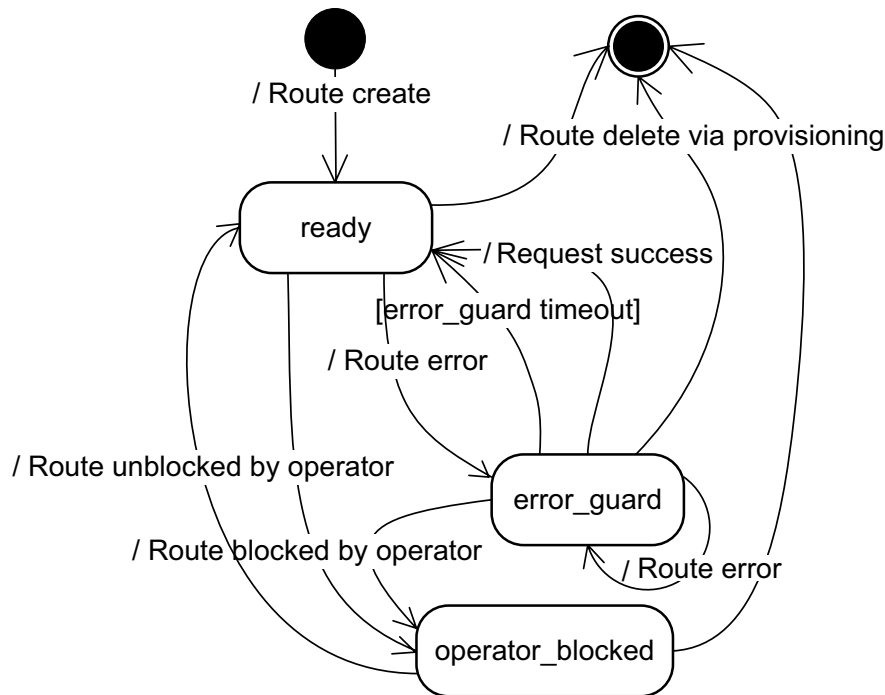


Figure 1 Route Selection State Model

The state machine for a PBX route handling is described in Figure 1. The state machine consists of a `ready`, `operator_blocked`, and `error_guard` states.

For PBX route used in the static mode PBX connect, the route is in `ready` state after creation, which occurs when the PBX Service document of the PBX has been retrieved and definition of all routes has been determined from it.

The operator can set the route into `operator_blocked` state or move it back to `ready` state through provisioning.

Note: Move the route from `error_guard` state to `ready` state, the route must be first blocked and then unblocked by the operator.

When ST AS attempts to use a PBX route and receives a permanent error related specifically to the route, the route is moved to the `error_guard` state. The route moves back from `error_guard` to `ready` after expiry of an `error_guard_timer`. The error guard time-out is an operator configurable node parameter. ST AS also attempts to use a route in `error_guard` state, if that is the only route option it finds. A successful request routed on a route in `error_guard` state results in the route state being moved to `ready` state.



2.1.4 Route Handling Dynamic Mode

For dynamic mode connect, a route is in `ready` state after the route has been registered. When selecting a route for a terminating call, a random selection is done among all routes in `ready` state.

2.1.5 Media Policing of Concurrent Media Lines

The media policing of concurrent media streams is triggered whenever a SDP-offer is received from any of the involved users during establishment of a PBX originating and PBX Terminating call, or after the call has been established when a session update is performed. If the number of active media lines in the SDP-offer is more than 10, the call or session attempt is rejected.

2.1.6 Feature Tag Processing

Table 1 describes the processing of feature tags in the Accept-Contact header. Where “Known” implies a feature tag configured in, `mtasStPrimaryFeatureTag` and `mtasStSecondaryFeatureTags`, and “Unknown” implies a feature tag not configured in, `mtasStPrimaryFeatureTag` and `mtasStSecondaryFeatureTags`.

The value `urn:urn-7:3gpp-service.ims.icsi.mmtel` is configured in the `mtasStPrimaryFeatureTag` attribute by default.

With the extended feature tag function, it is possible to tag the primary feature tag with a specified value defined in the attribute `mtasStExtendedStringFeatureTag`, for example, `audio;explicit;require`. The function extended feature tag is configured in `mtasStExtendedFeatureTag` and `mtasStExtendedStringFeatureTag` Managed Object (MO) attributes.

If the `mtasStExtendedFeatureTag` attribute is set to `true`, and the received INVITE does not contain an Accept-Contact header. Then, a new Accept-Contact header entry is added which contains the primary feature tag as defined by `mtasStPrimaryFeatureTag`, and extended by the feature tag extension as defined by `mtasMmtExtendedStringFeatureTag`.

*Table 1 Accept-Contact Header Actions Taken in MTAS*

Accept-Contact Headers	Action Taken
"Known" feature tag present in one of the Accept-Contact headers in the initial incoming INVITE.	<p>The session is accepted as an ST AS session if any of the known feature tags are present in the Accept-Contact.</p> <p>If the Accept-Contact header in the initial incoming INVITE does not contain the <code>mtasStPrimaryFeatureTag</code> but contains <code>mtasStSecondaryFeatureTags</code>, then the <code>mtasStPrimaryFeatureTag</code> is added. If there is a feature tag value defined in the <code>mtasStPrimaryFeatureTag</code> of urn-type, for example, <code>urn:urn-7:3gpp-service.ims.icsi.mmtel</code>, the following feature tag is added, unless <code>+g.3gpp.icsi-ref</code> is present in the <code>mtasStSecondaryFeatureTags</code>.</p> <p>The feature tag <code>+g.3gpp.icsi-ref="urn:urn-7:3gpp-service.ims.icsi.mmtel"</code> is also copied into the Contact header.</p>



Accept-Contact Headers	Action Taken
<p>“Unknown” feature tag present along with a recognized MMTel feature tag in one of the Accept-Contact headers in the initial incoming INVITE.</p>	<p>The session is accepted as an ST AS session if any of the known feature tags are present in the Accept-Contact.</p> <p>If the Accept-Contact header in the initial incoming INVITE does not contain the <code>mtasStPrimaryFeatureTag</code> but contains <code>mtasStSecondaryFeatureTags</code>, then the <code>mtasStPrimaryFeatureTag</code> is added. If there is a feature tag value defined in the <code>mtasStPrimaryFeatureTag</code> of urn-type, for example, <code>urn:urn-7:3gpp-service.ims.icsi.mmtel</code>, the following feature tag is added unless <code>+g.3gpp.icsi-ref</code> is present in the <code>mtasStSecondaryFeatureTags</code>.</p> <p>The feature tag <code>+g.3gpp.icsi-ref="urn:urn-7:3gpp-service.ims.icsi.mmtel"</code> is also copied into the Contact header.</p>
<p>“Unknown” feature tag present with no MMTel recognizable feature tag in any of the Accept-Contact headers in the initial incoming INVITE.</p>	<p>The session is rejected with a 503 if none of the configured feature tags in <code>mtasStSecondaryFeatureTags</code> or <code>mtasStPrimaryFeatureTag</code> matches any of the Accept-Contact headers.</p>



Accept-Contact Headers	Action Taken
No Accept-Contact headers present.	<ul style="list-style-type: none">• If there are no Accept-Contact headers present in the initial incoming INVITE, and if the <code>mtasStAllowNoFeatureTag</code> is true, then the session is accepted, and a feature tag is added to the Accept-Contact and contact headers, using the value defined in the <code>mtasStPrimaryFeatureTag</code>.• If there is a feature tag defined in <code>mtasStPrimaryFeatureTag</code> of urn-type, for example, <code>urn:urn-7:3gpp-service.ims.icsi.mmtel</code>, then the following two feature tags are added: <code>+g.3gpp.icsi-ref="urn:urn-7:3gpp-service.ims.icsi.mmtel"</code> The session is rejected with a 503 if <code>mtasStAllowNoFeatureTag</code> is configured false.
No feature tag present in any of the Accept-Contact headers in the initial incoming INVITE.	

2.1.7

O&M Operations

The O&M operations specific the SIP Trunking connection and control services in ST AS are as follows:

- Configuration of node level parameters.
- Management of alarm raised by ST AS.



3 Connection and Control Services in ST AS Configuration

The basic SIP Trunking functionality in ST AS is controlled by the *MtasSt* and *MtasStControlAccessProfile* Managed Objects (MOs).

In addition, a few attributes related to the SIP Trunking functionality in ST AS are controlled by the *MtasShIf* and *MtasSip* MOs.

An overview of the ST MO is shown in Figure 2.

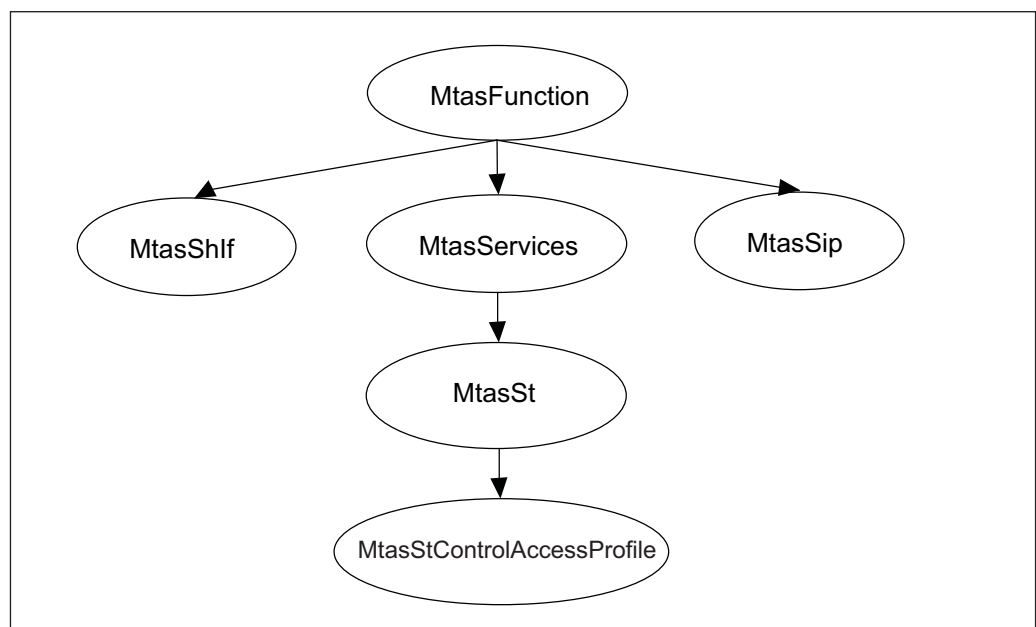


Figure 2 ST MO Structure

For MOs and attributes related to the ST AS, refer to *Managed Object Model (MOM)*.

3.1 ST AS Administrative State Configuration

The ST AS is enabled by setting the `mtasStAdministrativeState` attribute in the *MtasSt* MO to 1 (Unlocked). If the `mtasStAdministrativeState` is set to 0 (Locked), the ST AS is not provided by the MTAS.



3.2 Transit Function Address Configuration

For static mode connect over Ma interface, a PBX originating call is routed to the terminating side through the transit function in the home network. The address that can be a Fully Qualified Domain Name (FQDN) or IP address of the transit function is specified by the `mtasStTransitFunction` attribute.

3.3 SIP Trunking Charging Profile Reference

The `mtasStChargingProfileRef` attribute is the name of the charging profile that is used for the ST AS. A charging profile with this name must have been created before the name is set. If no name is specified, the charging profile with the name `default` is used. For more information about creation of charging profiles in MTAS, refer to *MTAS Charging Management Guide*.

3.4 Feature Tag Handling Configuration

The feature tag handling in ST AS is described in Section 2.1.6 Feature Tag Processing on page 7.

The `mtasStPrimaryFeatureTag` defines the primary SIP Trunking feature tag. The primary feature tag is added, if an ST session is detected.

The `mtasStSecondaryFeatureTags` attribute defines secondary feature tags used to recognize an ST AS session.

The `mtasStExtendedStringFeatureTag` attribute defines the string extension to add to the primary feature tag.

The `mtasStExtendedFeatureTag` attribute defines if the MTAS Extended ST feature tagging capability is to be used or not.

The `mtasStAllowNoFeatureTag` attribute defines if empty Accept-Contact headers without feature tags are allowed or not.

3.5 No Reply Timer Configuration

The `mtasStNoReplyTimer` attribute specifies the no reply timer for ST AS sessions. The timer is started when 180 Ringing is received. If the timer expires before a final response is received, the call is rejected.

3.6 Access Control Profile Configuration

The `MtasStControlAccessProfile` MO represents a set of properties describing certain behavior for handling of PBX routes.



3.6.1 PBX Route Error Definition

The `mtasStControlAccessProfileErrorCodes` attribute defines a list of final SIP response codes that is to be interpreted as an indication of a permanent error on a PBX route. If a PBX Terminating call is rejected with a response code that is included in the list, the route selected for the call is forced into the `Error Guard` state.

3.6.2 PBX Route Error Guard Timer

The `mtasStControlAccessProfileErrorGuard` attribute specifies the value of the Error Guard timer. The Error Guard timer is started when a route enters the `Error Guard` state and upon timer expiry, the route enters the `Ready` state.

3.6.3 PBX Route Out of Order Timer

The `mtasStControlAccessProfileTimeout` attribute specifies how long in milliseconds ST AS waits for a provisional or final response after sending an initial request towards a terminating PBX User, before it decides that a PBX route is out of order and forced into the `Error Guard` state. If the timer value is set to 0, the timer is not used.

3.6.4 ST SIP Port Configuration

It is possible to change the SIP port used for SIP Trunking Traffic in ST AS.

For configuration of the ST SIP port, refer to *MTAS SIP Management Guide*.





4 Fault Management

For alarms related to the Connection and Control services in ST AS, refer to *MTAS Alarm List*.