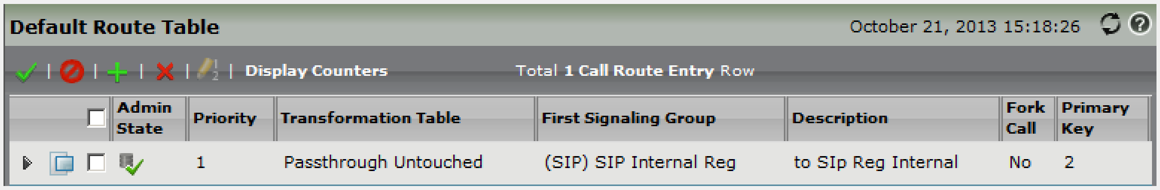
Creating and Modifying Entries to Call Routing Tables

To create or modify an Entry to a Call Routing Table:

1. In the WebUI, click the **Settings** tab.
2. In the left navigation pane, go to **Call Routing Table > Entry**.

Figure 1: Default Route Table


Default Route Table

October 21, 2013 15:18:26

Display Counters   Total 1 Call Route Entry Row

Admin State	Priority	Transformation Table	First Signaling Group	Description	Fork Call	Primary Key
	1	Passthrough Untouched	(SIP) SIP Internal Reg	to SIp Reg Internal	No	2

Modifying an Entry to a Call Routing Table

1. Click the **expand** (

) Icon next to the entry you wish to modify.
2. Edit the entry properties as required, see details below.

Resequencing an Entry in a Call Routing Table

1. Click the **Resequence** icon (

) at the top of the table.
2. Select the row(s) you want to move.
3. Click the **Move Selected Rows Up** (

) or **Move Selected Rows Down** (

) icon to reposition the row(s) in the table.
4. Click **Apply**.

Enabling an Entry in a Call Routing Table

1. Select the check box next to the entry you wish to enable.
2. Click the **Enable** (

) icon at the top of the table.

Disabling an Entry in a Call Routing Table

1. Select the check box next to the entry you wish to disable.
2. Click the **Disable** (

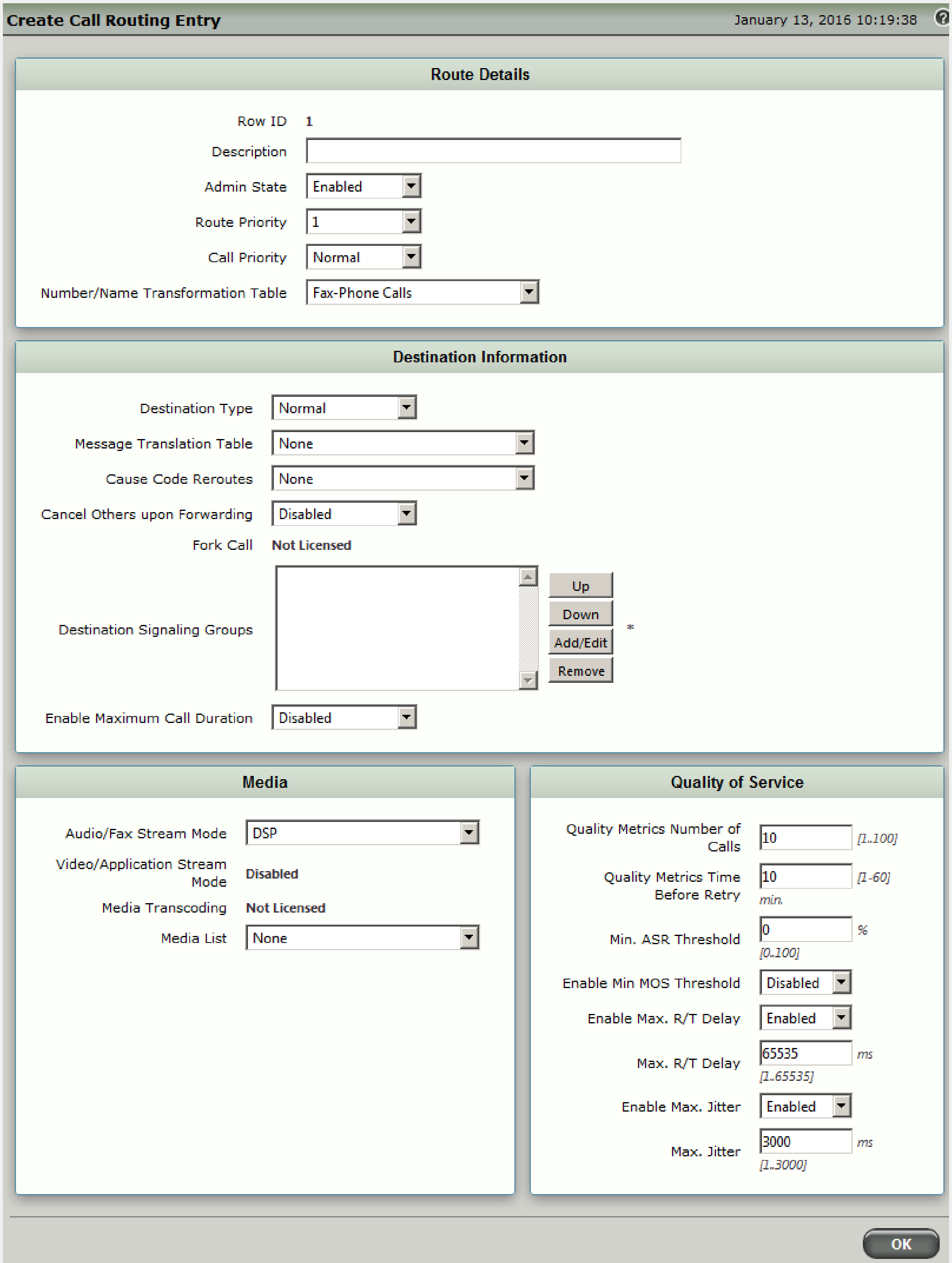
) icon at the top of the table.

Creating an Entry to a Call Routing Table

1. Click the **Create Routing Entry** (

) icon.

Figure 2: Create Call Routing Table Entry



Call Routing Entry - Field Definitions

Admin State

Specifies the admin state of the Call Route. Valid entry: **Enable** (enables the call route entry for routing the call, displays in configuration header as

) or **Disable** (disables the call route entry from being used, displays in the configuration header as

).

Route Priority

Priority of the route from 1 (highest) to 10 (lowest).

Higher priority routes are matched against before lower priority routes regardless of the order of the routes in the table.

Call Priority

Call Priority is used for emergency (911) calls in the U.S. and other countries. This field is used if the Priority header is **not** received in the ingress SIP INVITE and for PSTN interfaces. The default value is Normal. When integrating with E911 providers, Priority may have to be set to Emergency.

Number/Name Transformation Table

Specifies the Transformation Table to be used for this routing entry. This drop down list is populated from the entries in the Transformati on Table.

Destination Type

Specifies the destination type for calls using this route. Valid selections: **Normal**, **Registrar Table**, **Deny**, or **Trunk Group.**

Normal. Call routes to normal types such as ISDN or SIP signaling groups. Specify a list of signaling groups through Destination Signaling Groups.

Registrar Table. Call routes to a signaling group that contains the registrar table.

Deny. Call routes to a specific Q.850 cause code are rejected. Through the Deny Q.850 Cause Code field, select the specific Q.850 Cause code. When **Deny** is selected, the Deny Q.850 Cause Code field is displayed.

Trunk Group. Calls are routed to an incoming trunk group destination using the associated signaling group. This routing entry should be selected in order to route calls to a trunk group. When **Trunk Group** is selected, **Fork Call** and **Destination Signaling Groups** options are not available.

Deny Q.850 Cause Code

Specifies the Q.850 Cause Code for which the call is rejected. Select from the drop down list. This field is displayed only when **Deny** is selected as Destination Type.

Message Translation Table

Specifies which Message Translation Table to use.

Cause Code Reroutes

Specifies which Cause Code Reroute table to use.

Cancel Others upon Forwarding

Specifies whether or not forked calls should clear when one of the forked calls is forwarded.

Fork Call

Specifies whether or not to fork a call if this route is selected

Destination Signaling Groups

Specifies the Signaling Groups used as the destination of calls.

The first operational Signaling Group from the list will be chosen to place the call.

Helpful Hint
This field presents a multi-select widget when the **Add/Edit**/**Remove** button is clicked.
Click here for more information about using the Multi-select widget.

Enable Maximum Call Duration

Enables a maximum time (set in the **Maximum Call Duration** field) in which a call can stay in the connected state. This field provides another safeguard against stuck calls (i.e., if a call is up for two days and most likely not meant to stay connected).

Valid entry: **Enabled** (any call using that particular call route will disconnect after the configured duration in the connected state) or **Disabled** (default).

Maximum Call Duration

The maximum time (in minutes) in which a call can stay in the connected state. This field is available only when **Enable Maximum Call Duration** is set to **Enabled**. Valid entry: **1** - **10080** minutes.

Audio/Fax Stream Mode

Quick Start: How to Choose Media Mode

- To use DSP resources, (due to transcoding, Lync interoperability, etc.) select **DSP** mode.
- For SIP-to-SIP calls using Proxy, select **Proxy** mode.
- To use the same call routes where the inbound call route could be both TDM/SIP, select **Proxy preferred over DSP.**
- To use DSP resources, rather than proxy or switch media stream between endpoints, use **DSP preferred over Proxy.**

Media Mode enables you to choose how the SBC handles call signaling and Audio/Fax media stream during a call. Five options are available:

DSP (default entry). The SBC uses DSP resources for media handling (transcoding) but it does not facilitate the capabilities/features between endpoints that are not supported within the SBC (codec/capability mismatch). When **DSP** is configured, the Signaling Groups enabled to support **DSP** are attempted in order. To enable a Signaling Group for DSP mode, see Creating and Modifying SIP Signaling Groups.

Proxy. The SBC will proxy or switch the media stream between endpoints and let the endpoints negotiate common media capabilities and handle unsupported/unknown audio codecs. The media flows through the SBC without using DSP resources. For example, this mode is used when both inbound and outbound calls are SIP and both Signaling Groups are enabled for Proxy mode. If one Signaling Group has Proxy mode disabled or one of the inbound Signaling Groups is TDM, the call is rejected. Route configuration does not allow Destination Signaling Groups of TDM type if Stream Mode is set to Proxy. The failover mechanism can then be used to re-route to another Signaling Group. **We strongly recommend using Proxy preferred over DSP mode** since it allows the same routing entry with different Signaling Groups.

DSP preferred over Proxy. The SBC prefers to use DSP resources, rather than proxy or switch the media stream between endpoints. This mode is chosen during the call setup based on the call routing, and used for call routes where the inbound call route is both TDM and/or SIP (this avoids duplicate call routes). If the inbound Signaling Group is a SIP Signaling Group not enabled for DSP, then **Proxy** i s used.

Proxy preferred over DSP. The SBC prefers to proxy the media stream between endpoints rather than choosing DSP mode (which uses the DSP resource). This mode is chosen during the call setup based on the call routing result, and used for call routes where the inbound call route is both TDM and/or SIP (this avoids duplicate call routes). If the inbound or outbound Signaling Group is NOT in Proxy mode (either SIP signaling group is not enabled for Proxy mode or Signaling Group is TDM), then DSP is used.

Direct: The SBC media flows directly between compatible end-points without using either the SBC processor or a DSP resource.

Disabled. The Audio/Fax media selection process is disabled. When **Disabled** is selected, the SBC will reject incoming Audio/Fax streams.

Video/Application Stream Mode

Media Mode enables you to choose how the SBC handles call signaling and Video/Application media stream during a call.

Proxy. The SBC will proxy the media stream between endpoints. The media flows through the SBC without requiring DSP resources. For example, this mode is used when an inbound SIP call includes Video/Application streams and both inbound and outbound SIP Signaling Groups are enabled for Proxy mode. If one Signaling Group has Proxy mode disabled, the stream is disabled. If the outbound Signaling Group is TDM, the call will be connected using Audio/Fax media only and Video/Application media stream will be rejected.

Direct: The SBC media flows directly between compatible end-points without using either the SBC processor or a DSP resource.

Disabled. The Video/Application Stream Mode is disabled. The SBC will **not** proxy or switch media stream between endpoints.

Media Transcoding

Specifies whether or not to use media transcoding.

Media Transcoding requires a specific Transcoding License. Do not enable Media Transcoding unless your calling configuration requires it and SBC Edge is licensed for the Transcoding feature.

- If Media Transcoding is enabled without a Transcoding license, a critical alarm is generated regardless of whether or not the routed calls require transcoding.

- Transcoding must be enabled on SIP-to-SIP calls.

Media List

Specifies the Media List to use for this call route. This drop down list is populated with the Media List entries created through the Create Media list option. See Creating and Modifying Media Lists.

If the Media List configuration selected, then the Destination Signaling Group would be selected that has the common media set available. The media order from the call route's media list takes precedence over the Signaling Group's media list. Generally this field should be kept the default value "None" unless the media codec selection has to be controlled and manipulated for this route.

Quality Metrics Number of Calls

Specifies the number of calls over which the quality metrics are calculated.

Quality Metrics Time Before Retry

Specifies the period of time in minutes after which a route is tried again after failing quality metrics.

Min. ASR Threshold

Specifies the minimum answer/seizure ratio for this rule to be considered for use.

Click here to read more about Answer Seizure Ratio.

The ASR is a measure of network quality defined by the ITU. The answer/seizure ratio (ASR) is a measurement of network quality and call success rate in telecommunications. It is the percentage of answered telephone calls with respect to the total call volume.

The answer/seizure ratio is defined as 100 times the ratio of sucessfully answered calls divided by the total number of call attempts (seizures).

Busy signals and other rejections by the called number count as call failures. This makes the ASR highly dependent on end-user action or behavior and is out of control by the telecommunications carrier. Low ASR values may be caused by far-end switch congestion, not answering by called parties and busy destination lines.

Enable Max. R/T Delay

Specifies whether or not to use Round Trip Delay

Click here to read more about Round Trip Delay.

In telecommunications, the round-trip delay time (RTD) or round-trip time (RTT) is the length of time it takes for a signal to be sent plus the length of time it takes for an acknowledgment of that signal to be received. This time delay therefore consists of the transmission times between the two points of a signal.

The RTT was originally estimated in TCP by: RTT = (? - Old_RTT) + ((1 ? ?) - New_Round_Trip_Sample
Where ? is constant weighting factor(0 ? ? < 1). Choosing a value ? close to 1 makes the weighted average immune to changes that last a short time (e.g., a single segment that encounters long delay). Choosing a value for ? close to 0 makes the weighted average respond to changes in delay very quickly.

This was improved by the Jacobson/Karels algorithm, which takes standard deviation into account as well. Once a new RTT is calculated, it is entered into the equation above to obtain an average RTT for that connection, and the procedure continues for every new calculation.

Max. R/T Delay

Specifies the maximum average round trip (R/T) delay for this rule to be considered for use.

Enable Max. Jitter

Specifies whether or not Jitter will be considered as a quality metric for this Call Route.

Click here to read more about Jitter metrics.

Jitter is the undesired deviation from true periodicity of an assumed periodic signal in electronics and telecommunications, often in relation to a reference clock source. Jitter may be observed in characteristics such as the frequency of successive pulses, the signal amplitude, or phase of periodic signals. Jitter is a significant, and usually undesired, factor in the design of almost all communications links (e.g., USB, PCI-e, SATA, OC-48).

Jitter can be quantified in the same terms as all time-varying signals, e.g., RMS, or peak-to-peak displacement. Also like other time-varying signals, jitter can be expressed in terms of spectral density (frequency content).

Jitter period is the interval between two times of maximum effect (or minimum effect) of a signal characteristic that varies regularly with time. Jitter frequency, the more commonly quoted figure, is its inverse. ITU-T G.810 classifies jitter frequencies below 10 Hz as wander and frequencies at or above 10 Hz as jitter.

Jitter may be caused by electromagnetic interference (EMI) and crosstalk with carriers of other signals. Jitter can cause a display monitor to flicker, affect the performance of processors in personal computers, introduce clicks or other undesired effects in audio signals, and loss of transmitted data between network devices. The amount of tolerable jitter depends on the affected application.

Max. Jitter

Specifies the maximum average jitter for this rule to be considered for use.