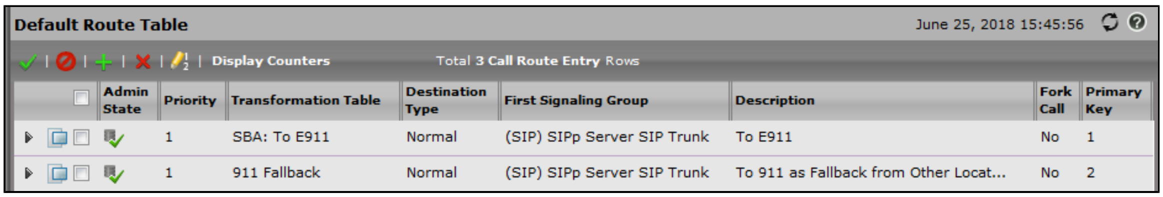
Viewing Call Routing Entry Counters

Route Entry counters display statistics about how Call Route Entries are performing and call quality.

To add 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 > Call Routing Table > Entry**.

Figure 1: Default Route Table



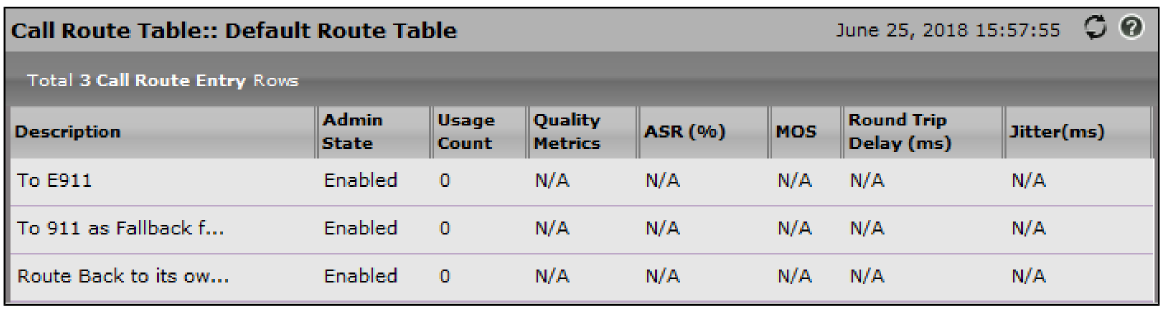
Default Route Table — June 25, 2018 15:45:56

Display Counters — Total 3 Call Route Entry Rows

Admin State	Priority	Transformation Table	Destination Type	First Signaling Group	Description	Fork Call	Primary Key
	1	SBA: To E911	Normal	(SIP) SIPp Server SIP Trunk	To E911	No	1
	1	911 Fallback	Normal	(SIP) SIPp Server SIP Trunk	To 911 as Fallback from Other Locat...	No	2

3. Click the **Display Counters** text at the top of the panel.

Figure 2: Display Counters



Call Route Table:: Default Route Table — June 25, 2018 15:57:55

Total 3 Call Route Entry Rows

Description	Admin State	Usage Count	Quality Metrics	ASR (%)	MOS	Round Trip Delay (ms)	Jitter(ms)
To E911	Enabled	0	N/A	N/A	N/A	N/A	N/A
To 911 as Fallback f...	Enabled	0	N/A	N/A	N/A	N/A	N/A
Route Back to its ow...	Enabled	0	N/A	N/A	N/A	N/A	N/A

Call Route Entry Counters - Field Definitions

Description

The description of the Call Route Entry as is appears in its parent Call Route Table.

Admin State

Indicates whether or not the entry is in an Enabled or Disabled State. For information about enabling and disabling entries, see Adding and Modifying Entries to Call Routing Tables.

Usage Count

Displays a count of how many times this route has been used.

Quality Metrics

Displays the average call quality for this call route.

ASR

Displays the Answer/Seizure Ratio for this call route.

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.

MOS

Displays the Mean Opinion Score for this call route.

The Mean Opinion Score (MOS) is used in telephony networks to obtain the human user's view of the quality of the network. In multimedia (audio, voice telephony, or video) especially when codecs are used to compress the bandwidth requirement (for example, of a digitized voice connection from the standard 64 kilobit/second PCM modulation), the mean opinion score (MOS) provides a numerical indication of the perceived quality from the users' perspective of received media after compression and/or transmission. The MOS is expressed as a single number in the range 1 to 5, where 1 is lowest perceived audio quality, and 5 is the highest perceived audio quality measurement.

MOS tests for voice are specified by ITU-T recommendation P.800.

The MOS is generated by averaging the results of a set of standard, subjective tests where a number of listeners rate the heard audio quality of test sentences read aloud by both male and female speakers over the communications medium being tested. A listener is required to give each sentence a rating using the following rating scheme:

MOS	Quality	Impairment
5	Excellent	Imperceptible
4	Good	Perceptible but not annoying
3	Fair	Slightly annoying
2	Poor	Annoying
1	Bad	Very annoying

The MOS is the arithmetic mean of all the individual scores, and can range from 1 (worst) to 5 (best).

Round Trip Delay

Ddisplays the average round trip delay for this call route.

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.

Jitter(ms)

Displays the average jitter for this call route.

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.