What are the different SIP Profile Parameters and their usage

Answer

SIP Trunk Profile Parameters:

- **DontFwdRefer Usage: DontFwdRefer=[0|1]**

When this parameter is set to 1, it inhibits the use of REFER for transfer on the trunk. It also inhibits sending INVITE with Replaces header. Peer must support INVITE without SDP for certain transfer call- flows

- **SendMacIn911CallSetup Usage: SendMacIn911CallSetup=[0|1]**

This parameter is used in conjunction with SIP based emergency gateways, such as those provided by 911 Enable. It appends the MAC address of the IP phone in the From tag of an outgoing emergency call.
From: "Dizzy Gillespie"
<sip:+14085551111@10.1.3.55:5060;user=phone>;tag=shorUA_1077733456- 103455277-EPID-001049042E4A

This only applies to ShoreTel IP Phones, excluding the IP-8000 conference room phone

- **StripVideoCodec Usage: StripVideoCodec=[0|1]**

This parameter should be set to 1 if the trunk does not support video properly. When set to 1, it strips video codecs from SDP in INVITE's being sent to the trunk and properly restores and rejects the video media lines in the 200 response from the trunk. It also strips video codecs from INVITE's coming from the trunk and properly restores and rejects the video media lines in the 200 response to the trunk

- **AddG729AnnexB_NO Usage: AddG729AnnexB_NO=[0|1]**

This parameter should be set to 1 if the trunk does not support G729 Annex B properly. When this is set, any outgoing INVITE with G729 in the SDP will have the attribute "a=fmtp:18 annexb=no" added to the SDP.

- **HistoryInfo Usage: HistoryInfo=[none|diversion|history]**

This parameter controls how information is presented when an external incoming call is forwarded out this trunk. In this case, the "From" header will indicate the actual caller, which may not be a valid number to present to the trunk. The Diversion or History-Info header will be used to indicate the DID number of the user on who's behalf the call was forwarded.

If set to 'none' or omitted, then no indication of the forwarding number is presented. If set to 'diversion', the SIP Diversion header is supplied, as dictated by RFC 5806. If set to 'history', the SIP History-Info header is supplied, as dictated by RFC 4244.

- **EnableP-AssertedIdentity Usage: EnableP-AssertedIdentity=[0|1]**

This profile parameter controls how Caller-ID is presented on outbound calls. If it is set to 0 or not pre- sent, then the old style or presenting caller-ID in From header is used when sending outgoing calls. Note that the style of presenting blocked caller-ID has changed in ShoreTel 12.

When set to 1, the Caller-ID is placed in the P-Asserted-Identity header. If privacy is indicated for the call (User dials *67, or trunk group is configured to not send Caller-ID), then a Privacy header is inserted with value “id”, and the From header is anonymous

- **Port Usage: Port=[5060|1-65535]**

This profile parameter changes the remote port used for the SIP trunk. Currently, there is no way to con- figure the port number for SIP trunks in ShoreWare Director. Only port 5060 is supported. This profile parameter allows the port number for a trunk group to be configured

- **HairPin Usage: HairPin=[0|1]**

This profile parameter controls if hairpin is allowed on SIP trunk calls, when enabled and available, features like Barge-in, silent monitoring, whisper-page, whisper-coach, call-record will be supported on the SIP trunks.

- **OptionsPing Usage: OptionsPing=[0|1]**

This profile parameter controls if OPTIONS message should be sent to remote party for detecting connectivity

- **OptionsPeriod Usage: OptionsPeriod=[180|60-3600]**

This profile parameter is used to control the time interval between SIP OPTIONS messages

- **OverWriteFromUser Usage: OverWriteFromUser=[none|UserID|BTN]**

This profile parameter is used to choose either user’s id or billing phone number in the FROM header when making calls

- **DontAdvertiseUpdate Usage: DontAdvertiseUpdate=[0|1]**

This profile parameter is used to decide if UPDATE should be sent in the SUPPORTED header

- **RFC2543Hold Usage: RFC2543Hold=[0|1]**

This profile parameter is used to decide if connection field should be set to 0.0.0.0 in case of sending out- going INVITE for hold

- **AlwaysSend180 Usage: alwaysSend180=[0|1]**

This profile parameter is used to decide if a 180 will be sent out right away after receiving an incoming INVITE

- **IgnoreEarlyMedia Usage: IgnoreEarlyMedia=[0|1]**

This profile parameter is used to decide if early media should be forwarded to the caller, when a SIP de- vice doesn't wish to accept early media, this parameter should be set to be 1

- **Register Usage: Register=[0|1]**

This profile parameter is used to decide if outgoing REGISTER messages should be sent

- **RegisterUser Usage: RegisterUser=[BTN|UserID|DID]**

This profile parameter is used to decide in what to use in FROM header in the outgoing REGISTER messages

- **RegisterExpiration Usage: RegisterExpiration=[3600|60-86400]**

This profile parameter is used to decide the time interval between outgoing REGISTER messages

- **1CodecAnswer Usage: 1CodecAnswer=[0|1]**

This profile parameter is used to decide if the SDP should contain only 1 codec for an outgoing answer.

SIP Extension Profile Parameters:

- **1CodecAnswer Usage: 1CodecAnswer=[0|1]**

Some devices do not honor the codec order specified in a 200 OK response to an INVITE. This causes several problems. First, some endpoints in the system do not support asymmetric codecs during a session. Second, any bandwidth calculations based on observing the offer/answer exchange will likely be wrong. When set to 1, only 1 audio codec is sent in a 200 OK response.

- **AddGracePeriod Usage: AddGracePeriod=[0-1800]**

Some SIP devices re-register too close to the expiration time, introducing a race condition where the sys- tem is in the process of deleting the record from the system when the re-register is received. This parameter adds a grace period to the expiration received in the REGISTER request.

- **AllowedCodecs Usage: AllowedCodecs=[any|[codec[,codec]*]**

Valid values are 'any' (default) or a comma separated list of codec names. The codec name must be for- matted as shown on the Supported Codecs page (Administration, Call Control, Supported Codecs). For example: 'PCMU/8000'. This should be used if the SIP device cannot follow the normal rules of codec negotiation for all codecs supported in the installation. For example, one particular implementation would rejected requests containing some codecs it didn't understand. This only applies to audio codecs. Video codecs and RFC 2833 'telephony-event' is not affected by this parameter.

- **DelayUnregister Usage: DelayUnregister=[0-20]**

Some devices, under certain circumstances, un-register, then immediately register again. This introduces a race condition similar to the one discussed in section 0. Usage of this parameter mitigates this problem.

- **FakeDeclineAsRedirect Usage: FakeDeclineAsRedirect=[0|1|400-606]**

Some SIP devices present an option to decline a call. When invoked, various different response codes have been used by various implementations. If set to 0, only a 3xx class response will cause the call to be diverted to the busy destination. If set to 1, 603 will be sent to busy destination as well. If set to a value from 400 to 606, the selected response code will be used to send the call to the busy destination.

- **MWI Usage: MWI=[none|subscribe|notify]**

This parameter defines how RFC 3842 Message Waiting Indication is handled. When set to "subscribe", an explicit subscription is required. If set to "notify", the NOTIFY messages are sent without requiring a SUBSCRIBE. If set to "none", then MWI is not supported.

- **OptionsPing Usage: OptionsPing=[0|1]**

ShoreGear switches can send a periodic OPTIONS message to SIP devices, and mark them Out-Of- Service if they don't respond. There are 2 benefits to this: Calls are diverted immediately to the busy destination, and there is logging of the event on the server.
The OPTIONS ping occurs periodically between 3 and 4.5 minutes.

- **OptionsResponse Usage: OptionsResponse=[200-699]**

Some devices reject OPTIONS requests, such as with a 405 "Not Supported" response. This can still be used to determine if the device is alive and on the network by using this parameter. Otherwise, a 405 response would put the device Out-Of-Service.

- **SendEarlyMedia Usage: SendEarlyMedia=[0|1]**

When set to 1, the device will be sent 183 response with SDP for certain call-flows. Currently, this is only used in error conditions when an announcement is played.

- **StripVideoCodec Usage: StripVideoCodec=[0|1]**

This parameter should be set to 1 if the device does not support video properly. When set to 1, it strips video codecs from SDP in INVITE's being sent to the device and properly restores and rejects the video media lines in the 200 response from the device. It also strips video codecs from INVITE's coming from the device and properly restores and rejects the video media lines in the 200 response to the device.

- **XferFailureNotSupported Usage: XferFailureNotSupported=[0|1]**

For scalability reasons, there are a few call-flows that use REFER as a means for the caller to hear ringback tone. These call-flows rely on the device's capability to recover from a transfer failure and keep the original call alive. If the device cannot do this, then this parameter should be set to 1, and an alternative means of providing ringback is used.

ShoreTel Dates

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