How to solve conference issues related to the RTP packetization interval

This article describes how issues related to the RTP Packetization interval can be solved in the case of conference calls on VoipNow servers. The recommended solution is to make sure that all parties involved in that call have the packetization interval set to **20ms** during conference calls on VoipNow servers. Otherwise, when channels are bridged, voice breaks, missing or dropped syllables, unexplained clicks or noises may occur.

Overview

Voice breaks, missing or dropped syllables, unexplained clicks or noises are triggered by different values of the `ptime` parameter when channels are bridged. According to the RFC3551 for packetized audio, the default packetization interval must be 20 ms or one frame, whichever is longer, unless required otherwise. However, a receiver should be able to accept packages with a packetization interval between 0 and 200ms.

Good to know

If `autoframing=yes` is set in `sip.conf` within the global section, all calls will try to set the packetization based on the preferences of the remote end point.

When autoframing is enabled at global level, it applies to all regardless of the selected codec packetization.

If one of the parties involved in a call does not provide a `ptime` interval, the recommended value that should be used is **20ms**.

In VoipNow, autoframing is not enabled in `sip.conf`. Consequently, VoipNow will use default values for `ptime`, depending on the selected codec. For example, if a call is made using G711 as codec, the `ptime` will be 20ms. This means that each RTP packet sent during that call will only have **20ms** of audio recorded in it.

Ptime and different devices

Polycom IP 650/ Soundpoint 330,320

The `ptime` parameter for codecs can be set through provisioning by adding a specific value for each codec in the `Device Configuration File #3`:

```
voice.audioProfile.X.payloadSize="20"
```

The value for the payload size is in milliseconds: 10, 20, 30,...80;

X can be replaced by G711Mu, G711A, G722, G7221, and G729AB.

For example, when setting a `ptime=20` for the G711A codec, the following line should be added to the provisioning template:

```
voice.audioProfile.G711A.payloadSize="20"
```

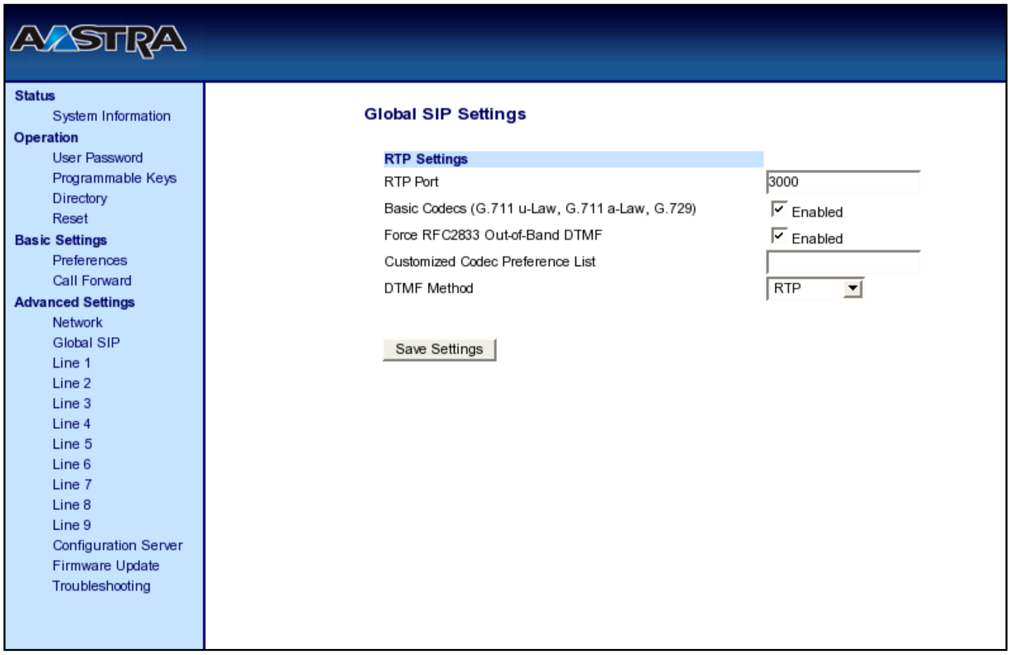
Aastra 480i

- To set the `ptime` parameter on a Aastra 480i codec, you need to go to the **Advanced Settings->Global SIP->RTP Settings** area;
- Enter a **Customized Codec Preference List** which allows you to use the codecs that you prefer for this IP phone.

For example:

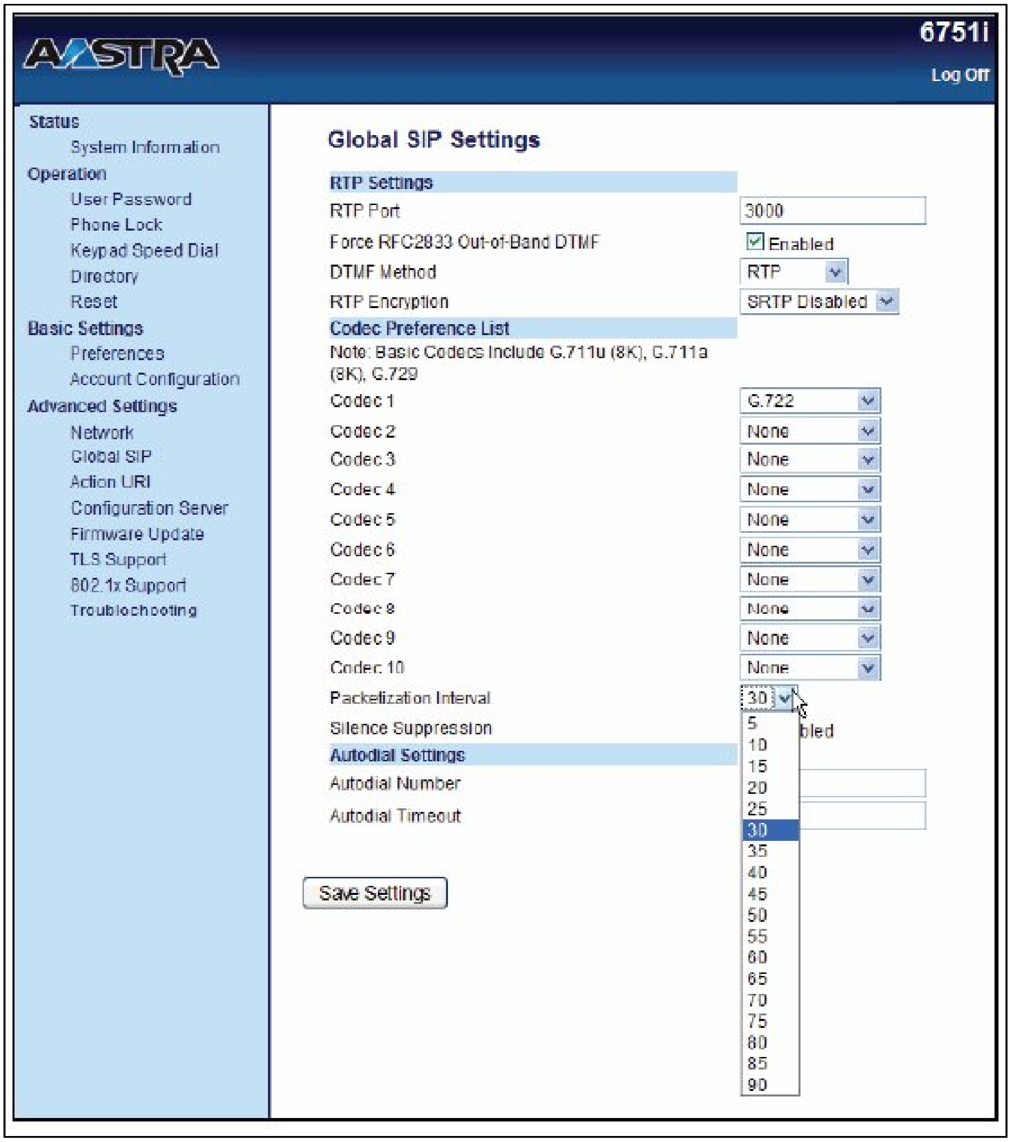
```
payload=8;ptime=20;
payload=0;ptime=20;
```

Please note that `payload 0` is for G.711 u-Law and `payload 8` is for G.711 a-Law.



Aastra 6751i

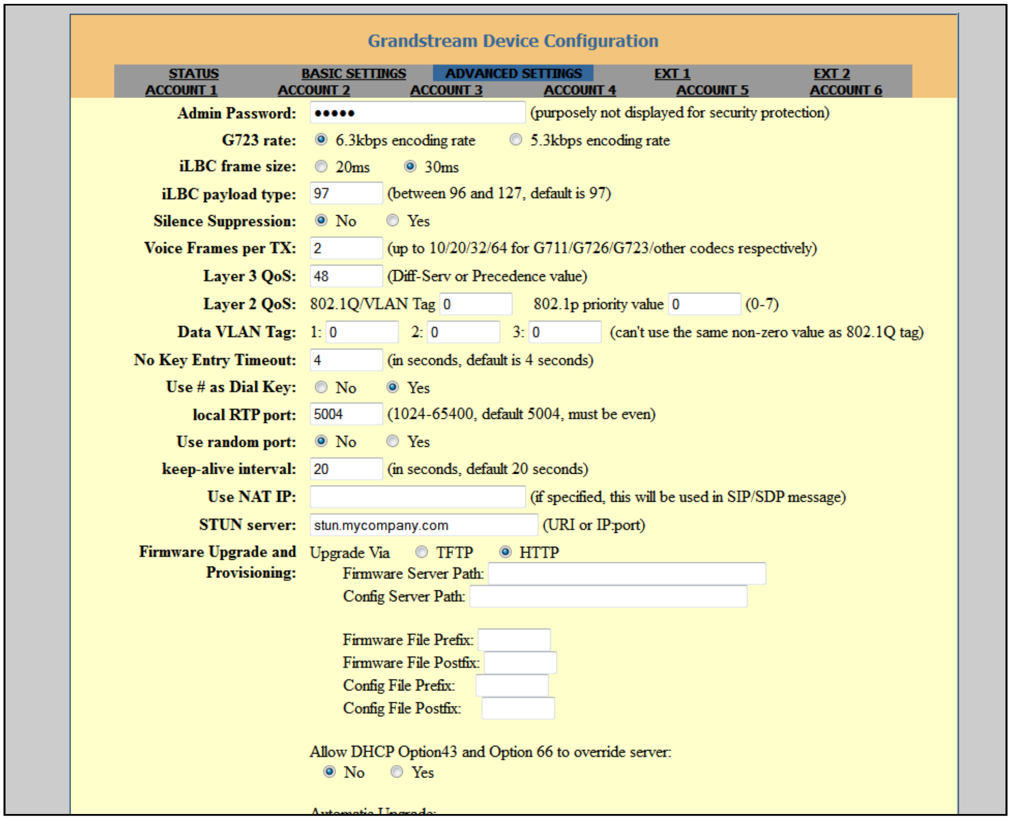
- To set the RTP packetization interval on Aastra 6751i, you need to go to the **Advanced Settings->Global SIP->Codec Preference List**;
- Valid values: 5 to 90, in increments of 5 milliseconds;
- To use this phone registered on a VoipNow server, you need to set the Packetization interval to **20ms;**



Grandstream Gxp2020

Web Interface

Go to **Advanced Settings** -> **iLBC frame size** and select a framesize of 30ms.



Provisioning Template

You need to set the `P97` variable to 1 as shown below:

```
# iLBC Frame Size. 0 - 20ms(default), 1 - 30ms.
P97 = 1
```

Yealink T20P

The `ptime` parameter can be set via provisioning templates.

In the device configuration file 1, you can set the ptime value as shown below:

```
[ account ]
path=/config/voip/sipAccount0.cfg
<yealink t20p configuration parameters>

ptime = 20

[ account ]
path=/config/voip/sipAccount1.cfg
<yealink t20p configuration parameters>

ptime = 20
```

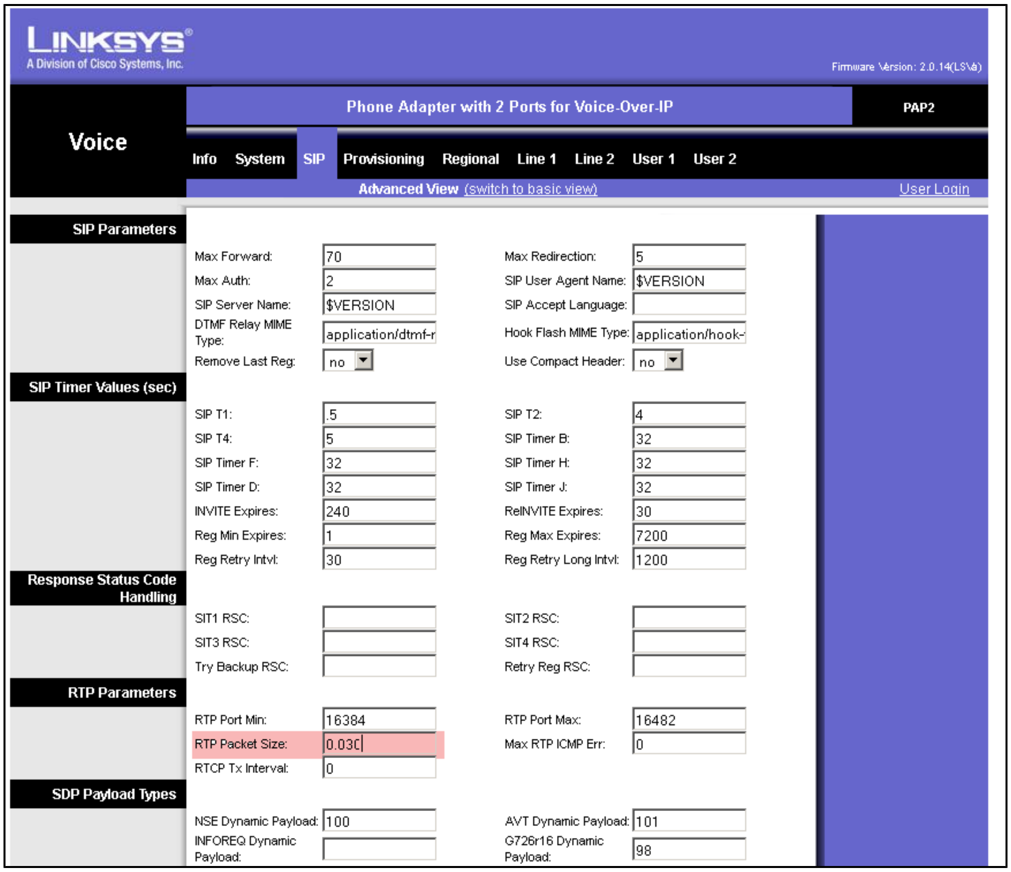
Linksys PAP2

Some Linksys devices, such as PAP2 or SPA2102, have the `RTP Packet Size` parameter set to 30ms by default.

In such cases, the `RTP Packet Size` parameter can be changed from the **SIP** tab of the web interface.

For most users, the 0.030 factory default preset should be replaced with 0.020.

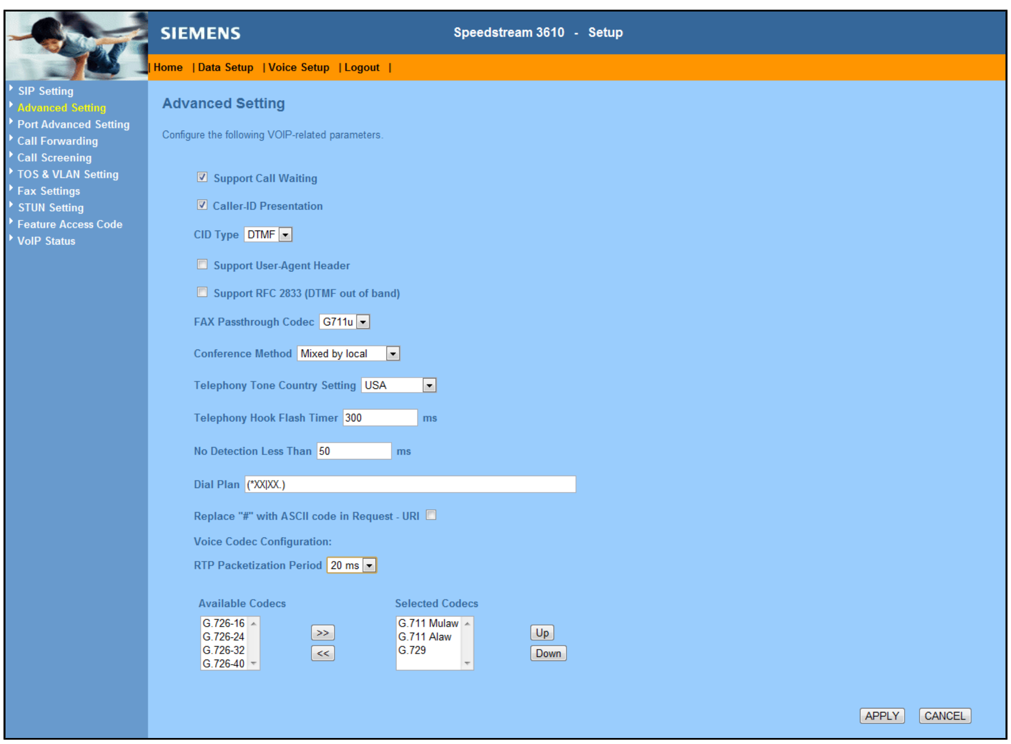
Please note that the `RTP Packet Size` parameter applies to all the lines served through that adapter.



Siemens Speedstream 3610

To set the RTP packetization interval on Siemens Speedstream 3610, you need to go to the **Voice Setup -> Advanced Setting**.

If you are using this device registered on a VoipNow server, please set the RTP Packetization period to **20ms**. Valid values: in increments of 10ms.



Resources

Asterisk Documentation

Related articles

- How to use the Conference features on a Phone terminal extension
- How to use the Conference feature in VoipNow
- How to solve audio issues caused by Asterisk timing configuration in VoipNow
- How to solve conference issues related to the RTP packetization interval