

# 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

Status

System Information

Operation

User Password

Programmable Keys

Directory

Reset

Basic Settings

Preferences

Call Forward

Advanced Settings

Network

Global SIP

Line 1

Line 2

Line 3

Line 4

Line 5

Line 6

Line 7

Line 8

Line 9

Configuration Server

Firmware Update

Troubleshooting

Global SIP Settings

RTP Settings

RTP Port

3000

Basic Codecs (G.711 u-Law, G.711 a-Law, G.729)

☒ Enabled

Force RFC2833 Out-of-Band DTMF

☒ Enabled

Customized Codec Preference List

DTMF Method

RTP

Save Settings

## 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**;

ASTRA

6751i

Log Off

Status

System Information

Operation

User Password

Phone Lock

Keypad Speed Dial

Directory

Reset

Basic Settings

Preferences

Account Configuration

Advanced Settings

Network

Global SIP

Action URI

Configuration Server

Firmware Update

TLS Support

802.1x Support

Troubleshooting

## Global SIP Settings

RTP Settings

RTP Port

Force RFC2833 Out-of-Band DTMF ☒ Enabled

DTMF Method

RTP Encryption

Codec Preference List

Note: Basic Codecs Include G.711u (8K), G.711a (8K), G.729

Codec 1	<input type="text" value="G.722"/>
Codec 2	<input type="text" value="None"/>
Codec 3	<input type="text" value="None"/>
Codec 4	<input type="text" value="None"/>
Codec 5	<input type="text" value="None"/>
Codec 6	<input type="text" value="None"/>
Codec 7	<input type="text" value="None"/>
Codec 8	<input type="text" value="None"/>
Codec 9	<input type="text" value="None"/>
Codec 10	<input type="text" value="None"/>

Packetization Interval

Silence Suppression

Autodial Settings

Autodial Number

Autodial Timeout

## Grandstream Gxp2020

### Web Interface

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

Grandstream Device Configuration					
STATUS ACCOUNT 1	BASIC SETTINGS ACCOUNT 2	ADVANCED SETTINGS ACCOUNT 3	EXT.1 ACCOUNT 4	EXT.2 ACCOUNT 5	EXT.2 ACCOUNT 6
Admin Password: <input type="password"/> (purposely not displayed for security protection)					
G723 rate: <input checked="" type="radio"/> 6.3kbps encoding rate <input type="radio"/> 5.3kbps encoding rate					
iLBC frame size: <input type="radio"/> 20ms <input checked="" type="radio"/> 30ms					
iLBC payload type: <input type="text" value="97"/> (between 96 and 127, default is 97)					
Silence Suppression: <input checked="" type="radio"/> No <input type="radio"/> Yes					
Voice Frames per TX: <input type="text" value="2"/> (up to 10/20/32/64 for G711/G726/G723/other codecs respectively)					
Layer 3 QoS: <input type="text" value="48"/> (Diff-Serv or Precedence value)					
Layer 2 QoS: 802.1Q/VLAN Tag <input type="text" value="0"/> 802.1p priority value <input type="text" value="0"/> (0-7)					
Data VLAN Tag: 1: <input type="text" value="0"/> 2: <input type="text" value="0"/> 3: <input type="text" value="0"/> (can't use the same non-zero value as 802.1Q tag)					
No Key Entry Timeout: <input type="text" value="4"/> (in seconds, default is 4 seconds)					
Use # as Dial Key: <input type="radio"/> No <input checked="" type="radio"/> Yes					
local RTP port: <input type="text" value="5004"/> (1024-65400, default 5004, must be even)					
Use random port: <input checked="" type="radio"/> No <input type="radio"/> Yes					
keep-alive interval: <input type="text" value="20"/> (in seconds, default 20 seconds)					
Use NAT IP: <input type="text"/> (if specified, this will be used in SIP/SDP message)					
STUN server: <input type="text" value="stun.mycompany.com"/> (URI or IP:port)					
Firmware Upgrade and Provisioning: Upgrade Via <input type="radio"/> TFTP <input checked="" type="radio"/> HTTP					
Firmware Server Path: <input type="text"/>					
Config Server Path: <input type="text"/>					
Firmware File Prefix: <input type="text"/>					
Firmware File Postfix: <input type="text"/>					
Config File Prefix: <input type="text"/>					
Config File Postfix: <input type="text"/>					
Allow DHCP Option43 and Option 66 to override server: <input checked="" type="radio"/> No <input type="radio"/> Yes					

## 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

Theptime parameter can be set via provisioning templates.

In the device configuration file 1, you can set theptime 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.

LINKSYS®

A Division of Cisco Systems, Inc.

Firmware Version: 2.0.14(LS14)

Voice

Phone Adapter with 2 Ports for Voice-Over-IP

PAP2

Info

System

SIP

Provisioning

Regional

Line 1

Line 2

User 1

User 2

Advanced View (switch to basic view)

User Login

SIP Parameters

Max Forward:

70

Max Auth:

2

SIP Server Name:

\$VERSION

DTMF Relay MIME Type:

application/dtmf-r

Remove Last Reg:

no

Max Redirection:

5

SIP User Agent Name:

\$VERSION

SIP Accept Language:

Hook Flash MIME Type:

application/hook-

Use Compact Header:

no

SIP Timer Values (sec)

SIP T1:

5

SIP T2:

4

SIP T4:

5

SIP Timer B:

32

SIP Timer F:

32

SIP Timer H:

32

SIP Timer D:

32

SIP Timer J:

32

INVITE Expires:

240

RelINVITE Expires:

30

Reg Min Expires:

1

Reg Max Expires:

7200

Reg Retry Intvl:

30

Reg Retry Long Intvl:

1200

Response Status Code Handling

SIT1 RSC:

SIT2 RSC:

SIT3 RSC:

SIT4 RSC:

Try Backup RSC:

Retry Reg RSC:

RTP Parameters

RTP Port Min:

16384

RTP Port Max:

16482

RTP Packet Size:

0.03C

Max RTP ICMP Err:

0

RTCP Tx Interval:

0

SDP Payload Types

NSE Dynamic Payload:

100

AVT Dynamic Payload:

101

INFOREQ Dynamic Payload:

G726r16 Dynamic Payload:

98

## 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.

SIEMENS

Speedstream 3610 - Setup

Home

Data Setup

Voice Setup

Logout

SIP Setting

Advanced Setting

Port Advanced Setting

Call Forwarding

Call Screening

TOS & VLAN Setting

Fax Settings

STUN Setting

Feature Access Code

VoIP Status

Advanced Setting

Configure the following VOIP-related parameters.

☒ Support Call Waiting

☒ Caller-ID Presentation

CID Type

DTMF

☐ Support User-Agent Header

☐ Support RFC 2833 (DTMF out of band)

FAX Passthrough Codec

G711u

Conference Method

Mixed by local

Telephony Tone Country Setting

USA

Telephony Hook Flash Timer

300

ms

No Detection Less Than

50

ms

Dial Plan

(XX0XX)

Replace '#' with ASCII code in Request - URI

Voice Codec Configuration:

RTP Packetization Period

20 ms

Available Codecs

G.726-16

G.726-24

G.726-32

G.726-40

Selected Codecs

G.711 Mulaw

G.711 Alaw

G.729

Up

Down

APPLY

CANCEL

## Resources

## 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](#)