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Overview

RTP issues can be difficult to diagnose without a healthy amount of packet captures, etc.

Devices with Issues

Sonus

Carrier Info

Carriers known to be using this device: L3, WilTel, Bandwidth.com, Broadvox, Global Crossing, XO, Qwest and T-Systems (a division of Deutsche Telekom).

- 19:40, 9 December 2008 (PST)
- Known to support exactly 729 with rfc2833 OR 711u with DTMF inband.
 - ◆ If you're using FreeSWITCH then you'll be stuck using 711u (PCMU) with DTMF inband. (**also rfc2833 look below how**)
- Dropped audio infrequently. Anthony discovered that they have a +500ms response time if you STOP streaming RTP data to the carrier.

RTP_Issues

Whatever hardware they use has a list of reasons to reset.

- Causes:
 - ◆ If your installation of FreeSWITCH want to change timestamp base and send them mark bit, they reset with 2 seconds of silence.
 - ◆ If you send 2833 on it's own timestamp base it resets.

Dropped Audio

If you are suffering from dropped audio, specifically regarding the first 2-3 seconds of a single audio stream then try the following:

In `conf/vars.xml`

```
<!-- This will continue a steady stream of RTP to the Sonus device and your dropped audio will ne  
<X-PRE-PROCESS cmd="set" data="send_silence_when_idle=400"/>
```

Take care when using this parameter, as it may suppress ringback tones by actually sending early media.

In `conf/sip_profiles/external.xml` *Make sure you are running a FreeSWITCH version newer than Dec. 9th, 2008*

```
<!--This will rewrite all new timestamps instead of passing them from the other leg.-->  
<param name="rtp-rewrite-timestamps" value="true"/>
```

DTMF Problems

If you are having DTMF problems and Sonus **is in your media path**, you should make sure you are using the latest version of FreeSWITCH. As of FreeSWITCH revision 10744, FreeSWITCH auto-detects Sonus end-points and applies a hack to "fix" the issue. The hack is described as follows (from `switch_types.h`):

Sonus wrongly expects that, when sending a multi-packet 2833 DTMF event, the sender should increment the RTP timestamp in each packet when, in reality, the sender should send the same exact timestamp and increment the duration field in the 2833 payload. This allows a reconstruction of the duration if any of the packets are lost.

```
final_duration - initial_timestamp = total_samples
```

However, if the duration value exceeds the space allocated (16 bits), the sender should increment the timestamp one unit and reset the duration to 0.

Always sending a duration of 0 with a new timestamp should be tolerated but is rarely intentional and is mistakenly done by many devices. The issue is that the Sonus expects everyone to do it this way instead of tolerating either way. Sonus will actually ignore every packet with the same timestamp before concluding if it's DTMF.

Note that the version of FreeSWITCH which has this patch auto-detects Sonus, so you don't need to configure anything.

If you are going through a provider who uses Sonus **only as an SBC and not in your media path**

RTP_Issues

FreeSWITCH may incorrectly identify your call as going through Sonus and actually corrupt your media stream's DTMF. You can override FreeSWITCH's auto-detection of end-user agents with a flag in your sip_profiles/ profiles. Just add:

```
<param name="auto-rtp-bugs" value="clear"/>
```

In some cases, you may have the reverse situation where Sonus or Cisco equipment sits in your media path, but FreeSWITCH can't auto-identify it. In that case, try these settings:

```
<param name="auto-rtp-bugs" value="CISCO_SKIP_MARK_BIT_2833"/>
```

if you are having trouble with Cisco equipment. For Sonus, try:

```
<param name="auto-rtp-bugs" value="SONUS_SEND_INVALID_TIMESTAMP_2833"/>
```



FreeSWITCH to Sonus testing paths

If your Call to a Sonus still won't work

For those of you who have tried the `auto-rtp-bugs` and still cannot get DTMF to work for G711 negotiated calls, there appears to be an issue with the Sonus using Fax treatment which drops the out of band DTMF packets that come from FreeSWITCH (among other platform). Some more in depth information for Sonus users (at least for 5.0 and 5.1 users, I can't speak for those who may have the latest and greatest code revisions which may fix this) is below.

In the GSX Navigator (Sonus Insight), go to Sonus Profiles Packet Service Profiles default. You may have some changes made to the Fax Treatment/Failure section. To resolve the DTMF you will have to change that to match the below examples.

An example of a base profile that will not work:

Audio Encodings:	G.711	DTMF	Modem	Fax
	Pkt Size/ Law/	Relay/	Treatment/	Treatment/

RTP_Issues

Priority	Codec	Code Rate	Send SID	Remove	Failure	Failure
3	G711SS	20	LAWFROMCIR	NONE	G711FALLBACK	G711FALLBACK
		0BPS	DISABLED	DISABLED	CONTINUE	CONTINUE
4	G711	20	LAWFROMCIR	NONE	G711FALLBACK	G711FALLBACK
		0BPS	DISABLED	DISABLED	CONTINUE	CONTINUE

What you would need to change it to, to fix the DTMF:

Audio Encodings:	G.711	DTMF	Modem	Fax		
Priority	Codec	Pkt Size/ Code Rate	Law/ Send SID	Relay/ Remove	Treatment/ Failure	Treatment/ Failure
3	G711SS	20	LAWFROMCIR	NONE	G711FALLBACK	FAXRELAYORG711FA
		0BPS	DISABLED	DISABLED	CONTINUE	CONTINUE
4	G711	20	LAWFROMCIR	NONE	G711FALLBACK	FAXRELAYORG711FA
		0BPS	DISABLED	DISABLED	CONTINUE	CONTINUE

The above profile will work if you are sending direct to the GSX using the dtg=TRUNKGROUP protocol. If you are routing your calls to the PSX, then you would make the appropriate changes to the Preferred Packet Service Profile that is attached to the trunk group that you are a) sending from or b) sending to.

It's a crazy world out there.

Other Options

In conf/sip_profiles/external.xml

```
<!-- This will use the async timer (this one you can also set to none if you don't want asynchron  
<param name="rtp-timer-name" value="soft"/>
```

```
<!--
```

```
RTP_BUG_SEND_LINEAR_TIMESTAMPS = (1 << 3),  
Our friends at Sonus get real mad when the timestamps are not in perfect sequence even  
With this flag, we will only increment the timestamp when write packets even if they ar
```

```
RTP_BUG_START_SEQ_AT_ZERO = (1 << 4),  
Our friends at Sonus also get real mad if the sequence number does not start at 0.  
Typically, we set this to a random starting value for your safety.  
This is a security risk you take upon yourself when you enable this flag.
```

```
RTP_BUG_NEVER_SEND_MARKER = (1 << 5),  
Our friends at Sonus are on a roll, They also get easily dumbfounded by marker bits.  
This flag will never send any. Sheesh....
```

```
RTP_BUG_IGNORE_DTMF_DURATION = (1 << 6),  
Guess Who? ... Yep, Sonus (and who knows who else) likes to interweave DTMF with the au  
2X as long as it should and sending an incorrect duration making the DTMF very delayed.  
This flag will treat every DTMF as if it were 50ms and queue it on receipt of the leadi
```

```
RTP_BUG_ACCEPT_ANY_PACKETS = (1 << 7),  
Oracle's Contact Center Anywhere (CCA) likes to use a single RTP socket to send all its  
This messes up our ability to auto adjust to NATTED RTP and causes us to ignore its aud  
This flag will allow compatibility with this dying product.
```

```
RTP_BUG_GEN_ONE_GEN_ALL = (1 << 8)  
Some RTP endpoints (and by some we mean *cough* _SONUS_!) do not like it when the times
```

RTP_Issues

So say you are generating a file that says "please wait for me to complete your call, o
Now you place and outbound call and you are bridging. Well, while you were playing the
But, now that you have a remote RTP stream, you'd rather send those timestamps as-is in
Ok, so this causes the audio to completely fade out despite the fact that we send the m
Sigh, This flag will tell FreeSWITCH that if it ever generates even one RTP packet itse
actual timestamps in the frames.

```
RTP_BUG_NEVER_CHANGE_SSRC_ON_MARKER = (1 << 9)
```

By default FS will change the SSRC when the marker is set and it detects a timestamp re
If this setting is enabled it will NOT do this (old behaviour).

```
-->
```

```
<param name="manual-rtp-bugs" value="SEND_LINEAR_TIMESTAMPS|START_SEQ_AT_ZERO|NEVER_SEND_MARKER|I
```

See also

For more information on these and other Sonus issues:

- http://www.submityoursip.com/wiki/Sonus_NBS
- <http://blog.krisk.org/2009/02/update-youve-been-waiting-for.html>
- <http://www.freeswitch.org/node/160>

Voiceoperatorpanel VOP

VOP-Softclients don't like changes in the RTP timestamp, e.g. when a REFER to the VOP-client occurs and the RTP-stream is replaced by the stream from a different call with a different timestamp. This leads to one-way-audio for the callee, the VOP Softphone doesn't process the incoming RTP-pakets with the "new" timestamp and the callee will not hear the caller.

Setting [rtp-rewrite-timestamps](#) to True in the affected SIP-profile eventually solves the problem.