Configuring and Troubleshooting Fax and Modem Calls in AOS

The following guide explains how to configure and troubleshoot fax/modem calls over a VoIP network using T1/E1/analog based voice trunks and users. There are numerous factors to consider when sending or receiving fax/modem calls over a VoIP network. Fortunately, some of these factors can be eliminated or mitigated by changing the configuration on the ADTRAN IP Business Gateway (IPBG).

Configuration

There are two different methods for provisioning T1/E1/analog based trunks and analog voice users to handle fax/modem calls appropriately. The first method is by having the unit dynamically provision the settings, based on its ability to detect fax/modem tones during a call. The other method is through static provisioning, where various settings such as the audio codec, echo cancellation mode, and jitter buffer settings are manually provisioned. The following sections cover each method in more detail.

Dynamic Provisioning

Dynamic provisioning is done through a feature called modem passthrough. Modem passthrough works by automatically adjusting how the call is setup when certain fax/modem tones are detected. This can be beneficial for users or trunks where voice calls are also present. **Table 1** below lists the fax/modem tones that modem passthrough will detect:

Table 1: Fax and Modem Tones

T.30 CNG	Series of interrupted 1100±38 Hz tone with 4000±600 samples on and 24000±3600 samples off.
V.21 Preamble	Periodic sequence of 1850±12 Hz / 1650±12 Hz / 1850±12 Hz tones having durations of 26 / 156 / 26 samples and repeating 38±6 times.
V.25 ANS **T.30 CED**	Uninterrupted 2100±15 Hz tone with 26400±5600 samples on.
V.25 ANS with Phase Reversal	ANS/CED tone with periodic phase reversals (180°±10°) at intervals of 3600±200 samples. At least two phase reversals are required to disable echo cancellation.
V.8 ANSam	Uninterrupted 2100±1 Hz tone with 20% amplitude modulation by a 15±0.1 Hz tone.
V.8 ANSam with Phase Reversal	ANSam tone with periodic phase reversals (180°±10°) at intervals of 3600±200 samples. At least two phase reversals are required to disable echo cancellation.

Fax machines and modems work by interpreting tones. When the tones being interpreted are compressed, there is a good chance they will not be interpreted correctly, which will typically result in failed fax/modem calls. For this reason, it is important to use an uncompressed audio codec, such as G.711 µ-Law. For countries that use a-Law companding, substitute G.711 a-Law for any G.711 µ-Law references throughout this document.

When any of the fax/modem tones from **Table 1** are detected, modem passthrough will change the codec being used to G.711 and disable the following:

ALC	Automatic Level Control is used to normalize the audio output for varying levels of input.
VAD	Voice Activity Detection is used to detect audio. If no audio is detected by VAD, comfort noise can be inserted.
PLC	Packet Loss Concealment can be used to minimize the effect packet loss has on audio being played out.

Echo cancellation is a means to detect and suppress echo on the audio coming into the unit's Digital Signal Processor (DSP). Echo cancellation is beneficial for voice calls, but it is not recommended for fax/modem calls, as it's conceivable that valid fax/modem tones could be altered or suppressed. Because of this, echo cancellation needs to be disabled. Echo cancellation is made up of the following two components:

Non-Linear Suppression (NLS)	NLS is used to minimize residual echo and not reduce near-end signals, regardless of the level of residual echo.
Non-Linear Processing (NLP)	NLP is used to replace low level residual echo with either comfort noise, if Comfort Noise Generation is enabled, or silence.

If a tone with phase reversal is detected, modem passthrough will disable echo cancellation completely. Otherwise the echo cancellation is changed to "data mode," which will disable NLP, but leave NLS on. Below is an example configuration with modem passthrough enabled on a voice trunk:

```
voice trunk T01 type isdn
  connect isdn-group 1
  modem-passthrough
```

Static Provisioning

Static provisioning is beneficial if a particular voice user is used for fax/modem calls only. This will ensure that all of the necessary fax/modem parameters are used for every call. It is not recommended that static provisioning be used on a voice trunk, because typically T1/E1/analog based voice trunks will be used for a mix of voice and fax/modem calls.

In AOS, codecs are applied to a voice user or trunk through the use of codec lists. For faxing, a separate codec list should be configured that defines a single codec, G.711 µ-Law. An example

configuration of static codec selection on a voice user is below. The codec list is given the name **FAX**.

```
voice codec-list FAX
  codec g711ulaw
!
voice user 1201
  connect fxs 0/1
  codec-group FAX
```

An example configuration that manually disables echo cancellation on a voice user is shown below:

```
voice codec-list FAX
  codec g711ulaw
!
voice user 1201
  connect fxs 0/1
  codec-group G711
  no echo-cancellation
```

The features below are disabled by default, and will not appear in the running configuration. If they are enabled on a voice trunk or voice user that will be sending fax/modem calls, it is recommended that they be manually disabled if modem passthrough is not configured.

ALC	Automatic Level Control is used to normalize the audio output for varying levels of input.
VAD	Voice Activity Detection is used to detect audio. If no audio is detected by VAD, comfort noise can be inserted.
PLC	Packet Loss Concealment can be used to minimize the effect packet loss has on audio being played out.

An example configuration with the above features disabled on a voice user is below:

```
voice codec-list G711
  codec g711ulaw
!
voice user 1201
  connect fxs 0/1
  codec-group G711
  no echo-cancellation
  no alc  *
  no vad  *
  no plc  *
```

Note: Since these options are default configuration settings, they will not show up in the running configuration.

FXS Port Settings

The electrical settings for the FXS ports can be modified to match the needs of the terminating equipment and the analog line conditions. There is no standard methodology that governs the adjustment of these settings, which are the transmit gain, receive gain, and line impedance. Some settings can be adjusted based upon the type of issue or the hardware specifications of the terminating equipment, but the process typically comes down to a trial and error approach.

The transmit gain on the FXS port will affect the gain on the audio being transmitted towards the DSP. Increasing the value makes the audio louder, while decreasing the value makes the audio softer. The following configuration example changes the transmit gain to 0 dB. The default value is -6 dB, and the range is from -12 dB to 6 dB in .1 dB increments.

```
interface fxs 0/1
  tx-gain 0
```

The receive gain on the FXS port will affect the gain on the audio being transmitted out the FXS port. Increasing the value makes the audio louder, while decreasing the value makes the audio softer. The following configuration example changes the receive gain to 0 dB. The default value is -3 dB, and the range is from -12 dB to 6 dB in .1 dB increments.

```
interface fxs 0/1
  rx-gain 0
```

The following configuration example changes the impedance to 600Ω, which is a common value to change it to when troubleshooting fax/modem issues. The default value is 600Ω + 2.16µF (600c).

```
interface fxs 0/1
  impedance 600r
```

Fax Machine Configuration

Adjusting the settings on the fax machines themselves can also help improve faxing consistency. The two primary settings that often need to be adjusted are the transmission baud rate and error correction mode (ECM). Lowering the transmission baud rate increases the chances that the fax data can be received and processed by the fax machine. ECM sometimes needs to be disabled to prevent the receiving fax machine from attempting to get the sending fax machine to retransmit data that was not received or could not be processed properly. The retransmission of this data will lead to longer fax call times and to possible communication errors if the corrected data is not received.

Troubleshooting

If fax/modem calls are still failing after following one of the provisioning methods above, further troubleshooting will need to be done. In the sections that follow, troubleshooting RTP delay,

codec verification, tone detection, and reINVITE glare will be explained.

RTP Delay

A common problem with fax/modem calls over an IP routed network is the delay the RTP traffic encounters through the network. Given how sensitive data transmissions are, excessive delay, jitter, lost, and out of order packets will cause major problems on fax/modem calls. Multiple page faxes are especially susceptible to RTP delay and jitter, which can result in not all pages being received.   Prioritization of this traffic through the use of Quality of Service (QoS) will need to be implemented on the IPBG and throughout the provider network to ensure RTP is handled correctly. If the provider's network does not provide the appropriate prioritization for the RTP traffic, the IPBG can be adjusted to attempt to compensate for some levels of delay within limits. However, it is not possible to mitigate the effect of lost and out of order packets.

The adjustment that can be made to attempt to mitigate delay and jitter is changing the jitter buffer settings on the voice trunk or voice user. The jitter buffer's primary goal is to reduce the effects of network delay and allow for the smooth playback of audio. It does this by adapting its size based upon the delay between the received RTP packets, or by statically defining the amount of audio that is buffered. By default, the jitter buffer adapts to the changes in delay with a nominal value of 50ms and a maximum value of 100ms. If the unit is configured for **modem-passthrough**, the jitter buffer will be set to a fixed buffer size of 50ms when a modem/fax call has been detected.

Before adjusting the jitter buffer settings, a baseline of the current network performance needs to be established. This can be done by examining the output of **show voice quality-stats** from privileged exec in the command line interface (CLI), provided that the adaptive jitter buffer settings are still configured. The output below is an example:

```
# show voice quality-stats
        Start                                       Out of  Discard   Delay
ID      Time     From      To         Length Codec  Order   Pkts*  Avg   Max
-----------------------------------------------------------------------------
1       10:30 AM 1201      2002        0:40   G729   0       0       60    70
2       11:37 AM 1201      2001        0:50   G729   0       0       61    70
3       12:01 PM 1201      2002        0:29   G729   0       0       65    70
4       1:06 PM  1201      2002        1:18   G729   0       0       60    60
5       2:04 PM  1201      2001        0:32   G729   0       0       50    50
6       2:07 PM  1201      2001        3:15   G729   0       0       50    50
7       2:10 PM  1201      2001        9:36   G729   0       0       52    60
8       2:20 PM  1201      2003        0:59   G729   0       0       63    70
```

Based on this output, the fixed fax/modem jitter buffer would need to be set to 70ms. The reason for this is that 70ms appears to be the maximum amount of delay that normal calls experience (as seen in the **Max Delay** column). The value could be raised to 80ms, but then the delay may be too great for the fax machine to overcome. 80ms is the longest amount of delay that a fax machine can reasonably be expected to handle. An example of setting the fax/modem jitter buffer for a value of 70ms on a voice user is below:

```
voice user 1201
```

```
connect fxs 0/1
codec-group G711
rtp packet-delay fax 70
no echo-cancellation
```

Codec Verification

It is important to make sure that the codec used during a fax/modem call is G.711 μ-Law. As previously mentioned, the codec can be statically set to G.711 μ-Law or modem passthrough can be used to re-negotate the call to G.711 μ-Law when one of the fax/modem listed in Table 1 is detected.

The first step in troubleshooting fax issues is to determine the codec that was used on the failed fax call. After making a test fax/modem call, check the output of the **show voice quality-stats.**

```
# show voice quality-stats
        Start                                      Out of  Discard   Delay
ID      Time    From       To        Length Codec  Order   Pkts*  Avg  Max
---------------------------------------------------------------------------
2       11:31 AM 1201       2001      0:37   G729   0       0      50   50
3       11:32 AM 1201       2002      0:14   G729   0       0      50   50
4       1:57 PM  1201       2002      3:29   G729   0       0      50   50
5       1:57 PM  1201       2001      0:18   G729   0       0      50   50
7       2:00 PM  1201       2001      2:31   G729   0       0      50   50
9       2:01 PM  1201       2002      1:14   G729   0       0      50   50
11      2:03 PM  1201       2001      9:26   G729   0       0      50   50
12      2:03 PM  1201       2003      1:00   G711u  0       0      50   50
```

If the call with an ID of 12 was the fax/modem call that was made, it was using the correct audio codec of G.711 μ-Law.

Another way to determine the codec used during the call is by using the debug output to analyze call control messaging. Before making a fax/modem call, issue the command **debug sip stack messages** from the privileged exec prompt. For brevity, we will not include debug of a full SIP call flow. The initial call setup consists of an INVITE being sent with an SDP offer. This SDP will contain the parameters that the calling party gateway is proposing for the RTP transaction. The called party gateway typically follows with a 100 Trying, a 180 Ringing or 183 Session Progress, and then a 200 OK once the called party picks up. The 200 OK will contain the SDP answer, which will finalize the initial codec selected. The calling party will send an ACK in response to the 200 OK.

At this point, the two fax/modems are attempting to negotiate the call parameters by sending tones over the RTP stream. If either the called or calling party gateway is listening for fax/modem tones, this is where a reINVITE will typically occur. A reINVITE is a new INVITE sent within the same dialog that established the session. In this case, the reINVITE is used to ensure that the codec used during the fax/modem call is G.711 μ-Law. It can be initiated by either the called or calling party gateway, depending on which one detected the fax/modem tones. After the reINVITE is sent with a new SDP Offer, the receiving gateway responds with an

SDP Answer in the 200 OK. The "m" parameter in the final SDP will contain the codec that will be used after the reINVITE. An example 200 OK is below:

```
# debug sip stack messages
15:33:03.328 SIP.STACK MSG      Rx: UDP src=10.1.1.1:5060 dst=10.2.2.1:5060
15:33:03.328 SIP.STACK MSG          SIP/2.0 200 OK
15:33:03.328 SIP.STACK MSG          From: <sip:5551200@10.1.1.1:5060;transport=UDP>;tag=43d8ba8-
7f000001-13c4-16aa6-e27022bf-16aa6
15:33:03.329 SIP.STACK MSG          To: <sip:5551100@10.1.1.1:5060>;tag=523c370-a13f0fa-13c4-
1282c0-33ff4f2e-1282c0
15:33:03.329 SIP.STACK MSG          Call-ID: 4415e58-7f000001-13c4-16aa6-e40300ae-16aa6@10.1.1.1
15:33:03.329 SIP.STACK MSG          CSeq: 1 INVITE
15:33:03.329 SIP.STACK MSG          Via: SIP/2.0/UDP 10.1.1.1:5060;branch=z9hG4bK-16aa6-5889aa0-
3f40b690
15:33:03.329 SIP.STACK MSG          Contact: <sip:8659@10.1.1.1;transport=UDP>
15:33:03.330 SIP.STACK MSG          Supported: 100rel,replaces
15:33:03.330 SIP.STACK MSG          Allow: ACK, BYE, CANCEL, INFO, INVITE, NOTIFY, OPTIONS, PRACK,
REFER, REGISTER
15:33:03.330 SIP.STACK MSG          User-Agent: ADTRAN_NetVanta_7100/A4.09.00.E
15:33:03.330 SIP.STACK MSG          Content-Type: application/sdp
15:33:03.330 SIP.STACK MSG          Content-Length: 236
15:33:03.331 SIP.STACK MSG
15:33:03.331 SIP.STACK MSG          v=0
15:33:03.331 SIP.STACK MSG          o=MxSIP 0 0 IN IP4 10.19.240.250
15:33:03.331 SIP.STACK MSG          s=SIP Call
15:33:03.331 SIP.STACK MSG          c=IN IP4 10.19.240.250
15:33:03.331 SIP.STACK MSG          t=0 0
15:33:03.332 SIP.STACK MSG          m=audio 50318 RTP/AVP 0 101
15:33:03.332 SIP.STACK MSG          a=rtpmap:0 PCMU/8000
15:33:03.332 SIP.STACK MSG          a=rtpmap:101 telephone-event/8000
15:33:03.332 SIP.STACK MSG          a=silenceSupp:off - - - -
15:33:03.332 SIP.STACK MSG          a=fmtp:101 0-15
15:33:03.333 SIP.STACK MSG          a=ptime:20
15:33:03.333 SIP.STACK MSG          a=sendrecv
```

The **"m"** parameter above is listed as "audio 50318 RTP/AVP 0 101."

The numbers listed after "RTP/AVP" represents the codec selection. In this example, the codec chosen was G.711 µ-Law, as 0 represents that codec (8 represents G.711 a-Law for countries that use a-Law instead of µ-Law). The 101 afterwards may or may not be present as this indicates that RFC 2833 is used to transport DTMF tones. If the first value after RTP/AVP was 18, the call would have used G.729 as the final codec.

If the call did not use G.711 µLaw, check the configuration on the trunk/user. If static provisioning was used, verify the correct codec list is referenced on the voice trunk/user and that the codec list contains G.711 µ-Law as the only option. If the dynamic provisioning approach was used, verify that the voice trunk/user has **modem-passthrough** configured and that **modem-passthrough** is enabled in the direction that the fax/modem call is being made. The default global setting is for modem passthrough to operate on both inbound and outbound calls. The global configuration command that controls this is **voice modem-passthrough-mode <both | inbound | outbound>**.

If the configuration has been verified, the issue will likely be a codec mismatch, tone detection issue, or a SIP reINVITE glare issue. A codec mismatch will present itself whenever the calling party or called party does not support G.711 µ-Law. The party that does not support the codec will likely send a 488 Not Acceptable Here or a 415 Unsupported Media Type. This could be an issue in either configuration approach.

If the issue is with tone detection, then the output of the SIP stack debug will show that the ADTRAN unit never sent a reINVITE (assuming that the global modem passthrough mode is configured to allow tone detection in the direction that the call was initiated). The command **debug voice toneservices** can be used to verify that tone detection is enabled on the call as well as whether or not a fax/modem tone was detected by the ADTRAN unit. After the called party answers the call, the following debug will indicate that tone detection is enabled for the call:

```
TONESERVICES.EVENTS fxs 0/1 – empty – Modem/FAX Detection: Request resource
TONESERVICES.EVENTS fxs 0/1 – dsp 0/1.1 – Modem/FAX Detection: DSP channel allocated
for the resource
TONESERVICES.EVENTS fxs 0/1 – dsp 0/1.1 – Modem/FAX Detection: constructed
TONESERVICES.EVENTS fxs 0/1 – dsp 0/1.1 – Modem/FAX Detection: starting
TONESERVICES.EVENTS fxs 0/1 – dsp 0/1.1 – Modem/FAX Detection: TDM map
```

The unit will then listen for fax/modem tones for 8 seconds by default. The amount of time the unit listens for fax/modem tones is controlled by the **modem-passthrough detection-time** command on the voice trunk/user.

Once a fax/modem tone has been detected, the debug output will show the detection event:

```
TONESERVICES.EVENTS fxs 0/3 – dsp 0/1.1 – Modem/FAX Detection: received T.30 CNG event
from TDM network
```

If no tone detection event occurs, a DSP capture will need to be performed on the unit to verify the validity of the tones. This will only be a concern when a dynamic provisioning approach is taken.

A SIP reINVITE glare issue can occur when the called and calling party gateways try to reINVITE simultaneously, which will only be a concern in a dynamic provisioning approach. This will usually result in each party sending a *491 Request Pending*. If the glare condition only occurs on inbound or outbound fax/modem calls, the configuration of the global modem passthrough mode can be changed such that a reINVITE will not occur in the direction of the glare event. For example, if all outbound faxes experience a glare condition, the modem passthrough mode could be changed to inbound only with the command **voice modem-passthrough-mode inbound**. If the issue occurs regardless of the call direction, the ability to initiate codec renegotiation will need to be disabled in either the ADTRAN unit or on the remote gateway. However, relying solely on the far end gateway to renegotiate the codec will prevent the echo cancellation mode from being set.