magrathea

Magrathea Telecommunications Limited,
5 Commerce Park, Brunel Road, Theale,
Berkshire RG7 4AB

0345 004 0040 info@magrathea-telecom.co.uk

INTEROPERABILITY FAQ AND BEST PRACTICE

Moving away from the legacy TDM infrastructure is an exciting time for the telecoms industry in the UK and will certainly encourage greater innovation and flexibility within the telephony network.

Whilst migrating our own network from the traditional TDM technology to a fully IP framework, we are finding that the switch from the well-established SS#7 protocol to SIP, which operates under a multitude of RFC and national guidelines, has allowed different networking interpretations which in some cases presents a more complicated interoperative process.

We are very happy to share our findings and recommendations as a result of our early migration and we hope that these can be used by our clients to assist with set up and/ or fault reporting for your own configurations.

This is not meant to be a definitive guide or replace any RFCs or vendor specific advice, it does not form a contractual part of our service – however we hope that some of the information contained in this article is useful to you.

CLI

The Magrathea network is designed to work with CLI in a variety of formats and headers to maximise compatibility with different types of equipment, but our recommendation for best practice is that you use both the FROM and the PAID headers, where the PAID field contains any assigned network number to identify the end point itself and the FROM field either contains the same value, or a different number you wish to be displayed to the end user.

The ideal format of the numbers is +E164 (e.g. +44**** for UK numbers).

If you wish to display a number that has not been allocated specifically to your network then this should be placed in the FROM header (and will be passed as the presentation number) with a number assigned to your network in the PAID (which will be passed as the network number.)

To withhold the number, you will need to add a privacy line to the invite, excerpt from INVITE below as an example:

```
Allow-Events: talk, hold, conference, presence, dialog, line-seize, call-info, sla, include-session-description, presence.winfo, message-summary, refer
  Privacy: id
  Content-Type: application/sdp
  Content-Disposition: session
  Content-Length: 201
  X-CID: f76e9782-c4e5-1236-288b-001236547393
```

P-Asserted-Identity: sip:01234567890@sip.company.com

Alternatively if you are using the RPID instead of the PAID, you will need to add a privacy flag like this:
Remote-Party-ID:
<sip:01234567890@sip.company.com;user=phone>;privacy=full;screen=yes

Codecs
Uncompressed codecs G711-PCMA(8) and G711-PCMU(0) are preferable.

It should be noted however that G711 mulaw (PCMU) has only had dominant use in North America and Japan and G711 a-law has been the PSTN standard in the rest of the world. At any time where we or our network partners have to perform a translation between the two systems there is potential for some degradation of the call to occur and therefore we recommend exclusive use of G711 a-law in most cases.

It is not uncommon however due to the North American connection for equipment default settings to be 'u' rather than 'a' and this should be checked when setting up equipment.

With increased HD voice now being available with IP interconnects it is worth offering G722 as well, preferably as the first choice, but this should never be the only choice. Good practice is to send G722 as an HD codec but with a-law included as well as a backup.

DTMF
Traditionally we always recommended that a-law used with inband was sent to and from our network, this was because the vast majority of the calls were converted to TDM and passed to a TDM network which would only be using inband, removing the requirement for our network to convert; less conversions would provide the most reliable experience.

Following the switch to IP we now however recommend that RFC2833 is the primary means of DTMF transmission. For customers where DTMF is extensively used in calls/IVRs etc, we still however recommend that G711-alaw is used as the preferred codec, as in the event that a terminating network cannot process RFC2833 then inband is still an option as a fallback.

We encourage networks to ensure that if RFC2833 DMTF sending/receiving is enabled on your network that inband detection is only ever in operation if RFC2833 is not negotiated. We find that having them operate concurrently always produced unreliable results.

Hold/release
Sending a=recvonly or a=inactive through a subsequent SIP INVITE are preferred and widely accepted by all IP Carriers and should be the method of choice.

There are other methods that can be used but are not recommended for maximum compatibility. If your equipment is unable to use the above protocol then you should ensure that you continue to send an audio stream of silence. If your RTP ceases without notification to us the call is likely to be terminated for safety after a period of time as our network will think that your equipment has failed and is no longer accessible. This time period is between 90-180 seconds.

p-time
Although the standards in use allow a variety of values to be selected, in practice we have come across many networks that can only work with ptime=20 so we would strongly recommend this as a practice. Sending other values may restrict our routing choices and affect call quality.

SIP late offer media

Whilst SIP standards and the Magrathea network support the use of late offer media for subsequent invites during the course of an established call, we do occasionally find interworking issues with some carriers when this form is used so we would recommend therefore that where possible its use is disabled when sending calls, most originating devices have an option for this.

However, we have also observed use of SIP late-offer media re-invites for established calls from major carriers. Our network does not initiate such re-invites, but if we receive them from carriers, we will often need to pass these on to customer networks.

Your equipment has to be able to cope with these correctly – in particular:
if a call is in HOLD state (via a=inactive / a=recvonly) – and you receive a late-offer re-invite (re-invite without SDP) – your equipment has to reply with 200-OK including (a=sendrecv) and take the call off hold. (We are aware that some older versions of popular open-source software, including Asterisk and Freeswitch get this wrong and should be updated to the latest version.)

SIP session timers

We encourage the use of SIP session timers to ensure that calls are safely disconnected in the event equipment malfunctions or connectivity issues. We recommend that a value of between 1200-1800 seconds is used for this timer.