Feature Overview for Software Release 6.3.2:

Abbreviations:
CUCM   Cisco Unified Communication Manager
PGW   Cisco Peripheral Gateway

- Supports ISDN (S_0, E1, U_{k0}), Analog, VoIP and GSM interfaces depending on used hardware platform and extensions
- Supports voice and data calls including channel bonding via SIP
- Supports early offer for SIP data calls
- Supports delayed offer for incoming SIP voice calls, outgoing SIP voice calls use early offer
- Interworking with CUCM 7.1.2
- Interworking with PGW 9.8.1 behind CUCM 7.1.2 (NovaTec → CUCM 7.1.2 → PGW 9.8.1)
- S3 can be configured in CUCM 7.1.2 as SIP end device ("TransNova S3")
- Name presentation/restriction is supported. The name of a user can be configured in the NovaTec system and will be sent to the CUCM.
- Supports Hold and Resume, Call forwarding and Call transfer with CUCM 7.1.2 (Call forwarding and call transfer will be handled locally in the NovaTec system, for S3 Music on Hold of CUCM will be used, for other NovaTec systems the Music on Hold is locally generated.
- The diversion header is supported and will be send out in INVITE messages for diverted calls. This is used to show the diverting number to the called user. It is also a required feature for diverting calls to voice mailboxes.
- Synchronisation for clock sensitive applications like videoconferencing is possible via GPS
- Network Management System version 7.0 is now supported. This includes the following new call home events:
  - Layer 1 active/inactive
  - Layer 2 active/inactive
  - CPU-Threshold
  - RAM-Threshold
- The # sign is now recognized as End Of Sign (EOS) to speed up the dialling process for outgoing SIP calls
- Supported Codecs: G.711 a-law/µ-law, G.726-16, G.726-24, G.726-32, G.726-40, G.728, G.729 and Clear Channel Codec for transparent data calls
- Supports silence compression, ISDN echo cancellation and comfort noise generation
- Supports T.38 (T.38 including annex D)
- VLAN support according to 802.1Q
- Supports MD5 authorisation
- Supports NAT mapping
- Supports STUN client and server according to RFC 3489
- Diagnostic information via HTTP
- Supports NTP
- Supports SIP session timer according to RFC 4028
- Supports SRTP (AES 128 encryption) for voice and data calls according to RFC 3711
- Remote access via ISDN and IP
- SIP support according to RFC3261
- Supports Digest authentication according to RFC2617
- Supports Reliability of provisional responses according to RFC3262
- Supports SIP offer/answer model according to RFC 3264
- Supports In-Band/Out-of-Band DTMF according to RFC2833
- Supports SIP Extensions for Caller Identity and Privacy
- Supports SIP media inactivity Timer
- Supports DNS name resolution and ENUM