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1 INTRODUCTION

1.1 Scope

This specification describes the IPCablecom Call Management Server (CMS) to CMS Signaling protocol intended for use by a CMS to communicate with another CMS in order to support packet-based voice and other real-time multimedia applications. The protocol exchanges between a CMS and a Media Gateway Controller (MGC) are identical to those between CMSs, and so for purposes of this specification the MGC is considered identical to a CMS. CMSs currently support multimedia endpoints (within the IPCablecom infrastructure) that use the Network-based Call Signaling [24] (NCS) protocol and the PSTN Gateway Call Signaling Protocol [25] (TGCP) for communicating signaling information between the endpoint and the CMS. In the future, other protocols may be supported as well, and the CMS to CMS protocol is intended to be sufficiently general to accommodate such protocols without change.

The CMS to CMS protocol uses the Session Initiation Protocol 2.0 (SIP) specification with extensions and usage rules that support commonly available local and CLASSSM services. This protocol is referred to as the Call Management Server Signaling (CMSS) protocol.

The CMSS protocol takes into account the need to manage access to network resources and account for resource usage. The usage rules defined in this specification specifically address the coordination between CMS Signaling and IPCablecom Dynamic Quality of Service (QoS) mechanisms for managing resources over the cable access network. In addition, this specification defines the protocols and messages needed between Call Management Servers for supporting these services.

This document specifies the protocols and procedures to use between CMSs belonging to a single service provider as well as between CMSs that belong to different service providers. In the case that the CMSs are owned by multiple service providers, it is assumed that the service providers have a mutual trust relationship.

Other IPCablecom documents describe interfaces between other system elements. These documents cover areas such as: Event Message recording for billing and other back office functions [23]; Dynamic Quality of Service [21]; Operations and Provisioning [61]; Electronic Surveillance [22]; and Security [26]. These other specifications indirectly place requirements on the signaling protocol to ensure that it transports the correct data needed to implement a complete system. This document includes syntax and protocols for implementing these requirements. Currently, the document does not address interworking with non-IPCablecom-compliant devices.

From time to time this document refers to the voice communications capabilities of an IPCablecom network in terms of "IP Telephony." The legal/regulatory classification of IP-based voice communications provided over cable networks and otherwise, and the legal/regulatory obligations, if any, borne by providers of such voice communications, are not yet fully defined by appropriate legal and regulatory authorities. Nothing in this document is addressed to, or intended to affect, those issues. In particular, while this document uses standard terms such as "call," "call signaling," "telephony," etc., it should be recalled that while an IPCablecom network performs activities analogous to these PSTN functions, the manner by which it does so differs considerably from the manner in which they are performed in the PSTN by telecommunications carriers, and that these differences may be significant for legal/regulatory purposes. Moreover, while reference is made here to "IP Telephony," it should be recognized that this term embraces a number of different technologies and network architecture, each with different potential associated legal/regulatory obligations. No particular legal/regulatory consequences are assumed or implied by the use of this term.